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Integration of Autonomous Sender for Hidden Log Data on Kleptoware for Supporting Physical Penetration Testing

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***Abstract*—Keylogger is a dangerous device that can capture all word that typed on the keyboard. There are two kinds of keylogger, it is hardware and software keylogger. It is very easy to detect them because both of them already listed as a malware. There are a lot of antivirus application that can detect software keylogger and for the hardware keylogger, it is easily can be seen if there is a strange thing that attached to our computer. Kleptoware is one of the solutions to hardware keylogger main problem. Another problem comes when we want to take all data had been capture on the device, we must take the keylogger first. This paper discuss about how to gain data from a kleptoware autonomously with client-server design on a local area network. Result in this paper shows that data must be send at least had same file size with the buffer that already determine first.**

Keywords—autonomous sender, keylogger, kleptoware

I. Introduction

Kleptoware is a hardware keylogger that hiding in a keyboard using Teensy USB, microcontroller with very small size. The biggest problem in hardware keylogger, which is easily seen when attached to a computer, can be solved. There are two kinds of kleptoware, USB-type and PS/2-type. Both of those kinds, works correctly like any other keyboard, but it steal all of the keystroke that had been pressed[1].

The biggest problem on hardware keylogger is the visibility of the keylogger itself. Now, for common people already understand if there is a strange device attached over the computer. The idea of the kleptoware is hiding a keylogger into a keyboard since a keyboard has space to implement other things inside. The secondary problem on hardware keylogger comes when we want to read all the data that have been typed over the keyboard. If we following the usual way to gain the data, we must secretly take the keylogger and read it in attacker's computer. This problem also solved when autonomous system implemented on the computer.

The focus in this paper is to implement and integrate an autonomous system can send the logged data that has been capture in the keylogger. The system that discussed in this paper covers a logged data that can be transferred from a victim to a chosen server and send it periodically and implemented over one-to-one communication method. Integration of autonomous system for kleptoware, make a kleptoware able to send data that had been captured to a chosen server.

II. Theory

A. Penetration Testing

In the basic concept, penetration testing has same meaning with hacking, but it is different if looked from other side. Main purpose in penetration testing is for early-detection in threat that own by a system. The result of this testing is useful for owner of the system to prevent a hacker that want to hack it. There are a lot of cases that make the system's owner losses material and non-material things over hacking activities that attacked their systems.

Penetration system needed by any corporation that hold criticals data such as finance data in a bank. The newest technology in banking system is internet banking. This feature uses internet and web as a protocol to communicate with customers. Since we cant control the internet, we must control the system in our area. Penetration system must be done at least twice in a year to avoid any mistakes.

B. Keylogger

Keylogger divided into two kinds, software and hardware. For each kind of keylogger, there are several weakness.In example, software keylogger detected by antivirus, hard to install, and so on. And for hardware keylogger, it is very easy to install, just plug the keylogger between computer and keyboard. It is very easy, but in this era, many people understand that there is something strange thing attach to their computer, and it is very hard when we must read the captured data.

Development of hardware keylogger makes a keylogger works not only for captured keystroke, but also picture and video. Some company already sell a keylogger that can capture a picture and video from our computer's screen[4]. This new type of keylogger, sold in the market freely. There is also a GPU-based keylogger that can store captured data on the GPU[7].

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Keylogger can be impelemented in two-way of hacking, black hat and white hat. It can be include in the black hat category when it is designed and implemented, but it will included in white hat category when creating a detection and prevention system[6].

As said before, there is a big problem when implemented hardware keylogger. There are a step how to use a PS/2 hardware keylogger[5] :

1. Set up correctly your keylogger.
2. Unplug the keyboard.
3. Plug the keylogger to the keykeyboard.
4. Replug the keyboard to the PC.
5. Pray no one sees it.

C. Kleptoware[1]

Seen at the step of implementing the keylogger, at the step 5, no one must not see the keylogger. The idea of kleptoware is how to make the keylogger definitely unseen. Kleptoware provides data capture on keyboard's keystroke that typed. It is success in hiding hardware keylogger in a keyboard, proven by testing on visibility the kleptoware. The keyboard works normally but there are also affected in several ways to use the keyboard. There are some delay when held keystroke, key ghosting, and key jamming.

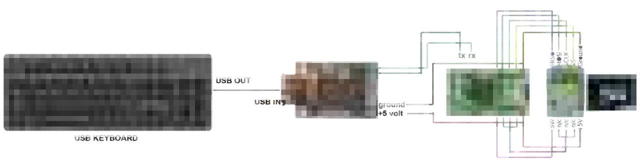


Fig. 1. USB-type Keylogger[1]

As shown in figure above, implementation of kleptoware in USB-type, need an additional device because USB can't communicate easily with other device. It needs USB Host Adapter to work well and helps kleptoware to translate USB scan code into customized ASCII character[3]. USB-type Keylogger had a minor problem in keystroke delay, key ghosting and jamming, but for overall USB-type runs well.

D. Winsock

Transferring data between client and server needs a way of communication, in this case it used an IP communication to transfer logged data. It used TCP/IP protocol to communicate. Communication between client and server can be implemented by using windows socket API, or usually called winsock. Socket is to transport endpoint, it used to send and receive data.

Process on the server side more complex than the client side. It can be seen in fig.2, on the server side, the socket must be bind first, and must listen the connection and accept whenever there is a connection from the client. In the client side, it should connect to server first, while server were listening for communication. After creating connection between client and server established, the communication began and client can send data to a server.

III. SYSTEM DESIGN AND IMPLEMENTATION

Design that proposed in this paper, provides autonomous sender from kleptoware and it designed only use one targeted computer and send it into a server which is still in one local area network. Fig 3. Shows about the system design that proposed in this paper.

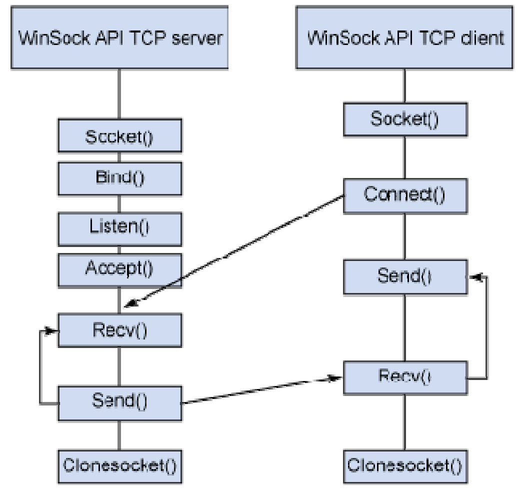


Fig. 2. WinSock client-server interaction[2]

Kleptoware itself works using extra small microcontroller called Teensy USB and add a MicroSD Card Adapter. It can store all the data that had been captured when any keystroke were hitted and stored all in a MicroSD card. Kleptoware implemented in target's computer secretly with swapping the real keyboard with the kleptoware. When target hit the keystroke, the kleptoware save all the keystroke that hit in its internal storage. The autonomous system detect whether there is a data log that captured in the kleptoware or not. If there is a data log, the autonomous sender will send it into local server in the network.

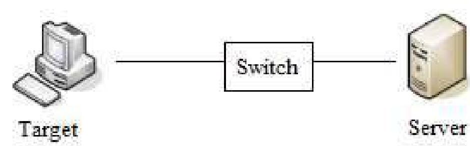


Fig. 3. System Design

This is an example of data log that captured by the kleptoware. Kleptoware can record any keystroke that typed in the keyboard, include all the supporting button in the keyboard such as, backspace, enter, etc.

20[KEY_BACKSPACE][KEY_BACKSPACE][KEY_BACKSPACE]testerscccccv.txta[KEY_DELETE]snippingcoba1[KEY_ENTER]
how w[KEY_BACKSPACE]make runtime[KEY_BACKSPACE][KEY_BACKSPACE][KEY_BACKSPACE][KEY_BACKSPACE][KEY_BACKSPACE][KEY_BACKSPACE][KEY_BACKSPACE][KEY_BACKSPACE] run prog[KEY_BACKSPACE][KEY_BACKSPACE][KEY_BACKSPACE][KEY_BACKSPACE]visuvisual studio program without visual sr[KEY_BACKSPACE]tudio runtime[KEY_ENTER]

Fig. 4. Example recorded data

As shown from the captured data above, this kind of data that will send into a server periodically using the feature of winsock. The type of data that stored in kleptoware is a txt file. This file that will be sent to server. The process of sending and receiving can be seen at Fig. 4.

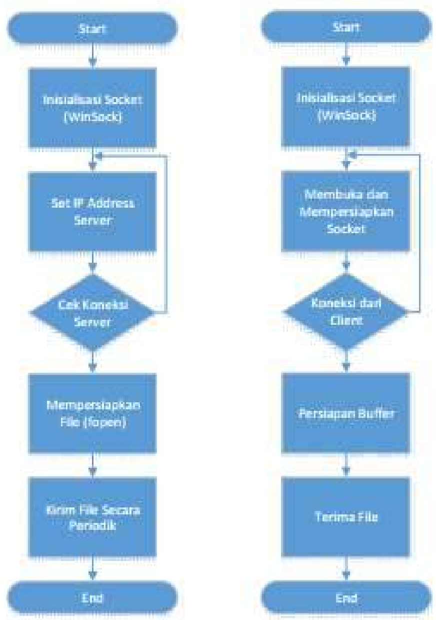


Fig. 5. Autonomous Send and Receive Process in Client and Server

These are steps for server's program :

1. Initialization WSA
2. Creating socket
3. Binding socket
4. Listening socket
5. Accept connection
6. Send and receive data – recv(), send(), recvfrom(), sendto() (In this program, server activated recv() function)
7. Disconnect

These are steps Client's Program :

1. Initialization WSA
2. Open Socket
3. Create connection to server
4. Send and receive data (in this program, client act as sender and send the data periodically to the server)
5. Disconnect

These autonomous process had a limitation with size of a file captured. It's limited due to buffer size that use. If the file size more than buffer size, sending process will be interrupted and at the server, received file will be different from client's file. These figures below describe the implementation of client and server program.

Fig. 6. Implementation of Client's Program

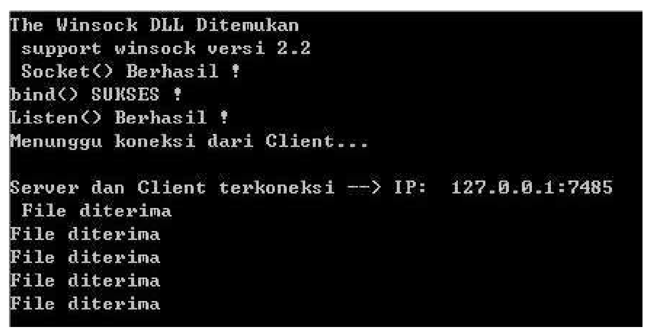


Fig. 7. Implementation of Server's Program

IV. Results and Analysis

Due to limitation of buffer size, data that captured will be changed into random data that can be reached the limitation file size. At Fig. 7 shows us logged client's data that changed into some text. This file will be sent to server periodically. Fig. 8 shows the client's data at the server side, before the program runs, the file at the server still empty. Buffer size in this case is using 4096.

Fig. 8. Logged Client's Data

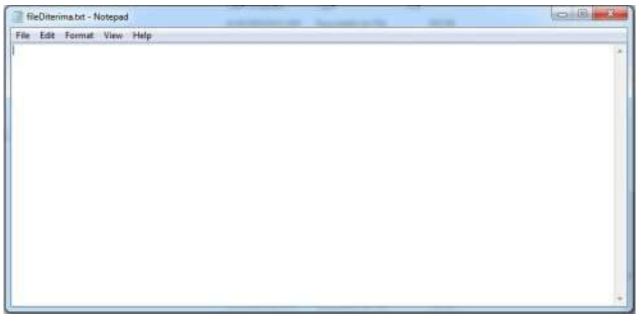


Fig. 9. Empty Client's Data on The Sever

When the application on client's side runs, it will automatically send the data periodically. Seen at Fig.9 server already received data from client. This first file received completely because the file size under the buffer size.

Fig. 10. Received Client's Data on The Server

As long application on client and server run, the data will always updated. More text typed in client, more text will be sent to server. Seen at Fig. 10, described any update on client side and it will be affected at server side (Fig. 11). Server will received data that already added at the client side.

Fig. 11. Additional Logged Client's Data

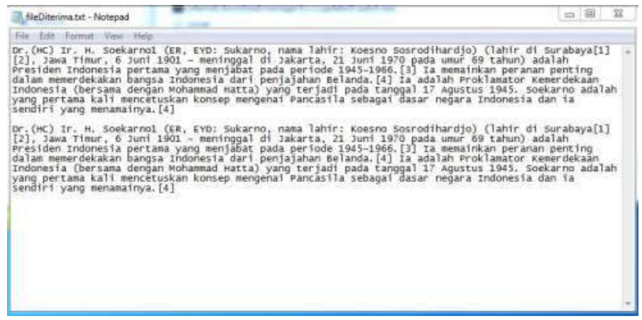
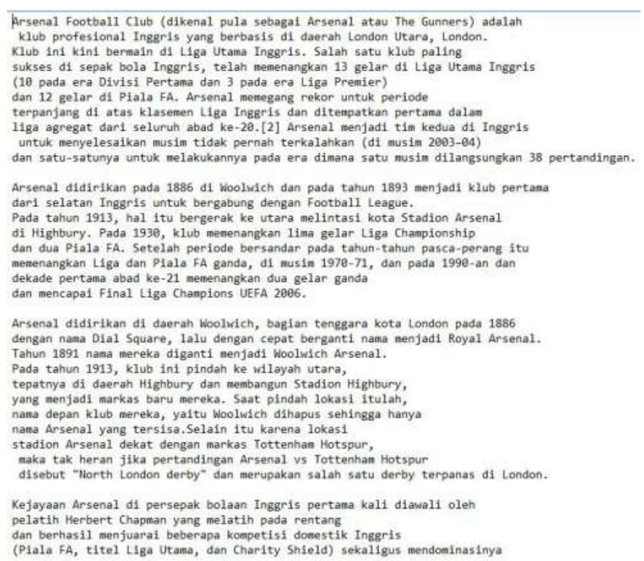


Fig. 12. Autonomous Update Received Client's Updated Data

At Fig. 11, all the data from client will be received, because buffer size that set at application is 4096, and the data is less than 4096. With buffer size 4096, it is equal 100 times using username and password with 20 character for each. In this research, testing the buffer size threshold. Fig. 12, shows us the example captured data that had size more than the buffer size.


Arsenal Football Club (dikenal pula sebagai Arsenal atau The Gunners) adalah
klub profesional Inggris yang berbasis di daerah London Utara, London.
Klub ini kini bermain di Liga Utama Inggris. Salah satu klub paling
sukses di sepak bola Inggris, telah memenangkan 13 gelar di Liga Utama Inggris
(10 pada era Divisi Pertama dan 3 pada era Liga Premier)
dan 12 gelar di Piala FA. Arsenal memegang rekor untuk periode
terpanjang di atas klasemen Liga Inggris dan ditempatkan pertama dalam
liga agregat dari seluruh abad ke-20.[2] Arsenal menjadi tim kedua di Inggris
untuk menyelesaikan musim tidak pernah terkalahkan (di musim 2003-04)
dan satu-satunya untuk melakukannya pada era dimana satu musim dilangsungkan 38 pertandingan.

Arsenal didirikan pada 1886 di Woolwich dan pada tahun 1893 menjadi klub pertama
dari selatan Inggris untuk bergabung dengan Football League.
Pada tahun 1913, hal itu bergerak ke utara melintasi kota Stadion Arsenal
di Highbury. Pada 1930, klub memenangkan lima gelar Liga Championship
dan dua Piala FA. Setelah periode bersandar pada tahun-tahun pasca-perang itu
memenangkan Liga dan Piala FA ganda, di musim 1970-71, dan pada 1990-an dan
dekade pertama abad ke-21 memenangkan dua gelar ganda
dan mencapai Final Liga Champions UEFA 2006.

Arsenal didirikan di daerah Woolwich, bagian tenggara kota London pada 1886
dengan nama Dial Square, lalu dengan cepat berganti nama menjadi Royal Arsenal.
Tahun 1891 nama mereka diganti menjadi Woolwich Arsenal.
Pada tahun 1913, klub ini pindah ke wilayah utara,
tepatnya di daerah Highbury dan membangun Stadion Highbury,
yang menjadi markas baru mereka. Saat pindah lokasi itulah,
nama depan klub mereka, yaitu Woolwich dihapus sehingga hanya
nama Arsenal yang tersisa.Selain itu karena lokasi
stadion Arsenal dekat dengan markas Tottenham Hotspur,
maka tak heran jika pertandingan Arsenal vs Tottenham Hotspur
disebut "North London derby" dan merupakan salah satu derby terpanas di London.

Kejayaan Arsenal di persepak bolaan Inggris pertama kali diawali oleh
pelatih Herbert Chapman yang melatih pada rentang
dan berhasil menjuarai beberapa kompetisi domestik Inggris
(Piala FA, titel Liga Utama, dan Charity Shield) sekaligus mendominasinya

Fig. 13. Captured Data more than Buffer Size

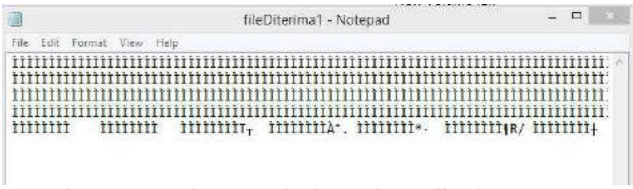


Fig. 14. Captured Data Received more than Buffer Size on Server

The result shows at Fig. 13, error happened at file in the server. Data from client is different from data at the server. Result in the server, shows data at client must have less size from buffer. At table 1., there are some further testing in file size. Testing number 16, had same size with buffer that already set in early testing (4096), and the result is success. Testing number 17 and above, shows that the sending process is failed, which means the received data in server is not same with client's data.

TABLE I. FILE SIZE TESTING

No	File Size (Byte)	Status	No	File Size (Byte)	Status
1	50	Success	11	1000	Success
2	150	Success	12	1500	Success
3	200	Success	13	2500	Success
4	300	Success	14	3000	Success
5	350	Success	15	4000	Success
6	400	Success	16	4096	Success
7	450	Success	17	5000	Failed
8	500	Success	18	6000	Failed
9	700	Success	19	7000	Failed
10	900	Success	20	8000	Failed

V. CONCLUSION

In this paper we implement a autonomous sender system into a kleptoware. With this ability, a kleptoware can send the captured data itself, and done it periodically. Results of testing and analysis shows that the autonomous sender can works perfectly as long as the file size of captured data smaller than the buffer that used in the application (in this case the buffer sets at 4096). Every update in client's side, can be seen in the server's side also. The limitation can be avoided when size of buffer set in bigger size.

REFERENCES

[1] Nasution, SM, Yudha Purwanto, Agus Virgono, Girindra Chandra Alam. Integration of Kleptoware as Keyboard Keylogger for Input Recorder Using Teensy USB Development Board. The 8th International Conference On Telecommunication System, Services, And Application. 2014.

[2] Manage mobile cloud socket connections
http://www.ibm.com/developerworks/cloud/library/cl-mobilesockconnect/

[3] USB Keyboard to ASCII Converter
http://www.hobbytronics.co.uk/usb-host-keyboard

[4] Hardware Keylogger - Wireless Keylogger - Do It Yourself! http://www.keelog.com/wireless_keylogger.html

[5] Protect Yourself Against This Popular Hack: Hardware Keylogger
http://www.instructables.com/id/Protect-Yourself-Against-This-Popular-Hack%3a-Hardwa/

[6] A.Wood, Christopher. dan Rajendra K.Raj .(2010). "Keyloggers in Cybersecurity Education". New York : Rochester Institute of Technology.

[7] Ladakis, E, Koromilas, L., Vasiliadis, G., Polychronakis, M., & Ioannidis, S. (2013). You Can Type, but You Can't Hide: A Stealthy GPU-based Keylogger.Prague : Czech Republic.

Frequency Interference Analysis between UMTS and CDMA 2000 in 2.1 GHz Frequency Band

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***Abstract*—As a natural limited resources, the radio frequency spectrum should be managed and used as efficiently as possible. The Government as regulator has a responsibility to manage and regulate the spectrum to prevent interference. One example of the many problems in Indonesia that must be solved is the spectrum in 2.1 MHz frequency band for 3G services. The CDMA 2000 and UMTS system occupy adjacent band and that very potential to cause interference each other. The CDMA 2000 downlink channel which is separated by only 3.125 MHz guard band, have the potential to interfere the UMTS uplink channel. It is especially in the last channel with a center frequency of 1977.5 MHz. In this paper we evaluate an interference that occur between the two system by computer simulation and experimentally measurement. We propose three methods to cope with this problem, namely spectrum segmentation with sufficient separation band, geographical separation, and additional filters on the BTS. Antenna spatial isolation method is also used so that both systems can operate co-location in which the level of the interference is acceptable. With the arrangement of the antenna vertically by 1,1 meters separation, the additional attenuation of 15.2 dB can be obtained to reduce interference.**

Keywords*—*Interference, UMTS, CDMA 2000, 2.1 GHz band, minimum separation distance

I. Introduction

In the evolution of communication technology, different types of technologies may be deployed in adjacent frequency bands. For example in cellular communication technologies, IS-2000 or CDMA 2000, operating in the PCS-1900 band and WCDMA (UMTS) operating in the IMT-2000 band. In Indonesia, CDMA 2000 and UMTS are deployed in the adjacent frequencies in 2.1 GHz band by goverment regulation as describe in Fig. 1. The CDMA 2000 system occupied frequencies from 1903.125 MHz to 1910 MHz (uplink) paired with 1983.125 MHz to 1990 MHz (downlink) and the UMTS system occupied frequencies from 1920 MHz to 1980 MHz (uplink) paired with 2110 MHz to 2170 MHz (downlink). There are several possible interference that can occur between CDMA 2000 and UMTS system. The worst case scenario is adjacent channel interference between BS CDMA 2000 transmitter and Node B UMTS receiver, which are separated with 3.125 MHz guardband. There are several ways can be done to prevent interference in both systems, such as spectrum segmentation with sufficient guarband [1][2], geographical separation [1][4][5][6], additional filters on both BTS [4][5][7], and antena spatial isolation [4][5].

To analyze the interference, the study was conducted geographical separation method and antenna spatial isolation method through statistical simulation based on Monte Carlo method. The geographical separation method was needed to obtain the minimum separation distance (MSD) between BS CDMA 2000 and Node B UMTS, where the interference was minimum. Meanwhile, the antenna spatial isolation method was needed to minimize the interference when the both systems co-located.

II. System Description

This paper focus to analyze the interference between BS of CDMA 2000 transmitter and Node B UMTS receiver. Interference simulation was performed by using SEAMCAT (Spectrum Engineering Advanced Monte Carlo Analysis Tool) software where the value of technical parameter refer to the 3GPP and 3GPP2 recomendation.

The design of the simulation begins by modeling both system of UMTS (WCDMA) and CDMA 2000 in the simulator. Environment and propagation condition are then modelled to represent the real condition of the problem to calculat dRSS and iRSS as well as the probability of interference. Interference simulation process begins with creating the workspace and set the parameters of control simulation. Victim and Receiver Wanted modeling based upon parameters and technical specifications such as the level of sensitivity, bandwidth, antenna gain, direktifity of antenna and receiver susceptibility. Meanwhile, modeling of wanted and interfering transmitter is based on the distribution of power level, gain, antenna height and directivity. We use the propagation modeling based on Extended-Hata propagation model within urban and suburban environment. After modeling the receiver, transmitter, channel propagation, and environment, we then simulate scenario of the signal interference level between NodeB UMTS and BS CDMA 2000.

A. System Model

The system model in the SEAMCAT terminology shown in Fig. 2. The victim link consist of Node B UMTS as the *victim receiver, Vr* and UE UMTS as *wanted transmitter, Wt*. The interfering link consist of BS CDMA 2000 as the *interferer transmitter*, *It* and MS CDMA 2000 as the *wanted receiver*, *Wr*. *Vr* received the *desired Received Signal Strength*

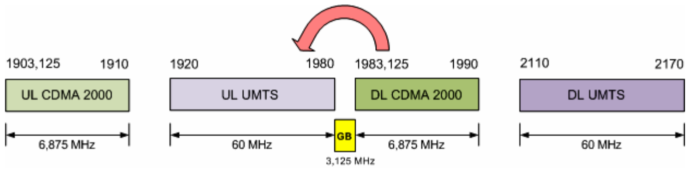

Fig. 1 Spectrum of UMTS and CDMA 2000 in 2.1 GHz Band.

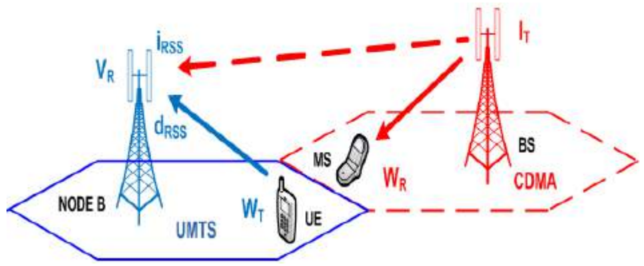

Fig. 2 The System Model in SEAMCAT terminology.

(dRSS) from *Wt*, and the *interfering Received Signal Strength (iRSS)* from *It*. The difference between *dRSS* and *iRSS* is defined as the *protection ratio* or *carrier to interference ratio (C/I)* which is measured in dB. This ratio must be more than the required C/I threshold if the interference to be avoided. Or the *iRSS* must be less than the required unwanted received signal level. We also perform modification of parameters and simulation scenarios in order to get the most minimal interference level indicated by the level of high protection ratio (C/I).

B. System Parameter

As the victim, Node B UMTS simulated to receive signal on the last channel of the uplink band, i.e. at 1977.5 MHz with a bandwidth of 5 MHz, as shown in Fig.3. The parameters of UMTS system shown in Table I. The parameters refer to the technical specification of NodeB UMTS for calculation. BS CDMA 2000 in this evaluation act as an interferer which is simulated to radiate the interferer signal. In our case we consider that BS CDMA 2000 generate two kind of signal, one channel (SR-1) and three channel contiguous (SR-3) at the frequencies and bandwidth as described in Table II. These parameters refer also to the specification of CDMA 2000. To make our evaluation more reliable, the environment and propagation condition between the two systems are collected from the real condition in the field.

III. Methodology and Scenario

In order to observe the level of interference perceived by the NodeB of UMTS, we use a statistical simulation model that uses Monte Carlo methods using SEAMCAT software. SEAMCAT applying simulation of random process by randomly taking values from a probability density function. The methodology and scenario that we consider in this paper

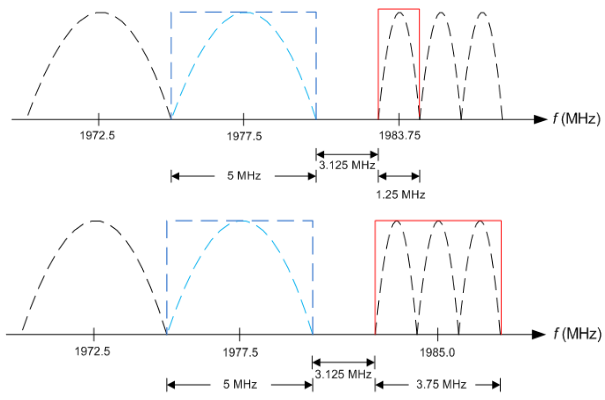

Fig. 3 UMTS Uplink Channel and CDMA Downlink Channel.

Table I Parameters of UMTS System.

Parameter	Value	Unit
Frequency	1977,5	MHz
Bandwidth	5	MHz
Node B Noise Floor	-104	dBm
Node B Rx Sensitivity	-103	dBm
Node B Antenna Gain	18	dBi
UE Tx Power	21	dBm
UE Antenna Gain	0	dBi

Table II Parameters of CDMA 2000 System.

Parameter	Value		Unit
	SR-1	SR-3	
Frequency	1983,75	1985	MHz
Bandwidth	1,25	3,75	MHz
BS Noise Floor	-104	-104	dBm
BS Tx Power	43	43	dBm
BS Antenna Gain	12	12	dBi
MS Rx Sensitivity	-116	-116	dBm
MS Antenna Gain	0	0	dBi

Table III Environment of Simulation.

Environment Parameter		UMTS		CDMA 2000	
		Node B	UE	BS	MS
Urban	Antenna Height	25 m	1,5 m	25	1,5 m
	Location	Outdoor	Outdoor	Outdoor	Outdoor
	Propagation Model	Extended Hata			
Sub Urban	Antenna Height	40 m	1,5 m	40 m	1,5 m
	Location	Outdoor	Outdoor	Outdoor	Outdoor
	Propagation Model	Extended Hata			

are as follows. (a) Unwanted emissions, consisting of the spurious emissions and out-of band emissions (OOBE) of the interfering transmitter, that are represented by the Adjacent

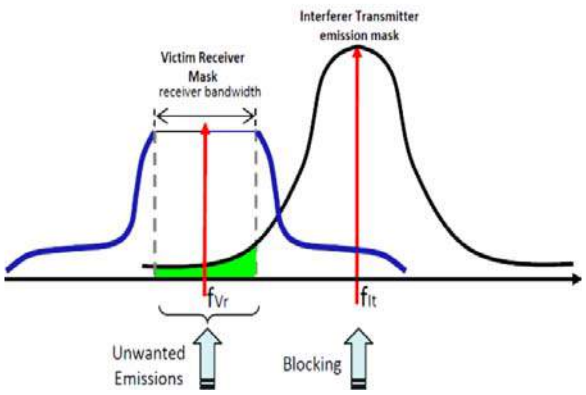

Fig. 4 Unwanted Emissions and Receiver Blocking Mechanism.

Channel Leakage Ratio (ACLR) falling within the victim's receiver bandwidth; (b) The receiver blocking power, a combination between the Adjacent Channel Selectivity (ACS), that refers to the filter receiver capacity to avoid unwanted emissions, and the blocking mode defined. The mechanism is illustrated in Fig.4 [8].

Interference will occur when the signal received by a receiver cannot meet the minimum value of C/I. To calculate the value of C/I experienced by the receiver, it is necessary to set statistics level of wanted and unwanted emission emission. Unwanted emission is assumed to come from another transmitter that transmits fall in a wide band receiver reception. When interference is introduced, the interference adds to the noise floor of the victim receiver. The difference between *dRSS* and *iRSS* is defined as the carrier to interference ratio or protection ratio, which is measured in dB. This ratio must be more than the required C/I threshold if the interference to be avoided. The Monte Carlo simulation methodology is used to check for this condition and records whether or not interference is occuring.

The systems assumed operating in wide area base station. The emission mask used in the SEAMCAT simulator referring to the 3GPP and 3GPP2 recommendations. *Wt* emission mask refer to 3GPP recommendation in Table 6.10. [10]. *It* emission mask refer to 3GPP2 recommendation in Table 4.4.1.3-2.[11]. The receiver blocking of *Vr* refer to 3GPP recommendation [12].

Unwanted signal threshold ($iRSS_{thd}$) is determind from the level of thermal noise floor (TNF) and the Node B receiver sensitvity degradation[4][5]. Thus the unwanted signal threshold is given by

$$iRSS_{thd} = 10\log\left[10^{\left(R_X\,Sensitivity\,\deg radation/10\right)} - 1\right] + TNF\,[dB] \quad (1)$$

$$TNF = 10\log\left(KT\right) + 10\log B + NF\ [dB] \quad (2)$$

where K denotes Boltzman constant, T denotes Room temperature (290 K), B is bandwdth (Hz), and NF is a Noise Figure (dB). The C/I threshold defined as the difference between dRSS minimum (receiver sensitivity) and $iRSS_{thd}$ as expressed by

$$\frac{C}{I} = Sensitivity - iRSS_{thd}\ [dB] \quad (3)$$

If the noise figure of receiver assumed by 3 dB and RF bandwith is 5 MHz, then TNF is -104.01 dBm. Considering the noise margin of 1 dB, the receiver sensitivity is taken -103 dBm. Assuming the receiver degradation is 0.4 dB, $iRSS_{thd}$ will be the -114.16 dBm, then the C/I_{thd} is 11 dB.

The scenario of simulation consist of non co-located and co-located scenario. Non co-located represented in Scenario 1 and Scenario 2, meanwhile the co-located represented in Scenario 3.

- *Scenario 1*
  The BS CDMA 2000 simulated to radiate one carrier at a frequency of 1983.75 MHz with a bandwith of 1.25 MHz in the urban and sub-urban environment. There is no antenna arrangement on Node B or BS. The distance between BS and Node B changed gradually to obtain a minimum separation distance (MSD) on the value of C/I or the unwanted signals level required.
- *Scenario 2*
  The BS CDMA 2000 simulated to radiates three contiguous carriers at a center frequency of 1985 MHz with a bandwith of 3.75 MHz in the urban and sub-urban environment. There is no antenna arrangement on Node B or BS. The distance between them changed gradually to obtain a minimum separation distance (MSD) on the value of C/I or the unwanted signals level required.
- *Scenario 3*
  The BS CDMA 2000 and Node B UMTS are placed co-location without any antenna arrangement. The BS was simulated as in Scenario 1 and Scenario 2 in the urban and sub-urban environment. The distance between UE and Node B changed gradually so that the received wanted signal level reaches the Node B sensitvity level. This scenario will obtained the coverage degradation of the Node B caused by the unwanted signals from BS.
- *Scenario 3 with Mitigation Methods*
  In this scenario, the BS CDMA 2000 and Node B UMTS are placed co-location with an antenna arrangement. The vertically spatial arrangement [3] at the antennas required to obtain the additional attenuation to met the value of C/I or the unwanted signal level allowed.

IV. Simulation Result and Analysis

In scenario 1, wanted transmitter Wt (UE) is located at separated two conditions, namely close to the victim receiver position Vr (NodeB) and at the end of the coverage of the victim receiver in which the received signal level is

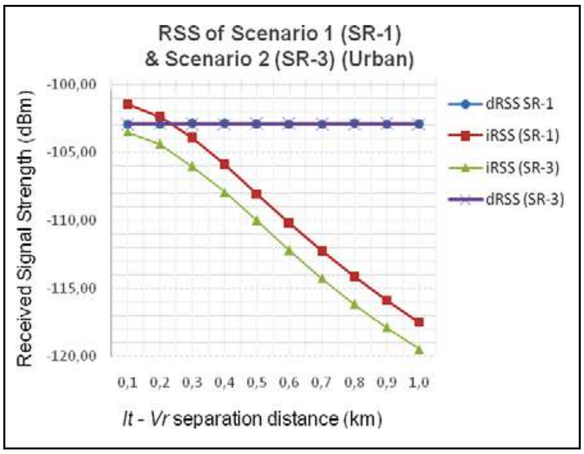

Fig. 6 Received Signals Strength of Scenario 1 and Scenario 2 in the Urban Environment.

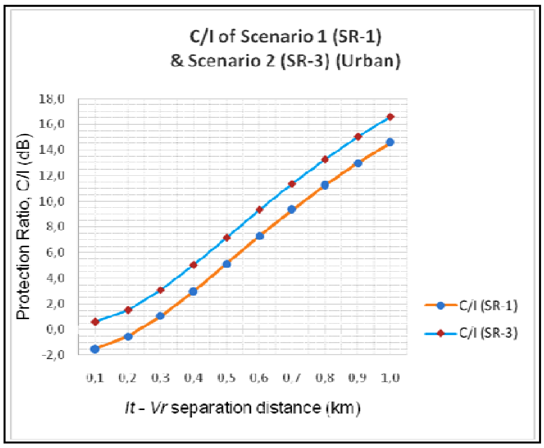

Fig. 7 C/I of Scenario 1 and Scenario 2 in the Urban Environment.

approaching the limit of its sensitivity level. Position of wanted receiver Wr (MS) is at the center of interfering coverage transmitter, (It BS), while the interfering transmitter position changed against the victim receiver to get the value of MSD. In this scenario, the position of the antenna on both systems is not determined.

In scenario 2, wanted transmitter, Wt (EU) also put on two conditions, which is close to the victim receiver position, Vr (Node B) and at the end of coverage victim receiver where the received signal level approaching the level limit sensitivity. Position wanted receiver, Wr (MS) at the center of the coverage interfering transmitter, It (BS), while the interfering transmitter changed position change against the victim receiver to get the value of MSD. Are not done setting the position of the antenna on both systems.

In scenario 3, the position of the victim receiver Vr (Node B) and the interfering transmitter It (BS) are simulated at the same location (co-located). The position of wanted transmitter, Wt (EU) is varied from the victim receiver. until the level of the received signal Vr approaching the limit level of sensitivity. Meanwhile, wanted receiver, Wr (MS) is located within the coverage of interfering transmitter, It (BS).

The result of Scenario 1 and 2 in th urban environment are shown in Fig. 6. The MSD, where the $iRSS_{thd}$ achieved, are at a distance of 800 meters for urban environment. The value of C/I is also observed and found to be 11 dB at that distance. In the Scenario 2, the MSD is observed at a distance of 700 meters in urban environment. The value of C/I is obtained 11 dB at that distance such as described in Fig. 7.

In Scenario 3, the unwanted received signal, *iRSS* is more than the $iRSS_{thd}$. So, the Vr coverage radius has reduced from 300 meters to 130 meters (SR-1) and from 300 meters to 145 meters (SR-3). Scenario 3 simulation results, is shown in Fig. 7 for urban environment. From the figure it can be seen dRSS received that level Vr approaches the sensitivity obtained in Wt against Vr distance is 300 meters. This distance is the radius of coverage of Vr against Wt when the absence of It. It is not meet the level of unwanted iRSS that the threshold value of -114 dBm, so Wt at a certain distance in Vr coverage will experience a call drop if the value of C/I does not meet the value of C/I threshold by 11 dB.

We also observe the value of protection ratio (C/I) experienced by Vr is always greater than the threshold value of the C/I desired as described in Fig. 9. It because Wt near dRSS always greater than the value iRSS. Because the value of protection ratio C/I is much larger than the threshold value in these conditions the EU do not experience drop calls. The value of C/I threshold that is above 11 dB, is obtained on the condition of the distance between It to Vr of 700 meters or more. This means that the EU within the scope of Node B does not experience the drop call, if the distance BS to the Node B over 700 meters or more. So it can be concluded in this scenario is the MSD value of 700 meters. Identical with such previous simulation results, the value of protection ratio C/I experienced by Vr is much larger than the threshold value of the C/I of 11 dB, so that the EU does not experience this condition drop call. C/I threshold above 11 dB is obtained on the condition of the distance between It to Vr 1.35 km or more. This means that the EU within the scope of Node B does not experience the drop call, if the distance BS to the Node B over 1.35 km or more. So it can be concluded in this scenario the value of MSD is at 1.35 km.

IV. Conclusion

The frequency interference between UMTS and CDMA 2000 systems in 2.1 GHz band (in UMTS uplink channel at 1977.5 MHz) may occur if the isolation between both systems not met, where the unwanted received signal level is more than -114 dBm or the value of C/I is less than 11 dB. Based on the simulation result, the minimum distance required to separate Node B UMTS from BS CDMA 2000 are 800 meters (SR-1) and 700 meters (SR-3) in the urban environment, and 1.7 km (SR-1) and 1.35 km (SR-3) in the sub-urban environment. When the Node B and BS co-located, the interference causes degradation of Node B coverage radius from 300 meters to 130 meters (SR-1) and 145 meters (SR-3) in the urban

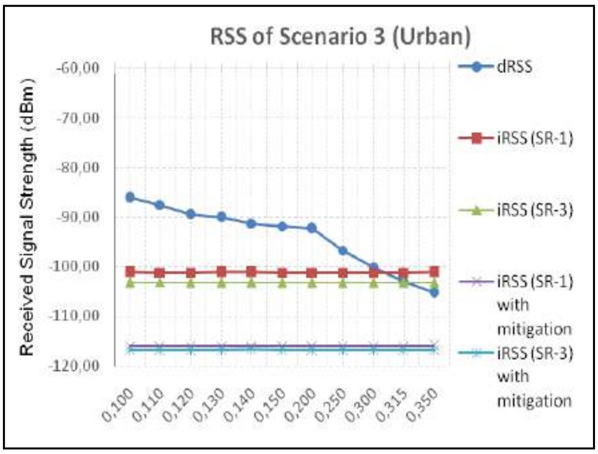

Fig. 8 Received Signals Strength of Scenario 3 in the Urban Environment.

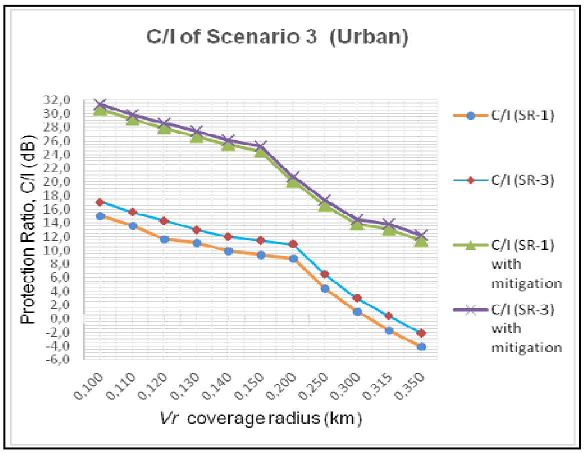

Fig.12. C/I of Scenario 3 in the Urban Environment.

environment, and from 825 meters to 310 meters (SR-1) and 355 meters (SR-3) in the sub-urban environment. Interference mitigation methods by antenna spatial isolation with 1.1 meters vertically separated will produce an additional attenuation of 15.2 dB which can reduced the unwanted received signal level when Node B co-located with BS.

V. References

[1] Settapong Malisuwan, Amorn Ieamgusonkit,Vichate Ungvichian, “BS to BS Interference of WCDMA and PCS1900 (IS-95)”, IEEE, 2008

[2] Indriani Timang, Herman Y. Kanalebe, “Interference Impact of WCDMA Mobile Station on IS-95 Mobile Station Performance”, IEEE, 2008.

[3] Okorugu Victor N., Onoh G.N.,Nwalozie G.C., Onyishi D.U.,“Antena Isolation Technique for Interference Reduction in A Co-Site Systems”, IJARCCE Vol.2 Issue 7, July 2013.

[4] Huawei Technologies Co., Ltd. Presentation,”Interference and Guarband Analysis between CDMA1900 and UMTS2100”, 2012

[5] Qualcomm Corporate Engineering Services Presentation,”Co-existence Check of CDMA2000 in PCS and UMTS in IMT-2000 Bands, Testing and Analysis Plan (80-W3641-1 Rev A)”, 2012.

[6] AUSPI,“Evaluation of Co-existence interference between CDMA 1900MHz & WCDMA 2100MHz”, Detailed interference analysis report for co-existence of CDMA/EVDO& GSM/WCDMA in 1900MHz frequency band, Oktober 2007.

[7] Qualcomm Engineering Service Group, "Interference Analysis and Guidelines for The Coexistance of Mobile Wireless Technologies (80-W2561-1 Rev A)", 2009.

[8] ECC Report 68 ”Monte-Carlo Simulation Methodology for the use in Sharing and Interference Studies Between Different Radio Services or Systems”, Electronic Communication Committee, Baden, 2002

[9] Report ITU-R M.2244, “Isolation between Antennas of IMT Base Stations in the Land Mobile Services”, ITU, 2011.

[10] 3GPP Rec. TS-125 101 v.12.5.0 (2014-10) “UMTS User Equipment (UE) Radio Transmission and Reception (FDD)”,3GPP, 2014

[11] 3GPP2 Rec. C.S0010-E v2.0 (2014-03) “Recommended Minimum Performance Standards for CDMA 2000 Spread Spectrum Base Stations”, 3GPP2, 2014.

[12] 3GPP Rec. TS-125 104 v.12.4.0 (2014-10) “UMTS Base Station (BS) Radio Transmission and Reception (FDD)”,3GPP, 2014

Mobile TV Interactive in DVB-T Broadcast Network Hybrid with WiFi

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***Abstract*–Digital television (TV) has been replacing analog TV in many countries. Content broadcasting applications therefore are developed widely including on the smart phone as the non-DVB enabler device. This paper aims at developing an application on the smart phone that receives program and transmits a request to the server as DVB-T transmitter. The target is to develop an application that allows smart phone interactively connected to DVB-T transmitter. As the first target we develop an android-based smart phone. We use DVB-T signal for downlink transmission and wifi for the uplink connection. This interactivity may include file sharing, polling, etc. using non-DVB enabler devices. Therefore the development of a system that can convert DVB signal into WiFi-IP based protocol signal is big challenge in this work.**

***Keywords:* DVB, RTSP, Mobile TV, ESG, polling, android.**

I. INTRODUCTION

The development of television broadcasting services has reached broadcasting technology with digital. It has many standard that determined by various world organizations in its implementation. One is the standard of DVB (Digital Video Broadcasting) which is the standard of ETSI and is the standard choice for broadcasting in many countries in Europe. Mobile Digital TV was born from the desire of people to be able to watch live TV anywhere and at anytime. Through mobile broadcasting, the public can receive the information they need on the spot. Mobile broadcast can also fill in the spare time when you're relaxed.

Mobile digital TV (DTV) is also beneficial to the content providers. With mobile DTV, broadcasters can extend the range of the audience. By using mobile DTV content providers can change the start time Prime Time (Hours with the largest audience) from 7-8 hours a night when the audience arrived home from work to hours of 5-6 pm when people are on the way home [1].

However, in its application, Mobile DTV with DVB technology faced some problems. Most of the existing devices are non-DVB enabler devices. However they have a WiFi connection. Therefore, in order non-DVB enabler devices enjoying digital TV broadcast on their devices, we have to develop an application on both side, at the transmitter side and at the receiver side. We need a tool or system that can capture and convert DTV broadcasts that can be resumed as the WiFi signal and received on handheld device users. This tool is a tool for bridging between WiFi-DTV signal. Thus we can setup a way to continue the DTV broadcast signals to mobile devices using WiFi connection. Next should have been a platform for developing this system. If we look to the smart phone operating system (OS), the most widely used are Apple's iOS and Google's Android OS. Android so easily get references required and no additional fee is required to carry out development with Android. Additionally Android devices in the market quite a lot and covers various economic layers of both low-end devices such as mini smart phones to high-end smart phones and tablet. For this reason we use Android-based smart phone to develop our application.

This paper aims at designing the software that is able to implement a system running in the platform of Android OS-based handheld device in receiving DTV broadcast via DVB technology. We also propose an application service that allows user to develop interactivity with the server.

II. DVB AND SYSTEM ARCHITECTURE

Digital Video Braodcasting or DVB is an open international standard (open standard). DVB standards set by the DVB Project. DVB Project is an industry consortium with more than 270 companies. DVB-T is the DVB standard for digital terrestrial television transmission. DVB-T is able to bring the data rate is high enough. Usually some broadcast digital television broadcast can be done simultaneously and can also be included audio channels. Each transmission is called a DVB-T interface multiplex. OFDM (Orthogonal Frequency Division Multiplex) is used as an air interface in a DVB-T technology.

The modeling is based on some of the activities that constitute steps in developing the software. The steps include planning; implementation; testing; and evaluation. Those are iteratively executed to achieve the target as described in Fig. 1.

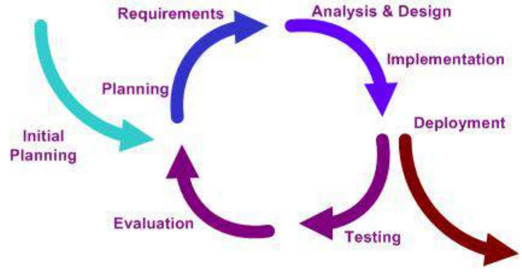

Fig. 1 Iterative development model.

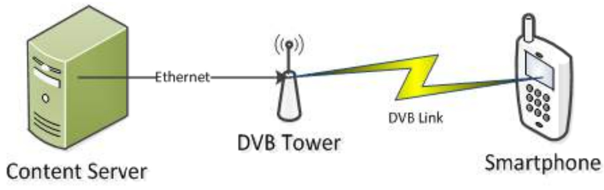

Fig. 2 The developed simple system architecture.

The simple system architecture consists of content server, DVB transmitter, and non-DVB receiver, which in this case we used a smart phone as depicted in Fig. 2. The problem is that smart phones are unable to receive DVB signal because they are not DVB receiver. To cope with this problem, we developed the system that can be replacing the function of DVB transmitter using PC to convert DVB signal into wifi signal. The system architecture as described in Fig. 3 then is probably used as one choice to realize this goal. In the Fig. 3, video content is transcoded using Transcoding and Restream Server to convert DVB signal into IP protocol signal. With this method, we maintain the broadcast to keep real-time.

In the whole system, it is needed two server. Those are content server and transcoding and restream server. Content server manages and control program that will be broadcasted, while transcoding and restream server prepare video broadcast to be ready for client to access. Some of the parameters considered in the developed system are operating frequency at 586 MHz or 578MHz, transmission bandwidth of 8 MHz, convolutional rate of 7/8, 8k transmission mode, guard interval 1/32, use QAM-64 constellation transmission, and using OFDM modulation types.

However, with such a scheme thus arise various questions. How a handheld device can receive data via DVB if the device does not have a DVB receiver. Then how the content server associated with the device to perform activities of interactivity? It is known that the connection is a connection DVB broadcast a one-way data transmission. With this simple scheme, we cannot answer our target. Therefore, the system is changed so that the scheme requires a special server to set up a list of events/ESG. To achive this purpose we need a computer to perform bridging DVB-WiFi. Finally, the system architecture we develop comprises two separate network. One is the broadcast video content via DVB channel for the link from DVB transmitter to the users and the uplink channel for the users to have interactivity with the network via WiFi.

In Fig. 3, content server is the device to store all the video and content applications and services. From this server, video are transmitted via DVB modulator to the trans-coding and re-stream server. In this server, DVB signal is converted to IP protocol format and then it is broadcasted via wifi to the users. We use also ESG server as a source of video content that delivered to the content server. ESG is an electronic service guide, which is the main part that governed the user experience in enjoying the content. Through the ESG server, the user can achieve on-demand services and interactive services.

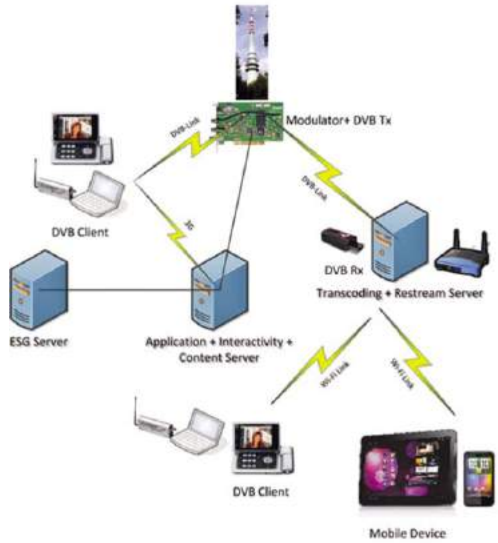

Fig. 3 Realized system architecture of developed system.

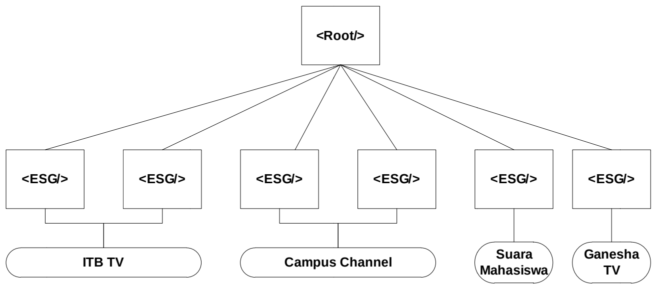

Fig. 4 Relationship model in ESG.

Because of there are so much data and function stored in the ESG, so it needs a fixed standard in the format of the data stored on ESG. Data on ESG is sent to the user device in the form of text to XML format. ESG is sent to the device at specific intervals so that users always get the latest data. The composition of the ESG always remains fixed, but the text data is changed along with the changes that occur in the system. ESG.xml's readings is done using a Java package named xml parser. Through xml parser, an XML file is converted into an XML DOM object and is stored in a variable type of documents. So that it can be processed using java package org.w3c.dom. When the file is ready to be processed, the program will scan the file and search for the required tag element. Once ESG was read, the data of each element will be stored in a global variable to be used by other classes when necessary. This ESG relationship can be described in Fig. 4.

An important application that developed in our application is the polling system. Polling system is the communication of multiple users to the content server. Thus, on the server side software needs to be created that can process the data. A polling function class need to do the following action:

- View the question and the list of choices to the users of the data stored in ESG.
- Can accept user input via the selection key.

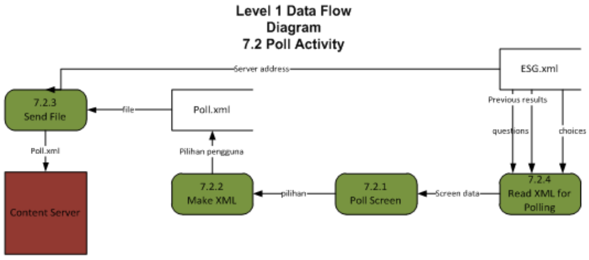

Fig. 5 Level 2 data flow diagram of the development poll activity.

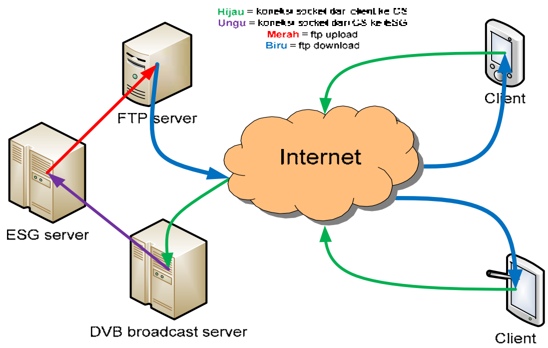

Fig. 6 Data flow diagrams polling service feature.

Fig. 7 Application installed on the smart phone showing an interactivity.

Fig. 8 Video is being played on tablet.

- Can create an XML file that contains the user's choice and send through the socket programming.

This activity starts by running the poll 7.2.1 screen. This procedure is used to display the poll to the user interface so that it can be seen. Poll screen gets input of data that has been read in the Read XML procedure. This data comes from ESG previously downloaded via FTP server.

User input is received and the choice be entered for the procedure 7.2.2 Make XML where the procedure is formed XML-formatted files named poll.xml. This file will then be sent to the content server for processing. Then created a procedure called send files that served to send this to the delivery method poll.xml socket using TCP/IP. At the polling activity there is no data processing such as the calculation and determination of winners, etc., because it is done on the content server.

Interactive communication is done by exchanging xml file. Attendance sends an xml file containing the choice to set up an ftp server. Furthermore, this file is processed, and the poll results are stored on the same ftp server. The audience then takes this xml file and then read the results of the poll. That is why the method of transmission used is FTP upload and FTP download. File exchange sub-system is given in Fig. 6.

To provide polling service feature, the server must be able to read files in xml format. The reason is that XML is a file format that can run on various operating systems. Xml is also a file format that can store information in a certain classification making it easier to transfer information between systems briefly. Besides being able to read the xml, the server must also be able to write an xml. It aims to store and transmit data computation and compilation of poll results to the ESG server. The essential elements that must be stored and transmitted by the content server is the channel number, the options on each channel, and most importantly, the amount of the poll results of each choice. For that, each channel has its own segment with the child provides options on each channel and has value in the form of the results of the number of voters that choice. Each channel can also be attribute contains a description of the channel or questions that will be polling about.

III. Interface and Implementation

Software that is being developed is called "Pocket TV" as described in Fig. 7. This naming is based on the original concept of making this software. At the testing of this software we use multiple devices with the Android operating system made by three different manufacturers. Through testing with a variety of different brands is expected to see the software compatibility across multiple platforms. Apart from the hardware side, there are also differences in the versions of the operating systems that run. Fig. 8 shows the video has been played by the tablet of *Samsung Galaxy Tab 10.1*. And also we can play the application on the *ViewSonic ViewPad 10s* that use Android OS version 3.0 *Gingerbread*. On other devices, we used a 2.2 version of *Froyo*.

This application has high portability for all devices with platforms android. The requirement of playing such a video was have a codec to play the video format mp4 and have a processor of at least 524 MHz (the lowest test). At the reception of video from all video formats are tried (mp4, avi, flv, mp2) the most compatibility with various devices is MP4. While this application has been authenticated way to the Android platform OS version 2.2 and above.

RTSP is a protocol used to send data via streaming video. RTSP reception naming format is: rtsp: //192.168.1.100: 5544 / stream

- rtsp: // protocol is a reliable address
- To unreliable used rtpu: //
- 192.168.1.100 is the IP address of the source video (content server)
- : 5544 is the port used to control stream.
- /stream is a stream server path

RTSP has two types of data sent. First there is the data packet itself containing data yan want to send. Secondly, the control message is a message that contains the settings controlling data flow and connections between host and client. Control of this message can be DESCRIBE, SETUP, PLAY, PAUSE, and teardown [4] In the RTSP connection, there are two types of data sent TCP and UDP. From 2504 bytes received, in the form of data is 2298 bytes. Overhead for controlling is the data packets until the termination of the formation of connections, including data residing on TCP. So the total on controlling overhead is 2504 bytes.Tipe second package is UDP. UDP the data contained on the video you want to watch by the user. In the UDP delivery also has overhead for routing, error detection, etc. Using wireshark obtained from 20617265 bytes sent by the server to the client, 20612527 bytes are data. So that the total overhead in the UDP for this session is 4738 bytes.Maka total overhead is raised for the delivery of video for 5 minutes (20.612 kB) is 7242 bytes.

IV. CONCLUSIONS

We have developed an application of interactive system based on DVB broadcast transmitter using non-DVB devices. User devices have been designed to be able requesting some program or application that are provided by content server through ESG server. The program application installed inside the non-DVB devices is able to request program or content through wifi network, while receiving program from DBV transmitter using DVB standard protocol.

REFERENCES

[1] Texas Instruments. (2005) Digital broadcast TV-Coming soon to a mobile phone near you. pdf.
[2] M. Shalahuddin Rosa A.S., Pemrograman Berorientasi Objek. Bandung, Indonesia: Modula, 2010.
[3] Nelson To James Steele, The Android Developer’s Cookbook - Building Applications with the Android SDK. Boston, USA: Addison-Wesley, 2010.
[4] Westerhoff. (2006, September) Wikipedia Images. [Online]. http://en.wikipedia.org/wiki/File:Iterative_development_model_V2.jpg
[5] H. Schulzrinne, et. all, “Real Time Streaming Protocol (RTSP),” IETF RFC 2326, April 1998, http://www.ietf.org/rfc/rfc2326.txt.

VSAT Network Quality Analysis on Mobil Pusat Layanan Internet Kecamatan in North Sulawesi

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Network quality analysis is conducted to ensure the quality of the network by using certain parameters. This paper describes the measurement results of Very Small Apperture Terminal/VSAT network quality in the District Internet Service Mobile Center or MPLIK which operated in the province of North Sulawesi. Network quality parameters that were being measured are bandwidth, delay, packet loss and upload and download. Standard measurement results using TIPHON version. Pointing the location of the sample, conducted at five different points in North Sulawesi region with a variety of geographical conditions, such as the highlands, coastal, dense location of the communication signals and other random location point. Network quality measurement results for various parameters indicate that MPLIK network quality is low or below standard TIPHON.

Keywords: MPLIK, VSAT, Quality, Network

1. Introduction

The Government initiated MPLIK program (District Internet Service Mobile Center) using VSAT technology so that every community, especially those in the rural areas that were not being reached by conventional internet access can discover and use the internet. However, this program, in its implementation have many complaints that are related to the quality of a given network.

This paper presents the results of the research related to the quality measurement of the existing VSAT network in MPLIK which is operating in the province of North Sulawesi. As for the writing of this paper is intended to answer the following research problem: How are the quality measurement results of VSAT MPLIK network using bandwidth parameter, delay, packet loss and upload-download speed?

The network quality measurement results, are expected to be a recommendation for the operators and policymaker stakeholders of MPLIK program.

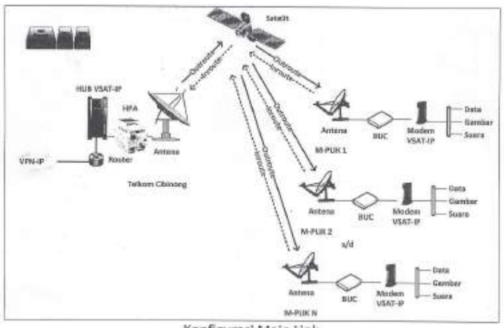
1 Theories

1.1 Mobil Layanan Internet Kecamatan

Mobil Layanan Internet Kecamatan (MPLIK) is a mandate from Article 5 of the Decree of The Minister of Communication and Information Technology No. 48/PER/M.KOMINFO/11/2009 on Internet Access Service Provider In the Subdistrict Internet Area of Universal Telecommunication Service, as amended by the Decree of the Minister of Communication and Information Tehnology No. 19 / PER / M.KOMINFO / 12/2010.

MPLIK is a synergy between KPU/ISO activity program with the Community Access Point (CAP), the providers target of 1,907 MPLIK is being spread throughout Indonesia and there are 16 MPLIK that were shared in every regency/city in North Sulawesi. Inside of an MPLIK contains a 1.2 m VSAT antenna, modem HN 7700S, 6 pieces of notebook, 1 PC servers, switch, UPS (Unit Power Supply), DVD (digital video disc) player, chair, folding table, TV (Television) LCD (Light Emiting Dioda) and a generator to provide backup powers if the electricity from PLN (Perusahaan Listrik Negara) went off. [1]

The topology being used in MPLIK is star topology, it can be seen in Figure 1 below.



Konfigurasi Main Link

Gambar 1 Topologi Jaringan VSAT di MPLIK

1.2 Very Small Apperture Terminal/VSAT

VSAT (Very Small Aperture Terminal) is signal receiver station of the sattelites with a dish-shaped receiver antenna and a diameter of less than three meters. The main function of VSAT is to receive and send data to a satellite. Sattelite works for transmitting signals to another point of the earth. In fact, the VSAT dish antenna is facing a geostationary satellite. A geostationary satellite is a satellite that is always in the same position in line with the earth's rotation on its axis as it orbits the same point on the surface of the earth, and follows the rotation of the earth on its axis.

1.3 VSAT Network Quality Parameters

Network quality parameters being used in this research are: bandwidth, delay, packet loss, and upload and download. Bandwidth is the wide frequency range used by the signals in the transmission medium in sending or receiving computer network data. It is usually measured in bps (bits per second) or kbps (kilo bits per second). Whereas Upload and Download are the number of the VSAT speed for transmitting and receiving data. Delay is the time delay of data packet or the length of time needed for transmission from server to the VSAT hub. The types of delay which being measured on VSAT network is a. Propagation delay, is the amount of time it takes for the information signals to travel inside the communication media such as cable, fiber optic, microwave and satellite; b. Transmission delay, is the amount of time a system takes to pass a number of data packet and c. Delay queue, is the length of time a data packet needs before the packet is forwarded to its destination. The standardization according to TIPHON (Telecommunications and Internet Protocol Harmonization over Networks).

Table 1. TIPHON Delay Standarization

Category	*Delay*
Very Good	< 150 ms
Good	150 s/d 300 ms
Average	300 s/d 450 ms
Poor	> 450 ms

Packet Loss is packets lost during network transmission, or in other words, Packet Loss is the failure of the data packet from reaching its destination. For some possibilities of network overload, a collision in the network, as well as a disturbance in the hardware.

Table 2 TIPHON Packet Loss Standarization

Category	*Packet Loss*
Very Good	0 %
Good	3 %
Average	15 %
Poor	25 %

2 Discussion

The research locations were conducted at 5 sites in North Sulawesi that were determined randomly based on geographic conditions, they are Tomohon City as it is conditioned in highlands area, Sam Ratulangi University as it is conditioned in downtown of Manado city, Manado Coast as its condition in coastal areas, Sam Ratulangi Airports for it has high level activity of data communication, and Kayuwatu that has Blindspot condition. Blindspot can be interpreted as a blind spot, where the particular area is difficult to be reached by the network.

The tools and materials that were used in this study are:

a) an MPLIK,

b) Axenet Tools software, a desktop-based application that can perform monitoring and measurement of a network, Axence NetTools can display the results in the form of a graph. The research of analyzing the performance of MPLIK VSAT network quality is using the NetWatch tools of Axence NetTools, to monitor the network.

c) Elnus Bandwidth Meter, which is a web-based application that can perform bandwidth measurement of the network

d) CBN Speedtest, which is a web-based application that can perform measurements of ping, upload and download from a network

The data collection phases are:

a) Preparation, preparing MPLIK equipments and software that will be used, including making a plan in determining the point of locations to collect data

b) Pointing MPLIK VSAT antenna with satellite. Pointing is the earth station antenna alignment process towards the satellite position in order to get maximum signal. To be able to do the pointing, azimuth angle adjustments and elevation should be made. Azimuth angle is the angle that is created by rotating an axis that is perpendicular to the horizontal plane and clockwise, with north as the reference point (zero calculations). Whereas the elevation angle is an angle created by rotating an axis parallel to the horizontal plane, the horizontal plane as reference point (zero calculations). The first step of pointing the antenna is releasing the VSAT lock which is in the form of 1 piece bolt, bolt that is located on top of the MPLIK as well as making sure the LNB coaxial cable and BUC on VSAT are installed or not detached as these would greatly affects the installation process.

c) After pointing is done, the next step is making an internet connection. If the Internet connection has occur then the next step is to start the process of collecting the network quality data by measuring the quality of the network. For the bandwidth measurement, the author were using ELNUS Bandwidth Meter tools, whilst for the data measurement of delay and packet loss were using NetWatch tools of the Axence NetTools software, and as for the uplink and download the author were using the tools of CBN Speedtest. Figure 2 shows the measurement results of the measuring softwares

Figure 2 The measurement results using Elnus BandwithMeter, Axence.netTools and CBN SpeedTest

Measurement data were collected from the 5 locations point, and then were compared to the standard measurement of TIPHON. The following table shows the results of the data analysis.

Table 3 Bandwidth Analyis

Location	**Available Bandwitdh (Kbps)**	**Bandwitdh Measurement (Kbps)**
Tomohon City	256	211.80
Sam Ratulangi University	256	220.90
Manado Coast	256	159.80
Sam Ratulangi Airport	256	213.40
Kayuwatu	256	178.40

Table 4 Delay Analysis

Location	**Delay (ms)**		**Feasibility**	
	Min	**Max**	**Average**	
Tomohon City	760	967	876	Poor
Sam Ratulangi University	703	996	824	Poor
Pesisir Pantai Manado	696	1000	905	Poor
Sam Ratulangi Airport	675	964	790	Poor
Kayuwatu	615	998	847	Poor

Table 5 Analisa Packet Loss

Location	**Packet Loss**			**Feasibility**
	Sent	**Loss**	**% Loss**	
Tomohon City	15	3	20	Average
Sam Ratulangi University	35	18	51	Poor
Manado Coast	20	14	70	Poor
Sam Ratulangi Airport	26	11	42	Poor
Kayuwatu	435	334	77	Poor

Table 6 Upload and Download Analysis

Location	Available (Kbps)		Measurement	
	Up	Down	Up	Down
Tomohon City	256	256	159	193
Sam Ratulangi Unversity	256	256	218	136
Manado Coast	256	256	136	168
Sam Ratulangi Airport	256	256	215	207
Kayuwatu	256	256	132	76

3 Conclusions

The conclusion that can be drawn are as follows:

The results of bandwidth, delay, packet loss, upload and download analysis on MPLIK VSAT network in North Sulawesi are as follows

Bandwidth Analysis: The result of bandwidth measurement in the research of MPLIK VSAT network quality performance analysis, either in the morning or afternoon were still below the available bandwidth on the MPLIK VSAT network which is below 256 Kbps, but visibly the difference is not too distant, so it can be concluded that the bandwidth on the MPLIK VSAT network is good enough.

Delay Analysis: Measurement results for delay in the whole area, both in the morning and afternoon were not feasible following the TIPHON standardization.

Packet Loss Analysis: The measurement result for packet loss on the entire area both in the morning and afternoon were not feasible following TIPHON standardization.

Upload and Download Analysis: The measurement results for upload and download in the research of MPLIK VSAT network quality performance analysis, in either morning or afternoon were still below the uploads and downloads that are available on MPLIK VSAT networks which is below 256 Kbps and also there are some data below 100 Kbps. The visible difference is quite distant, meaning that it can be concluded that the upload and download of the available MPLIK VSAT network were poor.

REFERENCES

[1] Aditya, Alanur. 2011, Mahir Membuat Jaringan Komputer, Dunia Komputer.

Axence NetTools. 2014, Axence NetTools User Manual, Axence Sofware Inc 2006 – 2013.

B.U, Donny. 2014, Catatan Ringkas Tata Kelola dan Praktik Internet Indonesia.

CBN. 2014, CBN Speedtest (http://speedtest.cbn.net.id/)

Dewo, S. (2010), Bandwidth dan Throughput. (http://www.ilmukomputer.org/wp-content/uploads/2006/08-dewobandwidth.zip.)

ELNUSnet. 2014, ELNUSnet Bandwidth Meter (http://bwmeter.elnus.net.id/)

Fatoni, 2011. "Analisis Kualitas Layanan Jaringan Internet (Studi Kasus : Universitas Bina Darma").

Grata Putra, Yevi. 2012, "Analisis Kualitas Jaringan VSAT Pusat Layanan Internet Kecamatan Kabupaten Lahat".

Metrasat, 2013. Bahan Training MPLIK, Kementrian Komunikasi Dan Informatika.

Sofana, Iwan. 2013, Membangun Jaringan Komputer, Mudah Membuat Jaringan Komputer (Wire dan Wireless) untuk Pengguna Windows dan Linux, Informatika Bandung.

Stallings, William. 2007. Data and Computer Communications 8th Edition.

Telkom Indonesia, 2010. Manual System MPLIK, Kementrian Komunikasi Dan Informatika.

Tiphon. "Telecommunication and Internet Protocol Harmonization Over Network (TIPHON) General Aspec of Quality of Service (QoS)", DTR/TIPHON-05006 (cb0010cs.PDF).1999.

Wikipedia. 2014, VSAT – Wikipedia Bahasa Indonesia, Ensiklopedia Bebas (http://id.wikipedia.org/wiki/VSAT)

Adult Image Classifiers Based On Face Detection Using Viola-Jones Method

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This research consists of a classification system to determine the adult and benign image. Adult image in this research was defined as image that is perceived as pornographic by Indonesian people. The method in this research combines face detection and HS skin color detection on an image. Face detection is done by using the Viola-Jones method. After face detection process, skin detection is performed on the image. Based on the results of face and skin detections, a set of features is extracted and inserted into the classifier. The classification used in determining adult or benign image will be based on the percentage of face area in the image, the position of face in the image, and the percentage of the skin color in the image. For each feature, the threshold value is defined in this research. The results of the classifier is whether an input image is benign or adult images. From 30 sample images, the classification process classifies 5 pieces of images as benign images and 25 images as adult images. False positive rate are 2 images and false negative rate is 1 image with the accuracy of 90%.

Keywords - Face detection, Viola-Jones method, Adult image.

I. INTRODUCTION

Image is a combination of points, lines, shapes, and colors to create and imitation or replica of a physical object or item (humans, animals, plants, ect) [19]. Along with the development of graphics technology, image are fully utilized for the welfare of mankind. However, on the other side, can also be a destructive force moral and human behavior, including a negative image as a pornographic image or adult image. Definition of pornographic images are very difficult to be interpreted because pornographic images that contain a lot of sense perception depends on each person. Indonesian Crime Law does not explain the definition of pornographic image clearly and detailed. In Major Indonesian Dictionary, pornography is defined as the depiction of erotic behavior with a painting or image to arouse desire. But the desire of stimulation can be very subjective, depending on the views of the beholder.

Pornographic image has penetrated the public, especially students. In general, students who are mostly young people have a tendency to try new things and his curiosity is very high. It is found in many cases that the pornographic images can lead to negative behavior and criminal sexuality, as well as to damage students' moral and mental. Therefore we need an action to restrict and inhibit the expansion of the pornographic or adult images in order not to be freely enjoyed by the general public as well as to protect society from these images that can damage behavior, mental and moral society, especially of the youths/students. One of the actions is classifying the images that are accessed via the internet using digital image processing with human face and skin detection method. In the last few years, many image processing methods used to detect human face in a frame. The detection of human face is very important in the development of digital image processing. From a face image, many features can be obtained, such as eyes, nose, and mouth features. Even the number of faces in a frame can be calculated. These kind of researches have been carried out with the advantages and disadvantages.

Currently, applications that use the face detection has been developed. Face detection can be done in various methods, one of them is using the Viola-Jones method, which combines support vector machines, boosting algorithms and cascade classifier [17]. This method is applied to a digital image, to obtain position of the face in the image. These methods get results quickly, accurately, and efficiently than other face detection methods [Viola et al, 2004]. Viola-Jones method is the mostly used algorithms to detect faces recently.

This research designs classification system to determine the adult and benign image. This is performed using a combination of face detection using Viola-Jones method and skin color detection based on Hue and Saturation values (HS). Sample images from this research of 30 which taken from the internet. Sample images is limited to only have one face with the position of the face image is upright (frontal), not hindered in part by another object, not a lot of cut, and did not move.

II. THEORY AND BASIS LITERATURE REVIEW

A. *Literature Review*

Research on the classification of adult images was performed by Ries et al. (2015) using the three groups in the approach to the introduction of adults among which the color-based approach, the approach shape information, and local feature descriptors [4]. Wijaya (2015) describes the pornographic image is an image containing the genital element of the human body that are affected by the pose, lighting and backgrounds variations. the method used is combining the the scale invariant descriptor of skin region of interests (ROIs) of pornographic images [5]. Chan et al. (1999)

which used skin segmentation algorithm and succeeded to automatically detect pornographic image [1]. Similar research was developed by Lin et al. (2003) by designing a system to detect pornographic image by using knowledge of correlation in skin and non-skin regions [6]. Then followed by Qing-Fang et al. (2004) by promoting pornography detection through a hybrid approach to discriminate benign image from the adult image by combining face and adaptive skin detections [8]. Zheng et al. (2004) using skin detection detection of adult images that appear in Internet. In this research Skin detection is of the paramount importance in the detection of adult images. They build a maximum entropy model for this task [9]. Mean while, Wang et al. (2005) conducted a research in detecting and filtering pornographic images and compared them with bikini images [12]. Hu et al. (2007) conducted similar reasearch about the introduction of pornographic Web pages through text and image classifications [14]. Agbinya et al. (2007) used a modeling value (HSV) for human skin color segmentation and detection of pornographic images to classify the human skin color in HSV space [15]. In this research, the previous research by Qing-Fang et al. (2004) is continued by designing a more detail classification system of adult and benign images. This is done by employing a combination of Viola-Jones face detection and skin color detection based on the color of HS value states.

B. *The Theory*

1. *Face detection*

Face detection is a problem of pattern classification where the input is images and the output is class label of those images. There are two class labels, face and non-face. Face recognition techniques is performed by the assumption that the face data possibly has the same size and background. In the real world, this assumption is not always valid because the face can appear in various sizes and positions with varied backgrounds [10]. Face detection is more important. This is why the face detection is put as the first step before face recognition process. Research fields related to face processing are:

- Face recognition is to compare the input face image with a database face and find the face that best matches with the input image.
- Face authentication is testing the authenticity/similarity of a face with the face of data that has been entered previously.
- Face localization is the face detection but assuming there is only one face in the image
- Face tracking is approximate location of a face in the real time video.
- Facial expression recognition is recognize human emotions.

The challenges faced in the face detection problem caused by the following factors:

- Position the face. The position of the face in the image can vary is upright position, tilt, turn, or viewed from the side
- The components of the face such as mustache, beard, and glasses.
- Facial expressions. Facial appearance is influenced by facial expressions, such as smiling, laughing, sad, talk, and etc.
- Hindered other objects. A face image can be some hindered by other objects or faces, for example, the image contains a group of people.
- Condition of image acquisition. The image obtained is strongly influenced by factors such as room light intensity, direction of light sources, and characteristics of the sensor and the camera lens.

2. *Skin Detection*

Skin detection is a process of finding the pixel or part of an image or a digital video that has a color (similar) the skin. The appearance of the skin on the image is very dependent on the lighting conditions. Humans are very good in identifying the color of the object in good lighting. But not necessarily with a computer, a new problem is how to detect the appearance of the skin with a variety of skin colors and different lighting conditions. Another challenge is the number of objects that have similar color to the skin such as wood, sand, hair and even clothing that have the same color skin. This makes difficult the skin detection. These problems can be minimized if the skin detection method is most suitable by analyze the results of any existing methods, to can be conclude that the best method

The skin detection to filter content on the internet that image with image processing and deciding the feasibility of that image to be displayed after taking into account other considerations. If you are using a skin detection method based on color, skin detector must have a data form a variety of shades of color in various lighting conditions that can be be used as a reference to decide the appearance of the skin. Them adaptive method of thresholds that do not require the data as a reference. Adaptive thresholds method work is to find the optimum threshold and then eliminate or remove the background.

III. METHODS

In this section, we explain our approach to separate benign images from adult images by analyzing image content, starting with the overall architecture and followed by a detailed discussion of the main components.

A. *Overview*

The data used in the form of a sample image taken from the internet as many as 30 images. This research uses MATLAB R2010a and OpenCV library version 2.2 for classifying and separating adult and benign images. In this research, Viola-Jones method is written as OpenCV library which is called by the developed program written in MATLAB R2010a. The Viola-Jones library also calls Haar library.

Based on Figure 1, image samples are the input of the developed system. Then face detection process is then carried out to determine whether each image is of a human image or not. After getting the results of face detection process, skin detection process is performed on the images to determine the human skin area. Based on the face and skin detection processes, facial features and skin regions are extracted and then inserted into a classifier. Finally, the system determines whether each image is of adult images or benign. The overall system flowchart of this research can be seen in Figure 1.

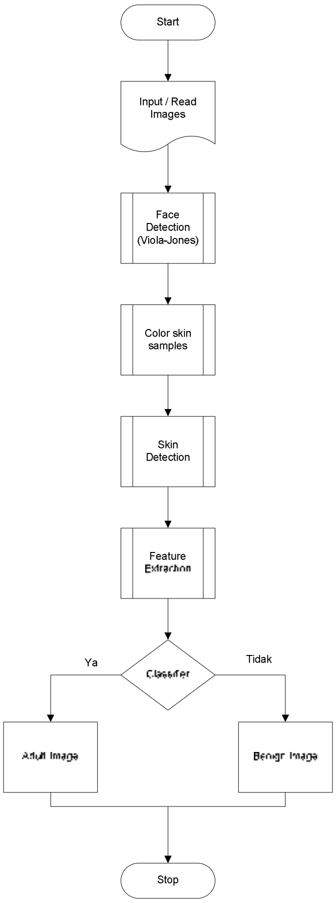

Fig. 1. Flowchart of research

B. *Viola-Jones Face Detection*

Face detection plays an important role in image classification in this research. Face detection process is carried out before the process of skin color detection. The face detection system uses Viola-Jones method, which has the advantages of fast and efficient in getting the facial region compared to previous methods of face detection [Viola et al, 2004]. The process is done by reading each sample image in JPEG using MATLAB R2010a. The image is then fetched into Haar feature classification. The classification read Haar features written in XML and converts the XML file into Matlab file. After the conversion process, a Matlab file named **haarcascade_frontalface_alt** serves as a process to call and determine Haar features in each image to detect face. Haar features are the features used in Viola-Jones method which can also be called features of a square wave. To two dimensions is referred to as one bright and one dark. The existence of Haar features is determined by subtracting the average of pixels in dark areas of an average of pixels in the bright areas. If the value of the difference is above the threshold value, it can be said that the feature is there. Next to determine whether or not of hundreds of Haar features in an image at different scales and efficiently by using the Integral Image. Generally, the integration is to add small units simultaneously. The small units are pixel values. Integral value for each pixel is the sum of all the pixels from top to bottom. Starting from top left to bottom right, the whole image can be summed with some integer operations per pixel.

Then to select the specific Haar features and to adjust the threshold value, we used a machine learning AdaBoost. AdaBoost combines many of weak classifier to make a strong classifier. by combining some of the AdaBoost classifier as an efficient filter circuit to characterize the image. The next stage of the cascade classifier. Filter in the cascade sequence is determined by the weight given AdaBoost. Filter with the greatest weight is placed on the first time, to erase the not face image as fast. During the classifying process, if there is one filter fails to pass an image area, the area it directly classified as non-face. But when the filter is missing an image area up through all the filters, then the area is classified as a facial image. The last stage of the display object has been detected on the face or not face, with a square mark if the object is considered as the face.

C. *Skin Detection*

Skin detection took part in this research. Because of a feature in determining the classification of benign and adult image, there are elements of skin detection in the image area. The first step skin detection is to read the facial skin samples based on result face detection process in previously by MATLAB R2010a. Next the skin sample RGB images are converted to form the Hue and Saturation (HSV). From the HSV value is then separated between Hue and the Saturated value. So we get the values of the color distribution in the form of HS, which can then be plotted on the 3D and 2D graphics. Threshold value using to determine the skin area on an image from the distribution of the skin of HS. The threshold values of skin color in this research refers to previous research [Still.J, 2007] who has determined in HSV color value is 0-11 for the Hue value and 20-70 for the Saturation value. the scale of each Hue and Saturation value is 0-100. Then do the skin detection process in whole part image based on threshold values of Hue and Saturation.

In the sample image will do the skin detection process, the first step to do is read the sample image by MATLAB R2010a. Then convert the RGB values image into HSV values. After getting the value of HSV, separated between the Hue and Saturation value of the image. The next step is to determine skin area by only selecting values within the rectangle skin range.

$$skin=(s>s\_range(1))\&(s<s\_range(2))\&(h>h\_range(1))\&(h<h\_range(2)) \quad (1)$$

Based on Equation 1, to determine the skin skin in an image is a combination of the S (saturation) value at the sample image is greater than the first threshold from saturation value and S (Saturation) value at the sample image is smaller than the second threshold from saturation value.

The H (Hue) value at the sample image is greater than the first threshold value from Hue value and the H (Hue) value at the sample image is smaller than the second threshold from Hue value. Then the results of skin detection is displayed in the form of a binary image, a binary 1 for the skin area and binary 0 for non-skin area.

D. *Feature Classification*

Face detection is beginning process of the whole method of this research, because the face is the most distinctive part of humans body. Detection of faces can allow the observer to form a hypothesis about the presence of humans in the scene and other measures can be taken to verify whether or not they are nudes. Detection of skin area in image also affect the decision to determine the adult image and benign image.

We empirically choose three features for benign or adult image classification. These features are all related to face:

1. Face area: the area of face regions. Images in which face regions cover too much area may be full-face portraits. We define K1 in Equation 2:

$$K1 = \frac{number\ of\ pixel\ in\ face\ region}{number\ of\ pixel\ in\ whole\ image} \tag{2}$$

2. Face position: This feature classifies the image that displays in the middle of the face in general is benign image. We defined the value of x1 and x2 is the threshold distance of the center of image pixel x. The value of y1 and y2 is the distance threshold y pixel image center, then Equation 3 can be defined as follows:

$$x1 < Distance\ center\ face\ x < x2, \quad y1 < Distance\ center\ face\ y < y2 \tag{3}$$

Equation 3 shows the face position is in the threshold area the center of image.

3. Comparison of skin: classification feature is the comparison between the number of skin area the image and the face area. So that in Equation 4 can be defined as the K3:

$$K3 = \frac{Number\ of\ Pixel\ Skin\ on\ the\ image}{number\ of\ pixel\ in\ face\ region} \tag{4}$$

Based on three features, can be made whole in the form of decision tree classification. Figure 2 shows the classification tree shape composed of nodes. Node in the tree involves testing each feature and compare with a constant threshold value. At first go through the first classification feature is K1, if K1 is greater than the threshold value (T1) it was decided as a benign image, and if the result is smaller than the threshold value (T1) then passed to the classification second feature of K2. On the K2 if the center of the face is in the center of image area threshold (<T2) it was decided as a benign image. But if it is outside the threshold region (> T2) then continued to the classification feature of K3. This classification feature is a last classification decisions as adult or benign image. If the classification of K3 is smaller than the threshold value (T3) it was decided as a benign image, and if otherwise is classified as an adult image.

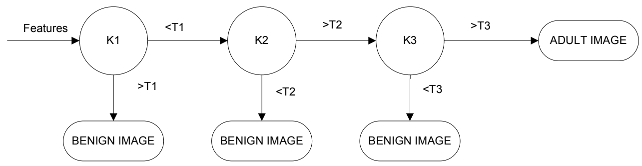

Fig. 2. Classification scheme for determining the image adult

IV. RESULTS AND DISCUSSION

In the classification process adult images and benign images, face detection is performed before the skin detection process on the image area. This section show the results of testing of face detection using the Viola-Jones method. The following are faces images that have been detected as a face.

Fig. 3. The results of face detection

The whole image is detected an face image. The first stage of classification system is adult image, it requires face detection on first stage, to determine whether there is a human image or not. If there is no face in that image, so not considered a human image, so the image is said to be the direct a benign image. Face detection process is the initial determination of the image adult or benign image. Image of the sample in general is image with upright or near-frontal face because some image are not upright or not frontal face can not be identified face in the image. Therefore, upright position of the face / frontal or not, very influence the face detection process.

The skin detection process becomes a role in the classification system of adult image and benign image. Skin detection is a factor in the classification of features that will determine the image adult or benign image. Image samples detected the existence of skin color on the overall image by giving binary classification of the image, and change to the binary color. Skin color samples obtained from the previous face detection process produces color distributions in the form of Hue and Saturation. The whole image sample performed of skin detection for the classification of adult or benign image. Here are the results of the skin detection process.

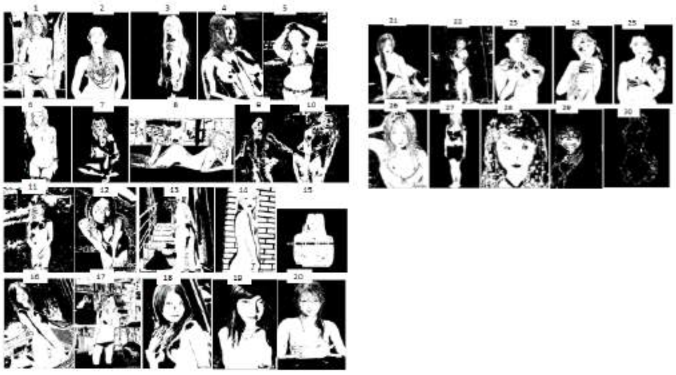

Fig. 4. The results of skin detection in all areas of the image sample.

The results of skin detection in this research are shown in Figure 4. Skin detection performed on the whole sample image, the end result of skin detectin to change the sample image into a binary image, the skin color is assumed to be logic 1 and the non-skin color assumed a logic 0. At the beginning of the process of detection of the skin, change the pixel RGB (Red, Green and Blue) value the sample image into the HSV (Hue Saturation Value). Then separated between the Hue and Saturation value image. Next obtained the distribution of color images, which show in graphic form in 2D and 3D. Here is a 3D graphic color distribution on the sample 1.

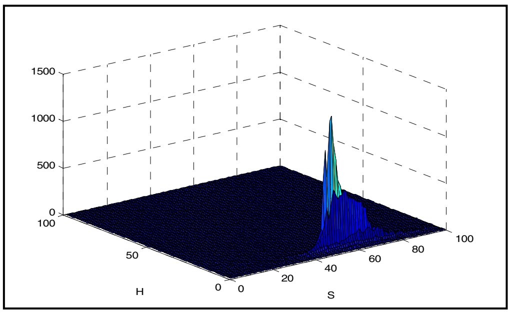

Fig. 5. 3D graphic color distribution on the sample 1.

Based on the 3D color distribution, it can be shown also colors distribution in the form of 2D graphs with maximum scale of 100 to the Hue and Saturation values.

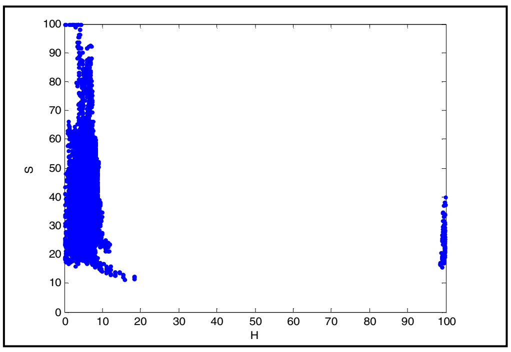

Fig. 6. 2D graphic color distribution on the sample 1.

Based on Figure 5 and Figure 6, the color distribution can be seen from sample 1, has a variety of colors (in HSV). The spread of colors in this image, dominates the value of 0-20 for the Hue and value of 0-100 for the saturation. Based on the spread of the distribution of these colors, then determined the threshold value to specify the skin color. Color range for the skin detection on the whole image is set based on the color distribution obtained from a sample image of the face detection. Based on the research [Still.J, 2007], skin color on skin detection process takes a Hue value of 0-11 and the saturation value of 20-70 as the threshold value is considered the color of human skin.

The classification used in determining adult or benign image will be based on the percentage of face area in the image, the position of face in the image and the percentage of the skin color in the image. With each of these classifications have value threshold. Each classification has a supporting components among which are size of the image, size of skin pixel image,size of the pixel face and the distance to the center of the face image. So it can be combined in the overall classification feature with a sequence K1-K2-K3 to determine adult images and benign images. Here is the result of cascade classification in this research.

TABLE I. Classification results

No. Image	Type of Image	Classify K1	K2	K3	Decision
1	Adult	0.0174	> T2	38.1941	Adult image
2	Adult	0.1539	> T2	2.6988	Adult image
3	Adult	0.0377	> T2	6.0469	Adult image
4	Adult	0.1069	> T2	3.8318	Adult image
5	Adult	0.0534	> T2	7.0806	Adult image
6	Adult	0.0440	> T2	7.4390	Adult image
7	Benign	0.0368	> T2	4.4194	Adult image
8	Adult	0.0174	> T2	32.1055	Adult image
9	Adult	0.0372	> T2	4.2919	Adult image
10	Adult	0.0809	> T2	3.9610	Adult image
11	Adult	0.0200	> T2	16.8464	Adult image
12	Adult	0.1566	> T2	2.4969	Adult image
13	Adult	0.0151	> T2	29.8048	Adult image
14	Adult	0.0737	> T2	9.1284	Adult image
15	Adult	0.0793	> T2	4.4476	Adult image
16	Adult	0.0376	> T2	14.1911	Adult image
17	Adult	0.0247	> T2	22.1199	Adult image
18	Adult	0.1623	> T2	3.2340	Adult image
19	Adult	0.2216	> T2	1.9443	Benign Image
20	Adult	0.1107	> T2	3.9555	Adult image
21	Adult	0.0373	> T2	9.0514	Adult image
22	Adult	0.0188	> T2	10.4988	Adult image
23	Adult	0.1082	> T2	4.3571	Adult image
24	Adult	0.1539	> T2	2.7687	Adult image
25	Adult	0.1539	> T2	2.9751	Adult image
26	Benign	0.2226	< T2	2.8631	Benign Image
27	Benign	0.0226	> T2	8.0918	Adult image
28	Benign	0.5493	< T2	0.8902	Benign Image
29	Benign	0.1963	< T2	0.4960	Benign Image
30	Benign	0.2538	> T2	0.0950	Benign Image

TABLE I which is a result of combination three classifications For each feature, the threshold value is defined in this research. The results of are 5 benign images and 25 adult images. False positive rate are 2 images and false negative rate is 1 image With the percentage of error classification system are the difference between the total number of sample images and the number of correctly classified images which compared with the total number of sample images are 10%. With the percentage accuracy of classification system are 90%.

V. CONCLUSION

This research designing a classification system to determine adult images and benign images. The face detection method in this study using the Viola-Jones, which has advantages in quickly and efficiently compare other face detection methods. The weakness of the face detection system with this method are not able to determine the face not upright or not frontal face. The upright position of the face or not upright determine the success of face detection. Face and skin detection plays a dominant role in this method, where the results of the face and skin detection is a key element for classifiying the images an adult or not. The classification used in determining adult or benign image will be based on the percentage of face area in the image, the position of face in the image and the percentage of the skin color in the image. For each feature, the threshold value is defined in this research are T1 = 0:17, T2 = 30% of the center's image and T3 = 1. Percentage of accuracy system are 90%.

REFERENCES

[1] Chan.Y, Harvey. R, dan Smith. D, "Building systems to block pornography" Challenge of Image Retrieval, Newcastle, 1999.

[2] Viola, P., Jones, M. J.: Rapid Object Detection Using A Boosted Cascade of Simple Features. IEEE Conference on Computer Vision and Pattern Recognition, Jauai, Hawaii, 2001.

[3] Y. Wang and B. Yuan, "A novel approach for human face detection from color images under complex background," in Pattern Recognition, pp. 1983–1992, 2001.

[4] Ries Christian, and Lienhart. R , "A Survey On Visual and Adult Image Recognition" Multimedia Tools and Aplication. April 2014. Vol 69. Issue 3, pp 661-688, 2014.

[5] Wijaya. I. G. P. S , "Phonographic image recognition using fusion of scale invariant descriptor" Frontiers of Computer Vision (FCV) Korea-Japan. Page 1-5, 2015.

[6] Lin. Y. C , Tseng. H. W, and Fuh. C. S , "Pornography Detection Using Support Vector Machine" Conference on Computer Vision, Graphics dan Image Processing (CVGIP 2003), 2003.

[7] Viola, P., Jones, M. J.: Robust Real-Time Face Detection. International Journal of Computer Vision, Kluwer Academic, Netherlands, 2004.

[8] Zheng. Q. F, Zhang. M. J, and Wang . W. Q, " A Hybrid Approach to Detect Adult Web Images," Springer-Verlag Berlin Heidelberg. 2004.

[9] Zheng. H, Daoudi, and Jedynak. B, "Blocking Adult Images Based on Statistical Skin Detection" Universitat Aut_onoma de Barcelona, Barcelona, Spain, 2004.

[10] Nugroho, S., "Sistem Pendeteksi Wajah Manusia pada Citra Digital" , Tesis Program Studi Ilmu Komputer Jurusan MIPA, Universitas Gadjah Mada, Yogyakarta, 2004.

[11] Ho. W. H and Watters. P. A," Identifying dan Blocking Pornographic Content" International Conference on Data Engineering (ICDE '05). 2005.

[12] Wang. Y, Wang.W, and Gao. W, "Research on the Discrimination of Pornographic dan Bikini Images" Seventh IEEE International Symposium on Multimedia (ISM'05), 2005.

[13] Still, J. D., Dohse, T., and Dohse K. C. "Touch table augmented with skin detection". Poster presented at the Emerging Technologies Conference, Ames, IA. April, 2007.

[14] Hu. W, Wu. O, Chen. Z, Fu. Z, and Maybank. S, "Recognition of Pornographic Web Pages by Classifying Texts dan Images" IEEE transactions on pattern analysis dan machine intelligence, vol. 29, no. 6, June, 2007.

[15] Agbinya. J. I, Lok. B, Wong. Y. S and Da Silva. S, "Automatic Online Porn Detection dan Tracking" Faculty of Engineering, University of Technology, Sydney, 1 Broadway, Sydney, 2007.

[16] Dwiyana, "Deteksi Kulit sebagai Penyaring Konten Gambar di Internet", Skripsi Jurusan Ilmu Komputer Universitas Pendidikan Indonesia, 2009.

[17] Andoko, " Perancangan program simulasi deteksi wajah dengan Support vector Machines – Viola Jones", Teknik Informatika – Matematika, Universitas Bina Nusantara, Jakarta, 2010.

[18] A Arihutomo, M., "Rancang Bangun Sistem Penjejakan Objek Menggunakan Metode Viola-Jones untuk Apiliasi Eyebot", Proyek Ahir Jurusan Telekomunikasi PENS-ITS, Surabaya, 2010.

[19] Pusat Bahasa Departemen Pendidikan Nasional, "Kamus besar bahasa Indonesia", Jakarta, 2008

Performance Comparison of Reactive Routing Protocols in Mobile Ad-Hoc Network using NS2 Network Simulator

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***Abstract*— Along with the rapid development of information and communication technology, there is need for network that can connect people without depending on fixed infrastructure. This type of network is called mobile ad hoc network (MANET). The main classes of MANET routing protocols are Proactive, Reactive and Hybrid. In this paper we compare performance of Reactive Routing Protocol by focusing on Ad Hoc On-Demand Distance Vector (AODV) and Dynamic Source Routing (DSR). In this paper our simulation tool will be NS2 Network Simulator. The performance of these routing protocols is analyzed by three metrics: delay, routing overhead and throughput, by varying the number of nodes and speed of nodes. The results show that AODV outperforms DSR in more stress situations, however DSR outperforms AODV in less stressful situations (smaller number of nodes and lower speed of nodes).**

Keywords*— *MANET, AODV, DSR, NS2, reactive routing protocol

I. Introduction

Along with the rapid development of information and communication technology, there is need for network that can connect people without depending on fixed infrastructure. This type of network is called mobile ad hoc network (MANET). In infrastructure or ad hoc network each node is connected through wireless links. The main reason to deploy this kind of network is the flexibility and easiness of deployment. In MANET, the mobile nodes not only can receive and forward packets but also can act as a router, that’s why its need an efficient and reliable routing protocol to do those tasks. Unlike fixed-infrastructure network, MANET has a different topology change while deployed that’s why it needs different kinds of routing protocol. By comparing the performance of protocols, will give the overall performance of each routing protocol. The main problem is to choose the reliable, efficient and correct routing protocol for MANET in different network conditions. This thesis presents the simulation results and performance analysis of reactive routing protocol Dynamic Source Routing (DSR) and Ad Hoc On-demand distance Vector Routing (AODV). The analysis is based on packet delivery ratio, routing overhead, average delay, and average throughput by varying the number of nodes and speed of nodes. The simulation is performed using NS2 network simulator.

II. MANETs ROUTING PROTOCOL

An ad-hoc routing protocol is a convention, or standard, that it improves the scalability of wireless networks compared to infrastructure based wireless networks because of its decentralized nature. Ad-hoc networks are best suited due to minimal configuration and quick operation.

Reactive routing protocol is also known as on-demand routing protocol. It takes a different approach of routing which overcomes the disadvantages of proactive routing. In reactive approaches those nodes which require connectivity to the Internet reactively find Internet gateways by means of broadcasting some kind of solicitation within the entire ad hoc network. This approach reduces the overhead of maintaining the route table as that of proactive. The node dynamically checks the route table, and if it does not find an entry for its destination or it finds an outdated entry it performs route discovery to find the path to its destination [1].

A. Dynamic Source Routing (DSR)

Dynamic Source Routing (DSR) is a reactive routing protocol that uses source routing to send packets. It uses source routing which means that the source must know the complete hop sequence to the destination. Each node maintains a route cache, where all routes it knows are stored. The route discovery process is initiated only if the desired route cannot be found in the route cache.

The protocol is composed of the two main mechanisms of "Route Discovery" and "Route Maintenance", which work together to allow nodes to discover and maintain routes to arbitrary destinations in the ad hoc network. All aspects of the protocol operate entirely on demand, allowing the routing packet overhead of DSR to scale automatically to only what is needed to react to changes in the routes currently in use. The protocol allows multiple routes to any destination and allows each sender to select and control the routes used in routing its packets, for example, for use in load balancing or for increased robustness [1].

B. Ad-Hoc On-Demand Distance Vector Routing (AODV)

AODV routing protocol is an on-demand reactive routing protocol that uses routing tables with one entry per destination. When a source node needs to find a route to a destination, it starts a route discovery process, based on flooding, to locate the destination node. Upon receiving a route request (RREQ) packet, intermediate nodes update their routing tables for a reverse route to the source. Similarly, the forward route to the destination is updated upon reception of a route reply (RREP) packet originated either by the destination itself or any other intermediate node that has a current route to the destination [2].

The AODV protocol uses sequence numbers to determine the timeliness of each packet and to prevent the creation of loops. Expiry timers are used to keep the route entries updated. Link failures are propagated by a route error (RERR) message from a broken link to the source node of the corresponding route. When the next hop link breaks, RERR packets are sent by the starting node of the link to a set of neighboring nodes that communicate over the broken link with the destination. This recursive process erases all broken entries from the routing table at each node. Since nodes reply to the first arriving RREQ packet, AODV protocol favors the least congested route instead of the shortest route. AODV protocol minimizes routing table information potentially leads to a large number of route requests being generated.

III. SIMULATION ENVIRONMENT

A. Simulation Model

Simulation model is done by considering the performance metrics of routes (routing protocol QoS) within mobility model and traffic pattern. The simulation parameters for this thesis are described in Table 1.

TABLE 1 SIMULATION PARAMETER

Parameter	Nilai
Transmission range	250 m
Simulation time	90 s
Simulation area	800 x 800 (m)
Ad-hoc node	10,20,30,40,50
Source node	1
Traffic type	Constant bit rate/ UDP
Data rate	11 Mbps (802.11b)
Packet rate	5 packet/second
Packet Size	512, 1024 bytes
Pause time	5 s
Maximum speed	5, 10, 15, 20, dan 25m/s
Routing Protocol	AODV, DSR

We carried out simulations on NS2 network simulator [3].

B. Performance Metrics

Three performance metrics will be used to determine the QoS of reactive routing protocols. A detailed explanation of these metrics is as follows:

- Routing overhead is the total number of routing packets (route request, route reply, route error) divided by total number of delivered data [4]. For each packet that delivered through multi-hop, count as a transmission.
- The end to end delay is defined as the time a data packet is received by the destination minus the time the data packet is generated by the source
- The average network throughput determines the amount of data that is transmitted from a source to a destination node per unit time (byte per second). Using the Template

IV. RESULTS

A. Routing Overhead

Routing Overhead is the total number of routing packets. The performance analysis graphs for DSR and AODV are as follows. Figure 1 shows routing overhead v/s number of nodes for packet size 1024 bytes and 512 bytes, and Figure 2 shows routing overhead v/s maximum speed of nodes for packet size 1024 bytes and 512 bytes. For each scenario by varying the number of nodes, AODV routing overhead is lower than DSR. DSR generates more routing load than AODV under stressful situation because it maintains multiple route cache entries for each destination. AODV on the other hand uses traditional routing tables, one entry per destination for maintaining routing information. When the sources become more, DSR with greater amount of routing information began to perform worse than AODV.

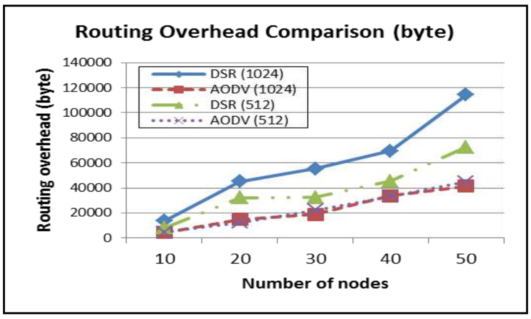

Fig. 1. Routing overhead comparison by number of nodes .

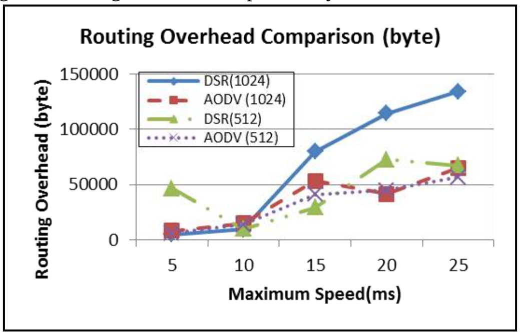

Fig. 2. Routing overhead comparison by maximum speed of nodes

B. Delay

Average delay is the calculation of typical time taken by packet to cover its journey from the source end to the destination end. As route failure occurs, it takes longer for a data packet to reach its desired destination resulting in higher end-to-end delay. In all scenarios, DSR have significant greater amount of average delay than AODV. In scenarios by varying the number of nodes when packet size 1024 byte, DSR achieves average amount of delay between 0,05 to 0,12 seconds, while AODV achieves 0,009 to 0,017 seconds. When packet size 512 byte, the average delay of DSR are between 0,02 to 0,11 seconds, while AODV are between 0,009 to 0,022 seconds. As the size of the packets increased, the time needed to sending packets (delay time) has also been increased.

In scenarios by varying the maximum speed of nodes when packet size 1024 byte, DSR achieves average amount of delay between 0,06 to 0,13 seconds, while AODV achieves 0,008 to 0,021 seconds. When packet size 512 byte, the average delay of DSR are between 0,06 to 0,21 seconds, while AODV are between 0,008 to 0,03 seconds. As the increase increase of maximum speed of nodes, delay time of DSR began to perform worse than AODV.

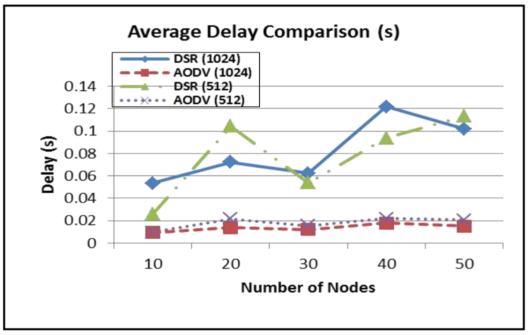

Fig. 3. Average delay comparison by number of nodes

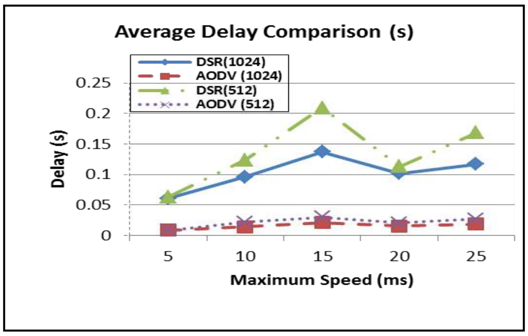

Fig. 4. Average delay comparison by maximum speed of nodes

C. Throughput

The average network throughput determines the amount of data that is transmitted from a source to a destination node per unit time (byte per second). As we can see, DSR outperforms AODV in less stressful situations (less number of nodes and lower speed of nodes). AODV, however, outperforms DSR in more stressful situations. The result shows that AODV can achieve an average throughput 272824,5 bps in scenarios with packet size is 1024 bytes, while 253453,8 bps. When packet size 512 bytes, the average throughput of AODV are 163440,9 bps and DSR are 156677,5238 bps. The results show that as the size of packets increased 100% (512 to 1024 bytes), throughput has been increased for each protocol.

In scenarios by varying the maximum speed of nodes when packet size 1024 byte, the average throughput of AODV are 274989,38 bps and DSR are 258505,76 bps. When packet size 512 bytes, the average throughput of AODV are 164485,94 bps and DSR are 162174,27 bps. The average throughput for both protocols are decreased as the maximum speed of nodes increased.

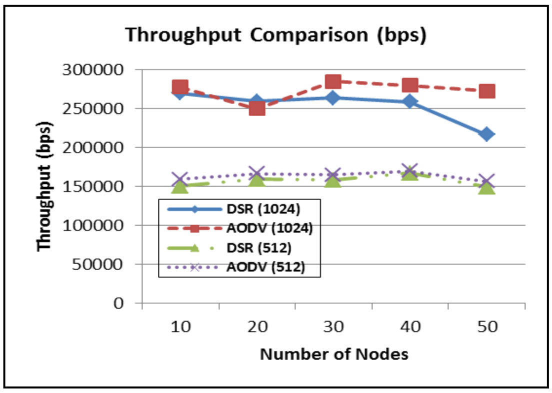

Fig. 5. Throughput Comparison by number of nodes

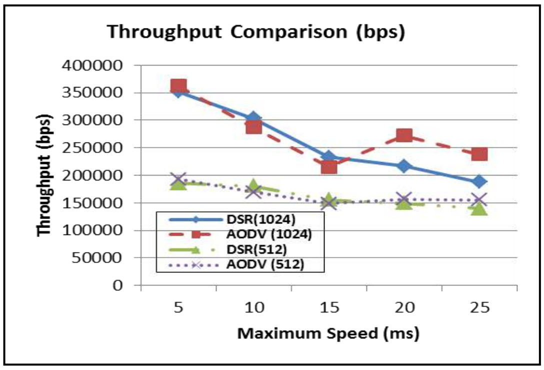

Fig. 6. Throughput comparison by maximum speed of nodes

V. RESULTS

The results show that AODV outperforms DSR in more stress situations because in DSR, the chances of find the route in one of the caches is much higher. However DSR outperforms AODV in less stressful situations (smaller number of nodes and lower speed of nodes). DSR produces higher delay in all scenarios, therefore this protocol not recommended for MANET with the need for low delay.

REFERENCES

[1] Rath, S. (2009). "Study of Performance of Routing Protocols for Mobile Adhoc Networking in NS2". National Institute of Technology, Rourkela

[2] Sari, R.; Syarif, A; Budiardjo.B. (2008), "Analisis Kinerja Protokol Routing Ad Hoc On-Demand Distance Vector (AODV) pada Jaringan Ad Hoc Hybrid: Perbandingan Hasil Simulasi Dengan NS2 dan Implementasi pada Testbed dengan PDA, " MAKARA, TEKNOLOGI VOL.12, NO. 1, hal 7-18.

[3] Anon. (2005). "The Network Simulator-NS2", http://www.isi.edu/nsnam/ns. Diakses tanggal 5 Juli 2011.

[4] Al-Maashri, A; Ould-Khaoua, M. (2006). "Performance analysis of MANET routing protocols in the presence of self-similar traffic", Proceedings of the 31st IEEE Conference on Local Computer Networks, 2006, 14-16 November 2006, pages pp. 801-807, Tampa, Florida, USA.

A Joint Approach to Multipath Routing and Rate Adaptation for Congestion Control in OpenFlow Software Defined Network

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***Abstract*—Software Defined Network is a new paradigm in data communication networks. In contrast to the traditional internet network, SDN separates the control function and the function forwarding. SDN controller operates on the principle of a centralized logical controller. This centralized control causes the controller to have a global network knowledge so that SDN can be more effective in managing the network. In this paper, we apply a congestion control mechanism by utilizing the global knowledge possessed by the controller. Congestion control mechanisms implemented by combining multipath routing mechanisms and rate adaptation in an integrated manner. The controller uses the information from the switches that are used to determine the route along the source and destination, as well as to determine the appropriate rate for each flow. The experimental results show that the combination of routing and rate adaptation can effectively be used to control congestion, resulting in a better performance than single path routing and multipath routing without rate adaptation.**

Keywords—software defined network; multipath routing; rate adaptation

I. Introduction

SDN is a new paradigm in network technology. Unlike traditional internet network that combines control plane and data plane in the same node, SDN separates the control plane and data plane. This separation makes the network policy can be done by software in the control plane. Control plane consists of one or more controllers are connected to the switches in the data plane. Separation of these functions makes the data plane on SDN is only a collection of dumb devices [1]. On SDN, the communication between the control plane and the data plane using a protocol. The protocol was first used is OpenFlow [2] and this protocol is open.While not everyone believes that OpenFlow is already mature enough to support future internet and resolve the real problems the network, but until now OpenFlow can be used to solve the challenges of the future internet experimentally. OpenFlow approach provides easy access to the flow table, providing real-time control via network switches and allows administrators to monitor and control the route packets flowing through the network [3].

National Science Foundation predicts that the Internet will have nearly 5 billion users by 2020. Other data mentions that the user Internet of thing (IOT) reached 7,065 billion. A very large number of customers have become one of the serious problems in the future internet [4]. The future internet network requires the availability of sufficient network resources and network management is good, otherwise it will cause network congestion. The adverse effects of network congestion are the decline in performance, such as high delay, low throughput, and packet loss [5]. This can reduce customer satisfaction and will harm the network and Internet service providers. Therefore, congestion must be controlled to maintain network performance remains excellent.

Network congestion problems begin to be solved since 1988 by Van Jacobson with the proposed TCP [6]. TCP uses the settings window (sending window) to adjust the sending rate. Traditional window-based congestion control does not work well when dealing with bursty traffic, resulting in additional queuing delay and packet loss. Besides that TCP has some limitations and drawbacks that are prone to instability, it is not fair to the connection with the high round-trip delay and low resource utilization. Already many of the proposed scheme for TCP improvements, among others set forth in [7] [8] [9] [10] [11]. Proposed improvements TCP has also been done on SDN, among others OpenTCP [12], CCC (Cooperating Congestion Control) [13] and in [14]. The proposed improvement is shown to improve network performance, but congestion control that is not supported by either routing mechanism making it ineffective.

On the other hand, there are some ideas to solve the network congestion by controlling routing. Routing mechanisms on the Internet network is largely based on a single path routing. However, routing with a single trajectory contributing to congestion on the network due to the possibility of some of the same links used as the shortest path from a different node pairs that causes convergence to link the same package [15], [16]. Improvements single routing path is the multiple path routing techniques (multipath routing) that are used as a method to reduce network congestion. The basic idea

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of multiple routing paths is to divide the package with the same origin and destination for some of the alternative pathway. The use of multipath routing proven to improve network resource utilization [17].

The use of the routing mechanism and a separate delivery rate setting is not effective enough to prevent network congestion. Combining multipath routing by setting the rate of delivery can be effective for congestion control and can improve performance better than if applied separately. Routing determines the path to be used while congestion control determines what rate to be sent through any predetermined path.

On SDN, routing and sending rate settings can be fully controlled by the controller. The controller has a global network of knowledge about the condition periodically based on information reported by the switch to decide the action to be taken. With these advantages, it is possible to develop a scheme of a combination of routing and rate adaptation to control congestion on OpenFlow [3]. By using the advantages offered by SDN OpenFlow, we applied a combination of multipath routing scheme and setting the pace of delivery with the aim to control network congestion and improve network performance. To achieve these objectives, we have proposed a design that contains components that function as the search for the best path and set the rate to be allocated on each path. In fact, OpenFlow 1.3 [18] already has support for (ECMP), but Al-Fares et al. In his paper [19] found the collision on ECMP that occur in the long term can cause a lot of lost bandwidth.

The rest of the paper is structured as follows. Section I. Introduces, section II Related work, section III Design, in section IV Evaluation and the last in Section V Conclusions and future work.

II. RELATED WORK

Some literature have discussed multipath path routing of different purposes, e.g. [20], [21], [22], [23] explains how to find a loop-free multipath routing. Two others, such as [24], [25], discussing the multipath routing optimization problem, finding the optimal way to pass traffic with high throughput and low congestion. Routing path as QoS multipath routing discussed in [16], [26], [27], [28], [29] wherein multipath routing used to meet the QoS guarantees. Multipath routing as load balancing methods are discussed in [30], [31], [32]. Likewise, within the scope of SDN, multipath routing has been discussed in [33], [34].

Multipath routing schemes that have been proposed as a whole can improve network performance. Multipath routing produces a better performance than the single path routing scheme. However, the schemes are still not effective. The effectiveness of a multipath routing scheme can be improved by combining multipath routing and rate adaptation of the predetermined path of the routing algorithm. Merging multipath routing by setting the sending rate for optimization purposes proposed by Fern Y. Hunt [35], Roberto Commineti [36], and Jiayue [37]. Most of the proposed method is done for traditional internet platform. Nodes in the network exchange information with each other and not doing congestion centrally. This system has a significant delay constraint resulting feedback to the adaptation rate to be expired. At the data center, the design of multipath algorithms has been proposed by Al Fares in [19], Heller [38] and Benson [39]. On the other hand, in [40] [41], the problem of multi-path QoS can be formulated as a Multi-Commodity Flow (MCF). Combining routing and rate adaptation on SDN proposed in [14]. They combine routing with explicit feedback congestion control. The proposed scheme generates flow average completion time (AFCT) is small. However, this scheme uses a single path routing and does not provide an alternative path in the event of overload.

III. DESIGN

Before discussion of the combination of multipath routing mechanisms and rate adaptation, it is helpful to consider the mechanism of how the flow to be processed in SDN OpenFlow. When the first packet of a new flow arrives at the switch, the switch will see the flow table to determine how the packet should be processed. If the package is not found in the existing rules, the switch, then encapsulates the packet to the Packet-inclusive message and sends the message to the controller. After receiving a message packet-In, OpenFlow controller determines how the flow should be handled. Handling flow based on policy settings is set and the network conditions globally. If this flow is permitted, the controller calculates the flow path and install a new entry corresponding flow switch along the path. The controller may flow modification command to switch on each switch along the route or may also send a message Packet-Out that explicitly tells the switch where to forward the packet first.

The scenario of a combination of multipath routing and rate adaptation in this paper begins when there is a request from the source node to the destination node. OpenFlow switches will be reported to the controller about the state of the link. The controller will collect and calculates the network conditions. The controller will determine all possible paths that will be used to pass flow from source to destination. The controller also calculates the rate that can be allocated to each flow is passed. The combination of multipath routing mechanism and the adaptation rate on OpenFlow SDN that we propose as shown in figure 1.

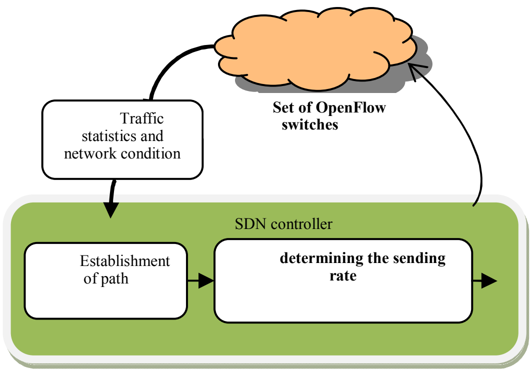

Figure 1: The mechanism of combination of multipath routing and rate adaptation

In general, the steps are as follows:

- Step 1: controller collects information on network conditions
- Step 2: controller finds the best path and several alternative paths from source to destination
- Step 3: the controller calculates rate that can be allocated in accordance with the utilization of a path
- Step 4: the controller ordered the switch to pass the traffic through the best path
- Step 5: if congestion marked path utilization exceeds the threshold, then the flow is distributed to an alternative route until a certain threshold

A. Network Model

Network modeled as a simple directed graph G (N, L). Where N is the node and L is the link. (i, j) is the pair where i is the outgoing node and j is the incoming node. If s is the source node and d is the destination node, then the entire route from source to destination is the entire path from the source to the destination or P_{sd} . Where a path from source to destination is expressed by equation (1).

$$P = \{s, i_1 . i_2 \ldots \ldots ., d\} \qquad (1)$$

$$P \in P_{sd}$$

B. Selection of the shortest path

In general, any routing protocol used in packet networks such as OSPF (Open Shortest Path First) is used as a link metric weight. Based on the weight of the link, routing protocols find the path end to end with the most minimal total weight for each pair of source and destination. This mechanism is called the path selection by choosing the shortest route "shortest path routing". In this paper, we use the DFS (Depth-first search) algorithm to search routes. All possible routes (paths) are stored in the flow table T and compared with each other. This comparison is to find a path with the maximum residual bandwidth. Path discovered regarded as the best path and then some other path as an alternative route.

$$p = max\ \{f_{resBW}(p) | p \in P_{sd}\} \qquad (2)$$

The capacity of a path (C_p) is the minimum capacity of a link that is a member of a path:

$$C_p = min\ \ \{C(l) | l \in p) \qquad (3)$$

So that the residual bandwidth of a path ($Res\ bw$) is the minimum capacity is reduced by a rate that is already in use.

$$Res\ bw = C_p - s(t) \qquad (4)$$

C. Distribution Traffic and Threshold Path

Traffic distribution is done when the bandwidth utility the main path through which the flow has reached a certain threshold. We use a bandwidth utility as a parameter to divert flow from the best route as well as the main route to the alternative route. Traffic was first passed to the best path and any particular period controller update path status. If the utility bandwidth on the main path reaches 0.8, the controller will divert traffic from the main road to the alternative path.

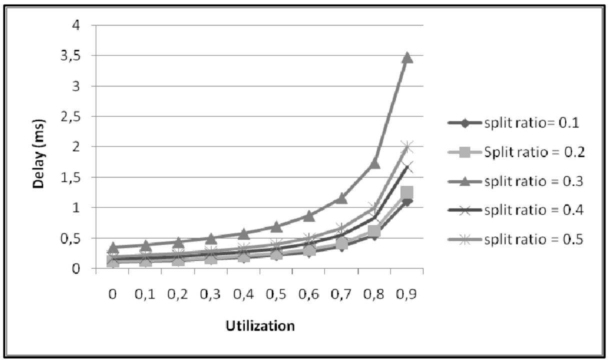

Figure 2: Relationships delay and bandwidth utilization in OpenFlow switch using the model Jackson

In this case, we use the threshold is 0.8 as the threshold for traffic diversion. Reason why the use value of 0.8, we can see in figure 2 that the delay has not increased significantly when the utility to the value of 0.4. Delay increases linearly when the utility reaches 0.4-0.8 and increase dramatically when the utility reached above 0.8. Delay increment can be used as a parameter that the path being traversed a flow is experiencing congestion. The relationship of the delay and bandwidth utilization in figure 2 is obtained from the equation 5.

$$T_s = \frac{\rho}{\lambda(1-\rho)} \qquad (5)$$

$$\rho = \frac{\frac{\lambda}{1-q}}{\mu} \qquad (6)$$

Where T is a delay, λ is the average traffic coming into the switch, ρ is bandwidth utilization and q is the split ratio (incoming packet to the controller as a new flow).

D. Rate adaptation

Rate adaptation is used to adjust the data transmission rate of the link load conditions during the routing process. Most of rate link adaptation is not based on this path as a whole. In this paper, rate adjustments made to find the maximum rate path for any paths that have been found. Rate adjustment is based on the condition of the path that has been chosen.

The rate is allocated according to the residual bandwidth is used to determine a path. The rate is updated every interval d, then for each flow that will be serviced using the rest of the path capacity.

$$R(t) = \frac{\alpha\,(Cp - s(t))}{N(t)} \quad (7)$$

So that each additional flow, rate path becomes:

$$R_p(t) = R_p(t-d) + \frac{\alpha(C - s(t))}{N(t)} \quad (8)$$

C as path capacity, $N(t)$ as a number of ongoing flow at time t, α as parameter stability, $R_p(t-d)$ as update rate and $s(t)$ as the measured rate. Rate path $R_p(t)$ is calculated by OpenFlow controller and sent to Switch OpenFlow. Switch OpenFlow then apply the instruction ordered a new controller for each flow or each flow after a certain period *d.*

E. Rerouting

If there is not enough capacity on the primary path or has exceeded a predetermined threshold, then the routing mechanism will divert traffic to another route. Flow will be diverted when the utility of the primary path reaches a value of 0.8 on a path capacity.

$$R_p(t) \;\geq\; 0{,}8\, C_p \quad (9)$$

F. Admission control

Admission control is used to restrict the incoming flow when the network is not able to serve. In this case if the flow is not allowed to enter when added to a new flow, the flow rate will fall below the desired rate of a flow. If the assumption Sending rate allocated to each flow is the same, then the flow rate is the total capacity of a path divided by the number of flows.

$$R\,(t) = \frac{Cp}{N(t)} \quad (10)$$

A flow entering the network will be rejected if it met the conditions when the flow is received, the flow rate provided is less than the required flow rate ($R\;req$).

$$\frac{Cp}{N(t)} < R\;req \quad (11)$$

G. Architecture

The system design includes the design of network topology on a data plane and application design SDN. The design of this system is implemented in mininet that allows users to use multiple topologies. This mechanism works by (1) reducing the load node, (2) traffic load balancing and (3) reroutes. To perform these tasks, the system consists of several modules as in figure 3, consists of several processes, namely: The collection of statistical data, calculation of metrics, the selection of best path, traffic distribution: and sending rate determination. We made a simple mechanism, flexible and modular. Consists of five components, namely: collector, metric calculation, rate calculation, and the last flow management.

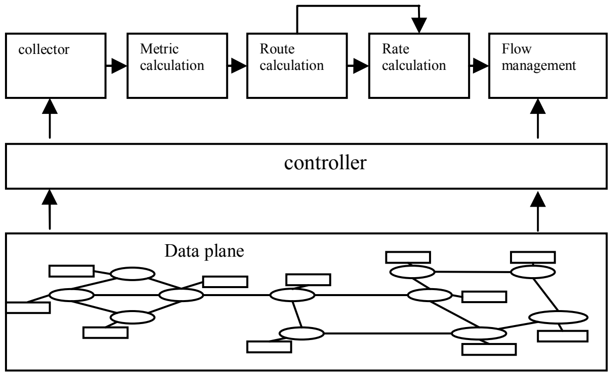

Figure 3: System design

1) Collector: collector module is used to collect information about the condition of the global network, and is responsible for storing all the information. Information can be the network topology, traffic, the condition or state of each device and port, etc.. The controller requested a variety of statistical data from the switch by sending a message FEATURE_REQUEST, and the switch sends a message FEATURE_REPLY contains statistical data requested. The mechanism is described in detail in the specification OpenFlow.

2) Metric calculation: metric calculation module is used for determining link cost parameters that will be used to determine routing and rate and save all the possible paths of each pair of switches.

3) Route calculation: Route calculation module is used to calculate the shortest path and the alternative path between source and destination. Route calculation is done when a message PACKET_IN came to the controller.

4) Rate calculation: Rate calculation module is used to calculate the sending rate. As specified in the OpenFlow specification, any flow of OpenFlow switch contains a set of instructions that run when a packet arrives appropriate entries. One kind of instruction is a meter that directs packets to a specific meter. Each meter has one or more band meter.

5) Flow Management: flow management module is responsible for determining the path of each flow and installs rule / flow entry into datapath or switch. This function is responsible for regulating the flow of efficient by distributing traffic or flow.

IV. EVALUATION

To evaluate the proposed scheme we simulate using mininet. We implement the design as an application on the Ryu OpenFlow controller. Simulations using 1 controller and 11 switches, where each switch is logically connected to the controller. Setting parameters based on network topology Abilene [42]. Our evaluation of network performance parameters such as delay, packet loss, and throughput. In this experiment, we apply a single flow and give a background traffic to the network to provide a load on the links and paths.

We observe the simulation results of several different data transfer methods. The method is a single path, multipath without rate adaptation and multipath routing with the rate adaptation. Measurements were made for the UDP data traffic on the link bandwidth of 100 Mbps and set a delay of 1 ms. We use D ITG as a source of traffic. For this scenario, UDP traffic generation, the constant inter-departure time (IDT) in rate 100 PPS and Poisson distributed packet size with = 48 bytes is used. The measurement results in the form of delay, throughput and packet loss as shown in figure 4,5 and 6.

When an incoming flow, the entire bandwidth can be allocated entirely to the flow. When there is subsequent incoming flow, the flow will be allocated a bandwidth of the residual bandwidth path as an equation (10) until when the utility of a path to reach 80%. When the utility path already exceeds 80%, the flow will be diverted to another route. Flow received will be rejected if a newly added flow rate will decrease to below the desired rate. Then the admission control will be applied if the conditions indicated in the equation (11). The simulation results show the throughput with the proposed mechanism to approach the desired value that is equal to 37.98518 kbps. Theoretically, the expected throughput is around 38.4 kbps. The average throughput of the simulation results as shown in Table 1.

TABLE I. AVERAGE THROUGHPUT

No	Data Service	
	Method	*Throughput (kbps)*
1	Single path	36,74179
2	Multipath	37,41376
3	Multipath with rate adaptation	37,98518

The whole of the experimental results showed that the combination of multipath routing methods produces better performance.

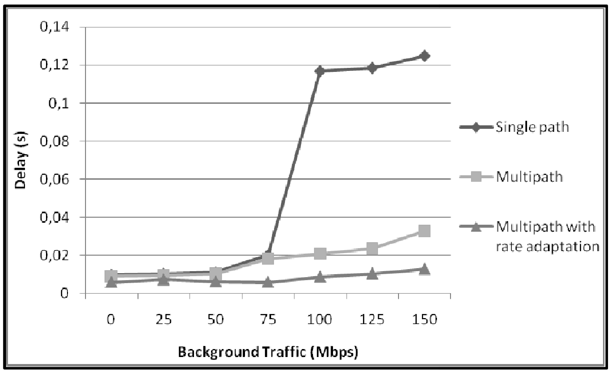

Figure 4: Delay for Data

Seen in Figure 4,5 and 6 when the network load is light experimental results show that the method of single-path, multipath and multipath levels is not too different. Network performance looks significantly better when the utility network is high. It shows that the combination of routing method and rate adaptation can be effectively used to address network congestion. Network load can be divided into various paths. Each path can be allocated a rate corresponding to the bandwidth that is still available.

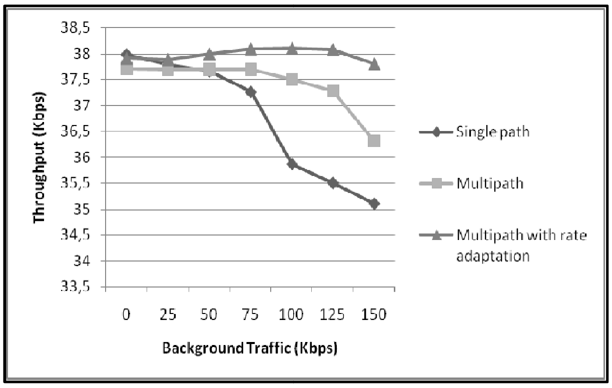

Figure 5: Throughput for Data

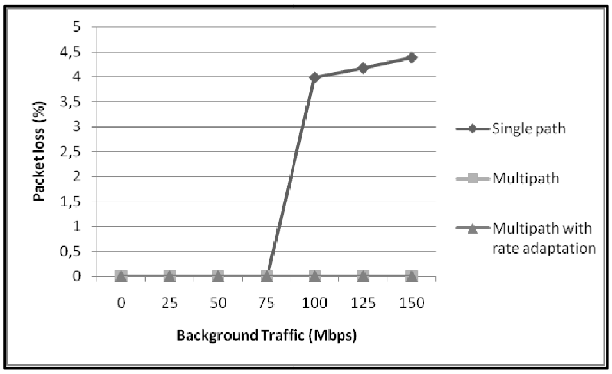

Figure 6: Packet loss for Data

V. CONCLUSION

In this paper, we apply a combination of routing mechanisms and rate adaptation for congestion control in OpenFlow-based software-defined networks. Experimental results show that the combination of multipath routing and rate adaptation produce a better performance than the single path and multipath routing without rate adaptation. On experiments, UDP data transfer at a high bandwidth utility shows that the performance was good and stable. However, the complexity of the multipath routing is higher than the single path. This is due to multipath procedure more complicated than the single path. Besides merging routing and rate adaptation require a longer time. The results listed in this paper are an initial experiment.

For future work, we will observe how much complexity, routing overhead and memory consumption needed. We will conduct experiments to transfer data TCP. We will use different metrics, involving other network parameters are more complex

REFERENCES

[1] Goth, "Software-Defined Networking could shake up more than packets," IEEE Internet Computing, Vol. 5, no. 4, pp. 6-9, July-August 2011.

[2] Lara, Kolasani, B.Ramamurthy, “Network Innovation using OpenFlow: A Survey,” IEEE Communications Surveys & Tutorials, vol.16, no.1, pp 493 – 512, August 2013.

[3] N. McKeown, T. Anderson, H. Balakrishnan, G. Parulkar, L. Peterson, J. Rexford, S. Shenker, and J. Turner, “OpenFlow: Enabling Innovation in Campus Networks,” ACM SIGCOMM Computer Communication Review, vol. 38, no.2, pp. 69-74, April 2008.

[4] Talbot. D, “The Internet is broken,” Technology Review, December 2005-January 2006, http://www.technologyreview.com/article/16356, visited in October 201.

[5] 5. George A. Rovithakis, “End to End Adaptive Congestion Control in TCP/IP Networks,” CRC Press, Print ISBN: 978-1-4398-4057-3, pp.1, 2012.

[6] Van Jacobson and Michael J. Karels, “Congestion Avoidance and Control,” ACM SIGCOMM Computer Communication Review, vol. 18, no. 4, pp. 314-329, August 1988.

[7] K. K. Ramakrishnan and S. Floyd, ”The addition of explicit congestion notification (ECN) to IP, ”Draft IETF (Internet Engineering task force), Sept 2001.

[8] S. Floyd.”TCP and Explicit Congestion Notification,” Comput. Commun. Rev., vol.24 no.5, pp.10–23, October 1994.

[9] Dina Katabi, Mark Handley, and Charlie Rohrs,” Congestion Control for High Bandwidth-Delay Product,” Proceedings of The Conference on Applications, Technologies, Architectures, and Protocols for Computer Communications, pp. 89-102, 2002.

[10] Nandita Dukkipati and Nick McKeown,”Why Flow-Completion Time is the Right Metric for Congestion Control and Why This Means We Need New Algorithms,” ACM SIGCOMM Computer Communication Review, vol. 36, no.1, pp. 59-62, January 2006.

[11] Ershad Sharifahmadian and Shahram Latifi, “A Cognitive Approach for Congestion Control in High Traffic Networks,” 21st International Conference on Systems Engineering, pp.263-266, Las Vegas, August 2011.

[12] Monia Ghobadi, Soheil Hassas Yeganeh, and Yashar Ganjali, “Rethinking End-to-End Congestion Control in Software-Defined Networks,” in proc HotNets-IX, pp. 61-66, Washington, October 2012.

[13] Lu, Yang Xiao, Haifeng Du, “OpenFlow Control for Cooperating AQM Scheme,” Signal Processing (ICSP), IEEE 10th International Conference, pp. 2560 – 2563, Beijing, October 2010.

[14] Kassa Debessay, “ A Network Congestion Control Protocol to More Quickly Finish Flows”, in IDEAL, Agustus 2013, available: http://hdl.handle.net/2142/35905, visit in December 2013.

[15] X. Yang and D. Wetherall, “Source selectable path diversity via routing deflections,” in SIGCOMM, vol.36, pp. 159–170, October 2006.

[16] S. Chen and K. Nahrstedt, “An overview of quality-of-service routing for the next generation high-speed networks: Problems and solutions,” IEEE Network Magazine, vol. 12, no. 6, pp. 64–79, November 1998.

[17] S. De and S. K. Das, “Dynamic Multipath Routing (DMPR): An Approach to Improve Resource Utilization in Networks for Real-Time Traffic,” In Proceedings of MASCOTS’01, pp.23-30, Cincinnati, OH, August 2001.

[18] OpenFlow Switch Consortium and Others. OpenFlow Switch Specification Version 1.3.0. 2012. Available online: https://www.opennetworking.org/images/stories/downloads/sdn-resources/ onf-specifications/openflow/openflow-spec-v1.3.0.pdf (accessed on 2 Apr 2014).

[19] 19. Al-Fares, M., S. Radhakrishnan, B. Raghavan, N. Huang, and A. Vahdat. 2010. “Hedera: dynamic flow scheduling for data center networks”. In Proceedings of the 7th USENIX Symposium on Networked Systems Design and Implementation, 19:1–19:15

[20] P. Narvaez and K. Y. Siu, “Efficient algorithms for multi-path link state routing,” in Proc. ISCOM, pp.320-327, Taiwan, November 1999.

[21] S. Vutukurya and J. Garcia-Luna-Aceves, “MPATH: a loop-free multipath routing algorithm,” Microprocessors and Microsystems, vol. 24, pp. 319–327, October 2000.

[22] S. Vutukury and J. Garcia-Luna-Aceves, “MDVA: A distance-vector multipath routing protocol,” in Proc. INFOCOM, pp.557-564, Anchorage, AK, April 2001.

[23] P. M´erindol, J. Pansiot, and S. Cateloin, “Improving load balancing with multipath routing,” in Proc. ICCCN, pp. 1–8, St. Thomas, US Virgin Islan, August 2008.

[24] Fortz and M. Thorup, “Internet traffic engineering by optimizing OSPF weights,” in Proc. INFOCOM, pp. 519–528, Tel Aviv, March 2000.

[25] R. Banner and A. Orda, “Multipath routing algorithms for congestion,” IEEE/ACM Transactions on Networking, vol. 15, no. 2, pp. 413–424, Apr. 2007.

[26] Z. Wang and J. Crowcroft, “Quality-of-service routing for supporting multimedia applications,” IEEE Journal on Selected Areas in Communications, vol. 14, no. 7, pp. 1228–1234, September 1996.

[27] Q. Ma and P. Steenkiste, “On path selection for traffic with bandwidth guarantees,” in Proc. ICNP, pp.191-202, Atlanta, GA, October 1997.

[28] R. Guerin, S. Kamat, A. Orda, T. Przygienda, and D. Wiliams, “QoS routing mechanisms and ospf extensions,” Global Telecommunications Conference GLOBECOM '97, IEEE, pp. 1903 – 1908, vol.3, Phoenix, AZ, 3-8 November 1997.

[29] Y. Jia, I. Nikolaidis, and P. Gburzynski, “Multiple path QoS routing,” in Proc. ICC, pp. 2583–2587, 2001.

[30] C. Villamizar, “OSPF optimized multipath (OSPF-OMP),” IETF Internet, Feb. 1999, http://tools.ietf.org/html/draft-ietf-ospf-omp-02. visit in December 2013.

[31] C. Villamizar, “MPLS optimized multipath (MPLS-OMP),” IETF Internet Draft, Feb. 1999, http://tools.ietf.org/html/draft-villamizar-mpls-omp-01, visit in December 2013.

[32] S. Nelakuditi, Z.-L. Zhang, and D. H. Du, “On selection of candidate paths for proportional routing,” Computer Networks, vol. 44, pp. 79–102, January 2004.

[33] H. Egilmez, S. Dane, K. Bagci and A. Tekalp, “OpenQoS: An OpenFlow Controller Design for Multimedia Delivery with End-to-End Quality of Service over Software-Defined Networks”, in signal and information Processing Association Annual Summit and Conference (APSIPA ASC) 2012 Asia–Pasific, pp.1-8, Hollywood, LA, December 2012.

[34] Ali Al-Shabibi, Brian Martin, “StepRoute - A MultiRoute Variant Based on CongestionIntervals,” ICN 2011: The Tenth International Conference on Networks, pp.52-56, St. Marteen, Antilles, Netherland, January 2011.

[35] Fern Y. Hunt, Vladimir M,”Dynamic Routing and Congestion Control Through Random Assignment of Routes” Proceedings of the 5th International Conference on Cybernetics and Information Technologies, Systems and Applications: CITSA 2008, pp.161-164, Orlando, July 2008.

[36] Roberto Cominetti, Cristobal Guzman, “ Network congestion control with markovian multipath routing,” NETGCOOP, pp. 1-8, Paris, July 2011.

[37] Jiayue, Ma’ayan Bresler, Mung Chiang and Jennifer Rexton, “Toward Robust Multi-layer traffic engineering: Optimization of Congestion control and routing”, Selected Areas in Communications, IEEE Journal, pp.868-880, vol. 25, June 2007.

[38] Heller, B., S. Seetharaman, P. Mahadevan, Y. Yiakoumis, P. Sharma, S. Banerjee, and N. Mckeown. 2010. “ElasticTree: saving energy in data center networks”. In Proceedings of the 7th USENIX Symposium on Networked Systems Design and Implementation, 17:1–17:16.

[39] Benson, T., A. Anand, A. Akella, and M. Zhang. 2010a. “The case for fine-grained traffic engineering in data centers”. In Proceedings of the 2010 Internet Network Management Workshop/Workshop on Research Enterprise Networking, 2:1–2:6.

[40] F. Shahrokhi and D. W. Matula, “The maximum concurrent flow problem,” Journal of the ACM (JACM), vol. 37, no. 2, pp. 318–334, 1990.

[41] P.-O. Bauguion, W. Ben-Ameur, and E. Gourdin, “A new model for multicommodity flow problems, and a strongly polynomial algorithm for single-source maximum concurrent flow,” Electronic Notes in Discrete Mathematics, vol. 41, pp. 311–318, 2013

[42] Internet2 Network Infrastructure Technology. Internet2. Retrieved 2015-02-16. [Online]. Available: http://www.internet2.edu/media files/1

Database Design for Agile Stakeholder Communication

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***Abstract*—Communication is the key to any successful project development, because it is a process of transmitting ideas and information. In many cases, a project developer communicates but still lack understanding. Stakeholders and the project developers often make the same mistake and too easily consider a model to be properly communicated or validated. They fail because of human factors which are an important and diverse, and non-technical issue [10]. The issue is especially important regarding to the process of communication. We focus mostly on communication with stakeholders, taking onto account diversity such as background, knowledge, skills, concerns, focus, interest, language (jargon, terminology), etc. Based on the principles, this research presents a database design of Agile stakeholder communication support.**

Keywords— database, database design, information modelling, stakeholder communication, Agile service development

I. Background

Communication is the key to any successful project development, because it performs a central role in the process of transmitting ideas and information. A project developer who fails to consider a prior communication before a project execution tends to have a blurred view of a project goal.

A clear communication ensures that project developers and stakeholders understand their roles, responsibilities, and objectives of their project. The understanding comes from an effective communication achieved through good communication activities. A best practice of communication activities complies with four-step system approaches which are a developing communication strategy, a developing communication plan, an executing communication activity and measured communication activity effectiveness [8]. The approaches reduce a number of issues because they work as a flow that supports a communication management.

In addition, there are some benefits to consider in a communication management. These benefits are: (1) Increased likelihood of the stakeholders' needs being met in terms of successful project resolution, (2) Much easier to track the progress of the project, (3) Can be amended, update and recorded accurately, (4) As the overall enthusiasm and commitment to the project [7].

Again, a lack of communication will influence the project development. First, the project developers who do not consider the importance of communication will lose time, money, and energy. Second, in many cases though the project developers communicate with stakeholders about a project but they still lack understanding. The project developers and the stakeholders often make same mistakes and easily consider a model to be properly communicated or validated [10]

Consequently, an Agile service development that is guiding principles and frameworks of a service development puts communication as a fundamental value. During project developments, the project developers and the stakeholders need to communicate effectively to share understanding, consensus, and commitment to achieve the project goals.

II. The Theoretical Basis

A. Stakeholder Communication in Agile Service Development

The communication in Agile service development means communications between project developers and stakeholders in order to report the results of work in achieving goals. Another reason to have such communications which mean meeting the customer requirements with the idea/solution of the project developer is to clarify either project goals or project requirements. Agile service development refers to three typical properties to indicate what makes such communication special [10]:

1. Intensive: Communication with stakeholders and others involved is intensive, and remains intensive throughout the projects time span, and possibly even after (maintenance).
2. Non-technical: Disregarding professional developers, such communication involves people who are not able or willing to talk in ‘technical terms’ unfamiliar to them, and not able or willing to use technical ‘tools of the trade’ like detailed specification or models. Moreover, these terms and tools may be not so adequate in capturing the essence of what stakeholder talk about.
3. Diverse: There is much diversity between stakeholder involved concerning their background, knowledge, skills, concerns, focus, interest, language (jargon, terminology), and so on.

Difficulties and issues will occur if we do not consider the properties above and will impact on communication without effectiveness and efficiency.

Developing information system is a complicated process because it will combine technical skills and business or organization needs. In contrast, two people from the different skills and backgrounds will find that it is not easy to communicate. This is because in most cases, the analysis results and design results will be translated into modeling languages or diagrams which are presented to the customer as approval materials. Not many people realize that this is the worst agreement during a communication because later on the developer and customer will easily say 'yes' even though they do not clearly understand the model and easily consider that the model is already communicated or validated.

So to avoid cases above, it is important to involve three basic levels of an agreement in communications [10].

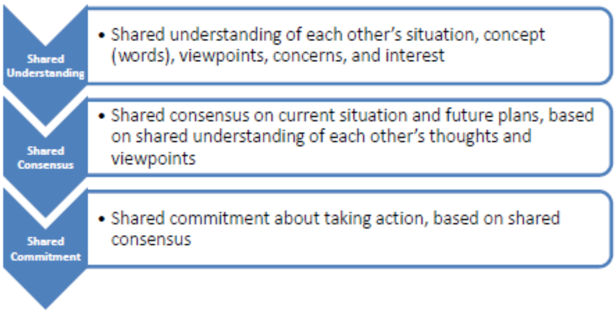

Fig. 1. Basic Levels of Agreement

This agreement level must be done sequentially. If one level fails, the level built upon it will also collapse.

Another aspect that is important to think of is communication material. We need to do distinction between 'model' and the communication material. We can say that this material is 'view'. The view means providing information about a system, so developers do not need to show the model to the customer. In other words, it does not show your model but only explains the effects of your model.

We can make it simple so developers and customers will understand each other more easily in order to achieve the goals. Thus, simplicity is a good approach for communication. Nevertheless many challenges perhaps will be faced later during the communication process.

Agile focused communication lies at the core of approach to 'stakeholder communication with and about models'. By identifying communication goals and abilities, and then selecting and applying certain practices or techniques to meet them, we can cope with the many diverse communication challenges might face [10].

B. Communication Situation

Communication situation (CS) is an interactive session between people involved in service development (information system development) during a structured, meaningful, purposeful exchange of information taken place [10]. In Agile service development, there are two Communication Situation Templates (CSTs):

1. Intentional CST (I-CST) covering goals, and given situation,
2. Operational CST (O-CST) covering the CS setup: how it is to be organized

Based on the theories above the outline of the communication situation is described into clearly structures as follows:

1. The project has goals
2. Each goal has sessions
3. Each session has inputs, processes and results
4. Each input is divided into two parts: Input artifact and People. The People part consists of Session Role, Project Role, Organization Role, People Profile, Personal Details, and Attitude.
5. Each process is divided into seven parts: Activity, Artifact Creation Pattern, Technique, Practice, Media, Space, and Time. The Activity part consists of purposes. The Space part consists of Location and Layout. The Time part consists of Date, Duration, and Time Type.
6. Each result is divided into two parts: Results Artifact and Social Results. The Result Artifact part consists of State, Domain, Level of Details, Language Concepts, Verbalization, Visualization, and Status. The Social Result part consists of Share Understanding, Share Consensus, and Share Commitment.

The flow diagram of communication situation is illustrated below.

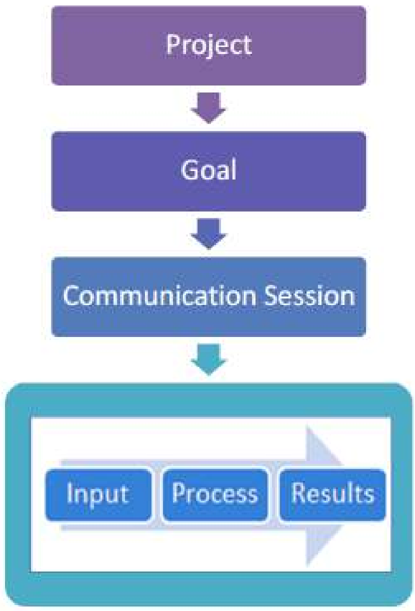

Fig. 2. The Communication Situation

Herewith the complete principles of Agile communication situation in hierarchy diagram below:

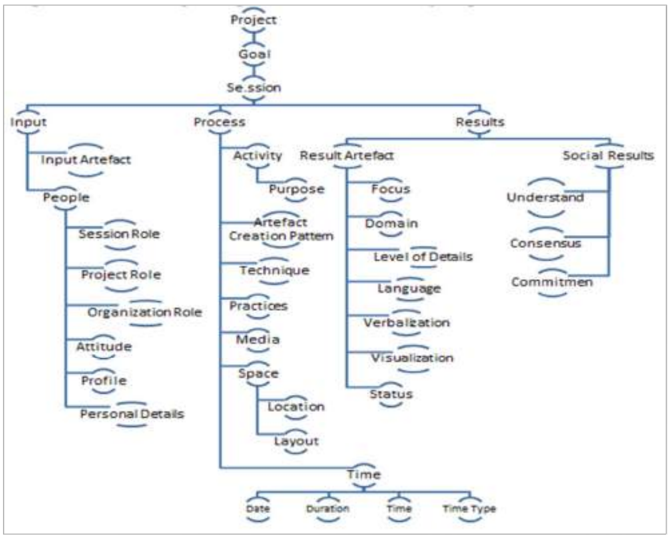

Fig. 3. The Communication Situation

III. RESULTS

A. *Conceptual information model using FCO-IM techniques*

To model the facts, information, and rules of the Agile stakeholder communication principles, FCO-IM techniques were used. The FCO-IM method steps are verbalization, classification and qualification, constructing an Information Grammar Diagram (IGD), adding constraints to the IGD (UC and TC), and GLR (grouping, lexicalizing, and reducing). In the verbalization step, the facts of the communication situation were described in figure 3 was verbalized in natural language. It means putting all the facts into words. Thereafter the classification and qualification step are taken. The classification means *arranging* things into classes (group) and the qualification means giving a meaningful name to each class. All the steps are done by using Case Talk application and the result of both those steps is the IGD. After that constraints are added to the IGD. The constraints added are uniqueness constraints (UC) and totality constraint (TC). The UC is a constraint indicating that values (combination of value) can occur only once (unique) and the TC is a constraint indicating whether the facts can be filled in or not. Finally the last step is GLR. The GLR carried out in order to derive a relational database schema from the elementary IGD. The relational database schema is created for recording information modeled in a redundancy freeway.

Below the Information Grammar Diagram (IGD) design are Input, Process, and Results as describe in the communication situation figure 2.

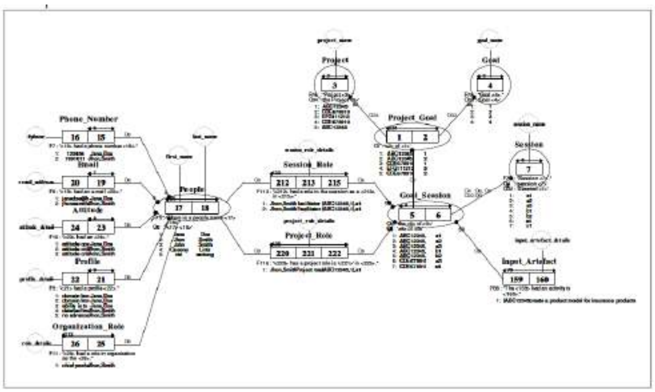

Fig. 4. Information Grammar Diagram (IGD) – The Input

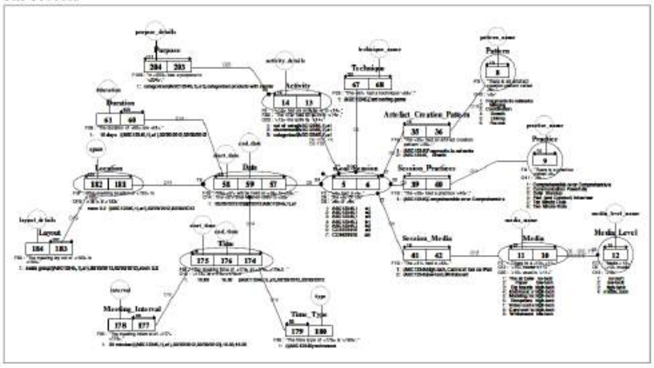

Fig. 5. Information Grammar Diagram (IGD) – The Process

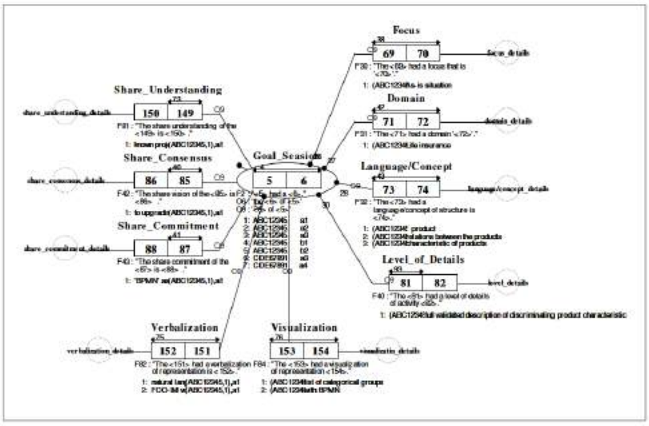

Fig. 6. Information Grammar Diagram (IGD) – The Results

B. *Physical Database Schema*

The following figures are designs of a database. These designs called physical database schema consisting of tables, columns, unique identifiers, foreign keys, data types, and data lenghts. The schema is retrieved based on an analysis phase result from the FCO-IM steps. After completing all the FCO-IM steps, the relational schema is retrieved and it is generated by Case Talk application. Next, according to the relational schema, improvement is made to edit data lengths and data types of the fields. The editing is done using the Power Designer application. In addition, the schema also covers the Agile communication situation processes including planning and the execution of planning.

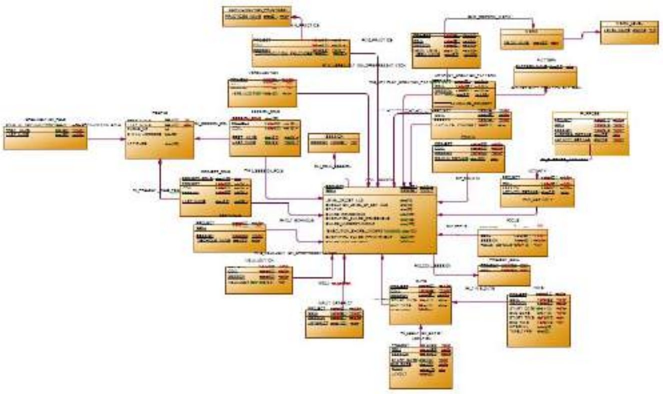

Fig. 7. Physical Database Design – The Palnning

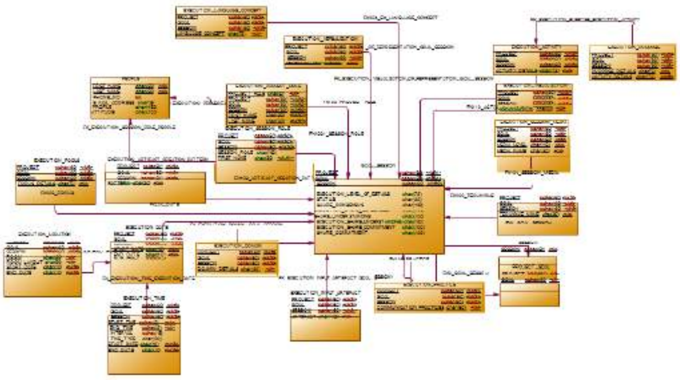

Fig. 8. Physical Database Design – The Execution

C. Database Implementation

The following are the implementation of the physical database design in figure 7 and figure 8. The physical database design was translated into a set of SQL statements that define a database. The database below generated by MSSQL Server application.

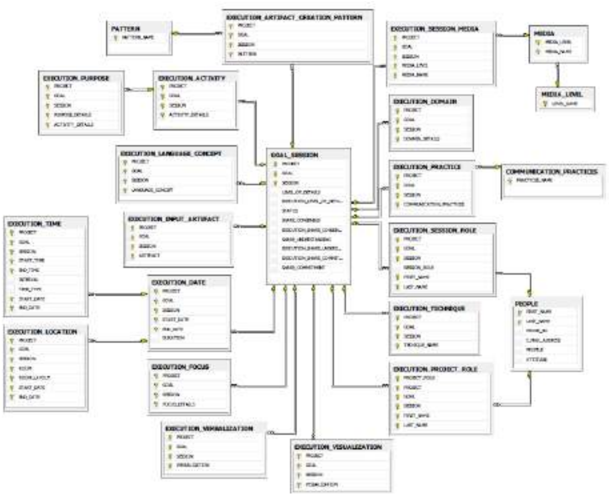

Fig. 9. Database Schema – The Execution

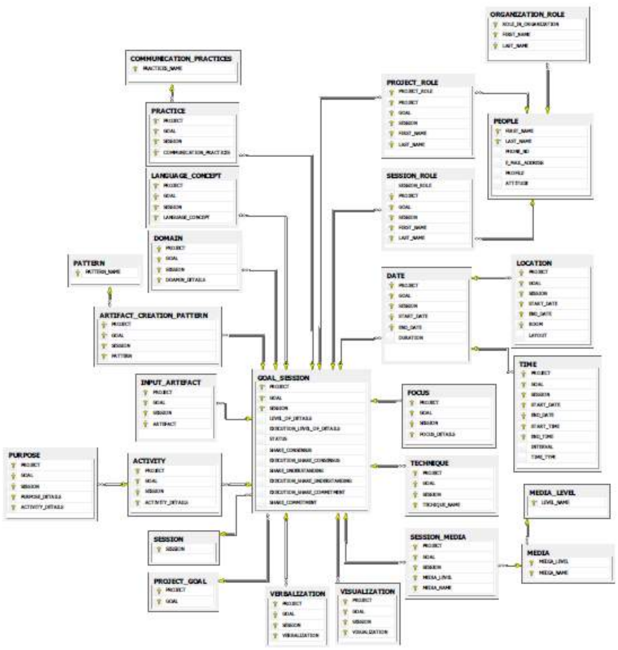

Fig. 10. Database Schema – The Planning

IV. CONCLUCION

One of an essential part of an information system is database because the database is the backbone of an application. Hence, in the research it is clearly understood that database is very important.

The process of the database development is divided into three phases: analysis, design, and implementation. In the analysis phase FCO-IM methods were used. Using the FCO-IM methods the facts, information, and rules of the Agile stakeholder communication principles are modeled. Furthermore, by using Case Talk application the relational schema of the database was generated. Next in the design phase, physical database was designed by using Power Designer application to get proper data lengths and data types of the field. Finally in the implementation phase the physical database was implemented using MSSQL Server application. The database implementation was obtained by generating DDL. The DDL consists of 50 tables within one database covering the communication planning and the planning execution. Equally the important thing is that using the FCO-IM methods can guarantee the database that is free of data redundancy. It means that the database development pass normalization rule. Pass the normalization means that a table must be in normal condition in order to make a database organization easily, to develop a database design flexible, and to handle database security better.

REFERENCES

[1] Atwood *et al.*, "Facilitating Communication in Software Development," New York.,2013.

[2] Bonnie Biafore, P. (2011, March 31*). The Project Communication Plan*. [Online]. Available:http://www.mpug.com

[3] Cegielski et al., "Communication Management and It's Impact on Successful IT Program," *IADIS, International Journal on Computer Science and Information Systems*, pp.14-28.

[4] Charvat, J. P. (2002, November 13). *Project communications: A plan for getting your message across* [Online]. Available: http://www.techrepublic.com

[5] Freedman, R. (2000, September 25). *Communication plans are key to project success* [Online]. Available: http://www.techrepublic.com

[6] Holten *et al*., “Communication in Organizations: The Heart of Information”.Germany.,201.

[7] Manor, W. (2006, August 30). *The Four Benefits of Effective Communication Management Software* [Online]. Available: http://ezinearticles.com

[8] MITRE. (2012, May 17). *Effective Communication and Influence* [Online]. Available:.http://www.mitre.org

[9] Romahn, E. and Hartman, F., ”Trust: A New Tool for Project Managers,” in Proc of the 30th Annu Project Management Institute, Philadelphia.,1999.

[10] S.J.B.A. Hoppenbrouwers, “*Stakeholder Communication”* in Agile services development:combining adaptive methods and flexible solution, Berlin; New York: Springer,2012, pp.144-176.

[11] Yoon, C. S. (2013, January 1). *Participatory Communication for Development* [Online]. Available: http://www.southbound.com.

The Application of Fuzzy Database for Smartphones Selection

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***Abstract*—Various kinds of smartphones cause the confusion among the people in choosing while they have their own criterias of smartphones. This study makes a web-based application programs that implement Tahani fuzzy database into a decision support system of smartphones selection, then the software of this application is evaluated by testing the usefulness (usability testing) based on the five aspects of usability. In Tahani model, the standard relationship is still being used but to obtain information on its query, this model uses fuzzy set theory. The user inputs the criteria, the system searches the data, then delivers the output in the form of smart phones recommendations according to criteria required by the users (consumers) and are shown through the fire strength (membership value) which lies between 0 (zero) to 1 (one). The higher fire strength (approaching 1) shows the product is closer to the search criteria or the greater recommendation, on the other hand, the lower fire strength (approaching 0) shows the product is further to the search criteria or the smaller recommendation. The software evaluation result of this application indicates that the value of usability acceptance by the user is over the number of 3 (over the median) in the scale of 5 or has an average value of 4. In general, the web application software has a good value of usability, i.e. learnability, efficiency, memorability, errors and satisfaction.**

Keywords*—*tahani fuzzy database; fuzzy set theory; query; the degree of* m*embership; fire strength; usability testing.

I. Introduction

The existence of the digital era that continues to move quickly accompanied by the sophistication of technology has an impact on the increasing number of high-tech tool that is inserted into the phone (mobile phone). The phone is not just a mobile phone that is used to speak but also it has been converted into a smartphone, cellular phone with microprocessors, operating systems, and memory. This smartphone combines the functionality of a PC (Personal Computer) and the handset so the display of luxury gadgets such as: text messaging, voice, camera, media player, video games, email access, digital TV, search tools, personal information managers, Internet access, GPS feature , 3G access, wi-fi network connection and Bluetooth. Smart phones consist of many attributes similar to a personal computer, i.e.: processor, memory, device input / output, and operating system / OS (Operating System), such as Symbian OS or Microsoft Windows Mobile OS. The operating system is stored in ROM (Read Only Memory) and is operated by the processor as an interface so that consumers can store data, change settings, and others.

By the completeness of the existing features in just one hand only, phone not only as a way of lifestyle but also has become a necessity in today's environment of modern society. There are various types of smartphones on the market and each has a specification and prices. This has an impact on the consumers who will buy and use mobile facilities, namely the onset of confusion in determining the choice when they buy mobile phones.

In the process to determine which phone is right for purchase, consumers possess certain criteria. These criteria have uncertain value (ambiguous) while the existing data in mobile databases are certainly valuable data (crisp). Therefore to deal with criteria that have uncertain value, we can use fuzzy logic.

This study would design a database application that uses Tahani model fuzzy database method that will process the data of the phone and it will generate output data according to criteria of consumer's recommendation. In the process of finding a recommended cell phone, fuzzy logic will be incorporated into the query and ease in entering the desired criteria, then the delivery of information is presented using the web facility.

II. The Concept Of Fuzzy Logic

A. *Definition Of Fuzzy Logic*

The word "fuzzy" is an adjective which means blurred, vague, unclear or uncertain. Fuzzy logic is a development of the concept of a fuzzy set was first introduced in 1965 by Lotfi A. Zadeh. Membership in a fuzzy set is not expressed in the form of true or false, but is expressed in the value of membership. Fuzzy logic works by using the degree of membership of a value [2].

B. *The Basic Concept Of Fuzzy Tahani Database Model*

Fuzzy database system is a method that uses fuzzy standard database. In a standard database, the data are classified by how the data are viewed by the user. Therefore on a standard database, the data presented will come out as the saved data. In fact, sometimes a person needs information from the data in ambiguous form. If it happens, the use of fuzzy database is necessary. Fuzzy Tahani database model is still using the standard database, but this model just uses fuzzy set theory to obtain information on its query [3].

C. *Fuzzy Database Application Program*

Fuzzy database application program is a data search program which applies theory of fuzzy database. The inputs of this program are the input of fuzzy variables, the input of fuzzy sets, the input of non-fuzzy variables and the input of data which is being searched. After inputting the data and selecting the search criteria, the next step is calculating the membership degree of data in each set of a variable based on the membership functions that have been previously selected. After the degree of membership is obtained, the next process is the calculation of fire strength which is closest to 1 (one). The higher fire strength the greater the recommendation, on the contrary, the lower the fire strength the smaller the chance of the pruduct being recommended. Fire strength could be called as the value of product recommendation.

III. RESEARCH METHOD

A. *Materials And Research Tools*

The materials were used in this research, namely: data from the object of research include: the type, specifications or features of smartphones; variables into system input, and the literature or reference books that related to the manufacture of application programs and the preparation of research reports. The research equipment consists of:

- The hardware, namely a computer that serves as a hosting site and have a specification sufficient for the purposes of designing systems or devices to run PHP and MySQL on Windows or Linux operating system.
- The software with the minimum specs are: Windows XP (OS), Apache 2.2.17 (web server), MySQL 5.0.24a (database server), PHP 5.1.6 (programming language), Adobe Dreamweaver 8 (PHP Editor) and JQuery. Using the XAMPP 1.5.4a package (Apache, MySQL and PHP).

B. *Research Workflow*

The workflow of this research can be explained in a diagrams on Figure 1.

C. *Limitation of Fuzzy Set and Formulation of Membership Function*

Basically there is no specific provision in determining a membership function which is used in the process of fuzzification because each system has different levels of conformity.

There are eight variables and three fuzzy non-fuzzy variables. Each fuzzy variable is divided into three categories or three fuzzy set. (Table 1)

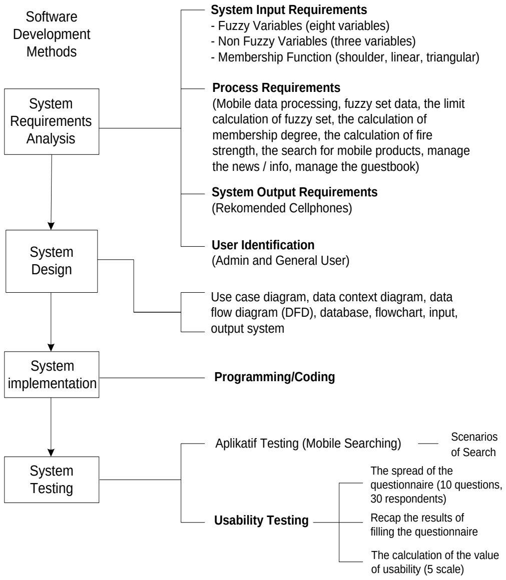


Fig. 1. Research Workflow

TABLE I. FUZZY VARIABLES, NON-FUZZY VARIABLES AND FUZZY SET

No	Fuzzy Variables	Fuzzy Set
1	Price	Cheap, Normal, Expensive
2	Length	Short, Normal, Long
3	Width	Tight, Normal, Wide
4	Thickness	Slight, Normal, Thick
5	Weight	Light, Normal, Heavy
6	Screen Size	Small, Medium, Large
7	Resolution of Camera	Low, Medium, High
8	Capacity of Memory	Small, Medium, Large
Non - Fuzzy Variables		
There are or no facilities: 1. GPS (Global Positioning System) 2. 3G/UMTS (*Universal Mobile Telecommunications System*) 3. HSDPA (*High Speed Downlink Packet Access*)		

In this study, every fuzzy variable uses shoulder-shaped membership function approach to obtain the degree of membership of each variable in a fuzzy set, shown in Figure 2, Figure 3, Figure 4, and Figure 5.

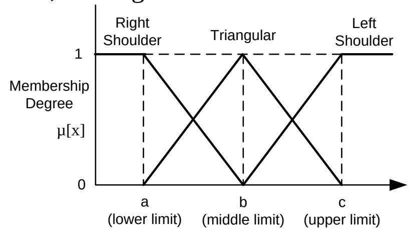


Fig. 2. Shoulder-shaped curve representation

The set 1 is using left shoulder-shaped membership function or a deacreased linear function.

$$\mu[x] = \begin{cases} 1; & x \le a \\ (b-x)/(b-a); & a \le x \le b \\ 0; & x \ge b \end{cases} \quad \text{.........} \; (1)$$

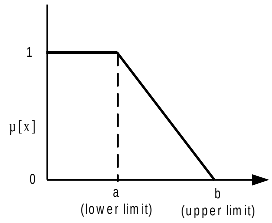


Fig. 3. Linear fuzzy set is decreasing (linear down)

The set 2 is using triangular membership functions.

$$\mu[x] = \begin{cases} 0; & x \le a \text{ atau } x \ge c \\ (x-a)/(b-a); & a \le x \le b \\ (c-x)/(c-b); & b \le x \le c \end{cases} \quad \text{.......... (3)}$$

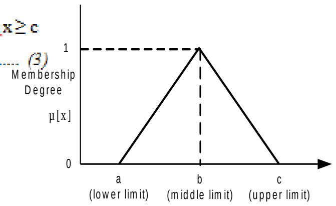


Fig. 4. Triangular fuzzy set

The set 3 is using right shoulder-shaped membership function or an increased linear function.

$$\mu[x] = \begin{cases} 0; & x \le b \\ (x-b)/(c-b); & b \le x \le c \\ 1; & x \ge c \end{cases} \quad \text{.......... (2)}$$

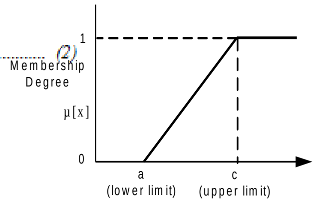
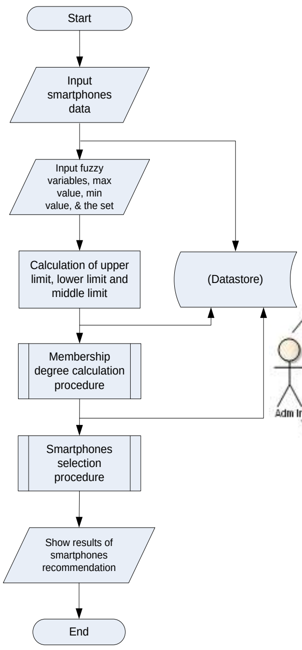


Fig. 5. Linear fuzzy set is increasing (linear up)

D. System Design

This is the stage of translation where the data are analyzed into an understandable form for the user, shown in Figure 6 and Figure 7.



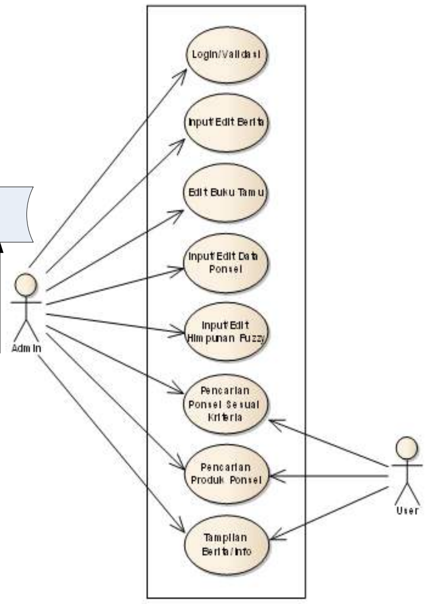


Fig. 6. The smartphones selection flowchart   Fig. 7. Use case diagram

IV. RESULT AND DISCUSSION

A. The Smartphons Selection Database

After going through the stages of analysis and design system, the next step is making the system that started from the database creation. DBMS that used in this database is MySQL. The database created is named ponselweb_db consist of eight tables, namely: dataponsel table, limits table, mu table, rekom table, merk table, guestbook table, login table, and the news table.

Fig. 8. Structure of ponselweb_db database with eight tables

B. User Interface System

Fig. 9. The main page interface (*index.php*)

C. Calculation of Membership Degree

To calculate the degree of membership variable of each fuzzy variable uses membership function approaches (membership function) as formulated on the Figure 3, Figure 4, Figure 5. In the crisp set there are two possibilities of membership values, 0 and 1, on the set of fuzzy membership value lies in the range 0 to 1.

The mobile phone Acer Liquid Metal with a price of Rp 4.200.000,- is included into 2 (two) categories of set: the set price **Cheap** with degrees of membership (μ_{CHEAP}) = 0.183 and also classified in **Normal** price set by the degree of membership (μ_{NORMAL}) = 0.817, this means that mobile phone prices will be closer to the set of Liquid Metal **Normal** price (close to the value 1 on the set of Normal) and **do not** belong in the set price Expensive, it is seen from $\mu_{EXPENSIVE}$ = 0.

TABLE 2. THE RESULT OF PRICE VARIABLE MEMBERSHIP DEGREE CALCULATED BY SYSTEM

VARIABEL HARGA						
Merk	Nama Ponsel	Harga (Rp)	Nama Himpunan			
			Murah	Normal	Mahal	Bebas
Acer	Liquid Mini E310	3482000	0.440143	0.559857	0	1
Acer	Iconia Smart	6093000	0	0.504301	0.495699	1
Acer	Stream	5000000	0	0.896057	0.103943	1
Acer	Liquid E Ferrari	4230000	0.172043	0.827957	0	1
Acer	beTouch E140	2490000	0.795699	0.204301	0	1
Acer	Liquid Metal	4200000	0.182796	0.817204	0	1

D. *Mobile Phone Search Testing*

Mobile Phone Search Scenario

The user select and input the criteria value of price, width, weight, camera resolution and memory; for Long, Thick and Screen Size are emptied (not selected). The value entered is not certain but a predicted value. After clicking the "Search" button, the system will display the recommended mobile products.

Fig. 10. Recommended Mobile Phone Search Input Interface

TABLE 3. RECOMMENDED MOBILE PHONE OUTPUT

REKOMENDASI PONSEL SESUAI KRITERIA			
Merk Ponsel	Tipe Ponsel	Harga (Rp)	Nilai Rekom
LG	P500 Optimus One	2245000	0.736842
Nokia	E5	1980000	0.5
Motorola	MB508 Flipside	3336000	0.492473
LG	C550 Optimus Chat	2225000	0.491228

The recommended cellphones have a price and rekom value. Value of the recommendations that provided by the system lies between 0 - 1. Value of rekom = 1 (one) indicates a full recommendation, if the value of rekom approaches the 0 (zero) value then the phone is not recommended. The table above have been sequenced from mobile phones with the highest to lowest recommendation. LG P500 Optimus One has the highest recommendation value (0.737) where as the LG C550 Optimus Chat has the lowest recom value (0.491).

Recommendation value is the value of fire strength, which the membership value of the operating results of two sets or more. Fire strength as a result of the operation with AND operator is obtained by taking the smallest membership value between elements in the sets within the set of Price, Length, Width, Weight, Screen Size, Resolution Camera and Memory, while for non-fuzzy variables, if "There is" the facility, the value is 1 and if "There is not" the fasility, the value is 0 [2].

E. *Usability Testing*

Usability is often related to be an acceptance value (acceptance) of a person towards a product or systems based on understanding and precision of action/reaction of a person towards an interface [5].

According to Nielson [4], usability aspects of this testing includes five things, namely: *Learnability*, *Efficiency*, *Memorability*, *Errors*, dan *Satisfaction*. The contribution of the question of questionnaire to every aspect of usability above can be seen in Table 4. From the interpretation of usability testing results in Table 5, the obtained Usability Value Recap of each attribute can be seen in Table 6.

TABLE 4. USABILITY ASPECTS PLOT

No.	Question	Aspect of Usability				
		Learn	Eff	Mem	Err	Sat
1.	Is the web interface can be known from the beginning?					
2.	Are you able to find a product with the desired specifications mobile phone?					
3.	Whether you can fill in the criteria (values) on the Menu Support for mobile search?					
4.	Is the mobile phone product that's recommended by the system (search results) are easy to find along with their specifications?					
5.	Are there letters easy to read?					
6.	Are the symbols easy to understood pictures?					
7.	Does this web color design pleasing?					
8.	Is the language displayed can be understood?					
9.	Are you able to access information on each page?					
10.	Can you recall the menus and page views that exist on the web application?					

TABLE 5. QUESTIONNAIRE ANSWERS PERCENTAGE

PK	KMS	KM	CM	M	SM
	1	2	3	4	5
1.	0%	0%	10%	60%	30%
2.	0%	0%	7%	63%	30%
3.	0%	0%	7%	67%	26%
4.	0%	0%	4%	63%	33%
5.	0%	0%	10%	63%	27%
6.	0%	0%	7%	60%	33%
7.	0%	3%	10%	60%	27%
8.	0%	0%	10%	63%	27%
9.	0%	0%	7%	63%	30%
10.	0%	3%	17%	60%	20%

*PK=Pertanyaan Kuisioner; **KMS** = Kurang Mudah Sekali; **KM** = Kurang mudah; **CM** = Cukup Mudah; **M** = Mudah; **SM** = Sangat Mudah*

TABLE 6. USABILITY VALUE

No.	Attribute	Value
1.	Ease of the interface identified.	4.2
2.	Ease of mobile products and specifications found.	3.93
3.	Ease of filling criteria (value) for the mobile search.	4.19
4.	Ease of mobile products are recommended by the system (search results) and its specification is found.	4.29
5.	Ease of web reading.	4.17
6.	Ease of picture symbols understood.	4.26
7.	Color design comfortable.	4.11
8.	Ease of understandable language.	4.17
9.	Ease of access to information on every page.	4.23
10.	Ease of recall menus and web pages display.	3.97

The result of Usability Value Recap shows that all attributes have the value of acceptance usability by the user, an average over of 4, so it can be said that the web application software that was created has good value of usability: Learnability, Efficiency, Memorability, Errors, and Satisfaction.

V. CONCLUCION AND SUGGEST

There are three conclusions that can be concluded from this research:

1. A software can be designed and implemented to make a decision as a consideration in purchasing smartphones by applying fuzzy Tahani database model.
2. The program can generate the name, type, degree of membership and dinamically draw the specification of the recommended mobile phones. The greater the degree of membership, the greater value of recommendations.
3. The result of Usability Value Recap shows the web application software that was created has good value of Usability: Learnability, Efficiency, Memorability, Errors, and Satisfaction.

There are two suggestions that can be considered in the next fuzzy database research. Those are:

1. Fuzzy and non fuzzy variables in the next studies can be added, for example: external memory capacity, data access speed, etc.
2. The software can be developed as a system of ordering or purchasing mobile phones from the recommended result.

REFERENCES

[1] Kusumadewi, S., Purnomo, H. 2004. *Aplikasi Logika Fuzzy untuk Pendukung Keputusan*. Edisi pertama, Graha Ilmu, Yogyakarta.

[2] Kusumadewi, S. 2007. *Basisdata Fuzzy Untuk Pemilihan Bahan Pangan Berdasarkan Kandungan Nutrien*. Seminar Riset Teknologi Informasi (SRITI) 2007, STMIK AKAKOM, Yogyakarta.

[3] Nielson, J. 2000. Usability 101: *Introduction to Usability*. Sumber: http://www.useit.com/alertbox/20030825.html. Tanggal akses: 7 Juli 2011.

[4] Wingnjosoebroto, S., Sudiarno, A., Harenda, D. 2009. Perancangan Interface Prototype Web Berdasarkan Pada Aspek Usability (Studi Kasus: Laboratorium Ergonomi dan Perancangan Sistem Kerja Teknik Industri ITS). Sumber: http://digilib.its.ac.id/public/ITS-Undergraduate-8877-2504100030-Paper.pdf. Tanggal akses: 7 Juli 2011.

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Vertical Information System

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Abstract— **Information systems have become important factors in a company, where information systems are useful in providing useful information for the company itself. The information system helps companies implement corporate activities at the beginning of the process until the decision making. In the concept Information system, we know the term of vertical information system, which is another strategy for increasing vertical information capacity. Vertical information system includes the periodic report, written information, and computer-based communications distributed to managers.**

In this article, discusses the approach in the development of vertical information system which will be preceded by comparison of operational systems, management information systems, and business intelligence systems. Next, we will give explanations about the role in the BI system in an organization, the challenges faced in the implementation of vertical information systems, approaches in vertical information systems, and approaches that we choose.

Keywords—vertical information system, periodic report, operational system, management information system, business intelligence system

I. Introduction

In defining control mechanisms in an organization, one needs to consider the information base of the organization. This information base provides historical trends and analysis of the data over a period of time. As De Leeuw states in his paper, control must know the overall objectives of the organizational system and should have a clear understanding of the model of the system to enforce the right mix of control. However, this approach may face challenges when the information base of the organization grows beyond proportions and a broader view of the organizational information pool is required.

Such requirement can easily be met by introducing vertical information systems (also called Business Intelligence solutions, or Data Warehouses). Such solution typically allows managers to have a clear picture of their organization’s performance within the industry. It allows managers to clearly see key performance indicators (KPIs) on different dimensions (time, region, product, etc.).

II. Operational System vs. Management Information System vs. Business Intelligence System

A. Operational System

An operational system is a term used in data warehousing to refer to a system that is used to process the day-to-day transactions of an organization. These systems are designed so processing of day-to-day transactions is performed efficiently and the integrity of the transaction data is preserved. Operational systems are generally designed to support high-volume *transaction processing* with minimal back-end reporting. It is also generally process-oriented or process-driven, meaning that they are focused on a specific business process or tasks. The operating system is generally concerned with current data and optimized to perform fast inserts and update of relatively small volumes of data. Data within operating system are generally updated regularly according to need. They are generally application-specific, resulting in a multitude of partial or non-integrated systems and redundant data and require a non-trivial level of computing skills among the end-user community.

Example tasks: billing, registration, etc.

B. Management Information System

A management information system (MIS) is a system or process that provides information needed to manage organizations effectively. The term MIS appears to depict a variety of applications is developed for the manager where the application is to provide information about sales, inventories and other data that will assist in the process of managing the company. An 'MIS' is a planned system of the collection, processing, storage and dissemination of data in the form of information needed to carry out the management functions. The terms MIS and information system are often confused. Information systems include systems that are not intended for decision making.

C. Business Intelligence System

Business intelligence (BI) refers to specialized system for retrieving decision support information, by analyzing huge amounts of data to help decision makers improve their performance of their company of organization, gain competitive edge, and optimize business process. BI technologies provide historical, current, and predictive views of business operations. Common functions of business intelligence technologies are reporting, online analytical processing, analytics, data mining, business

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performance management, benchmarking, text mining, and predictive analytics. Business intelligence aims to support better business decision-making. Thus a BI system can be called a decision support system (DSS).

BI uses technologies, processes, and applications to analyze mostly internally, structured data and business processes while competitive intelligence gathers, analyzes and disseminates information with a topical focus on company competitors. Business intelligence understood broadly can include the subset of competitive intelligence. Business Intelligence System (BIS) has become the essential tools in helping management make decision in most organizations. With the vast amount of data collected through the transactions made through the years, organizations can stay ahead of competition. Business Intelligence is also supported in strategic planning and processes.

Discussion about Business Intelligence solution wouldn’tn ben completen withoutn introducingn then threen heavyweights in the BI community. Ralph Kimball, Bill Inmon and Dan Lindsted are leading specialists who have their own different theories on how to best go towards a business intelligence solution. Kimball believes that organizational information should be put in vertical information systems at the atomic level, but in a de-normalized manner, whereas Inmon argues that such endeavors should be attempted in a normalized way as done in operational systems. For both, Lindsted argues that a hybrid approach is the best path sincenKimball’sn denormalized approach is pertinent only for small data warehousesnwhereasnInmon’snapproachnisnpertinentntonoperational systems. The BI community seems to fall in the pathnofnLindsted’snapproachnrecently.

Here are highlights from approaches proposed by these BI gurus:

TABLE 1. THE HIGHLIGHTS FROM APPROACHES PROPOSED BY BI GURUS

Bill Inmon	***Ralph Kimball***	***Dan Lindsted***
Firstly, process the design with carefully, accommodate the user need from that time to next time and then built the data warehouse.	Built the data mart in departments which have needs and initiatives, then process the integration data mart if needed.	A collection of detail-oriented, history-tracing and uniquely linked collection of normalized tables.
Data marts sourced from data warehouse after the data warehouse builds, data marts which in departments take the information source from one (and only one) data warehouse.	The data warehouse is a collection of data marts. It means, first built the data marts and then built the data warehouse.	A hybrid approach combining the best of 3NF and dimensional modelling.
Architecture name is Corporate Information Factory (CIF)	Name architecture is bus data warehouse.	Dan argues that his Data Vault is not an architecture, it is more like a standard.
Information stored in the 3NF relational form.	Information stored in multidimensional database.	Flexible, scalable, consistent and adjustable to business need.

III. CHALLENGES OF IMPLEMENTING VERTICAL INFORMATION SYSTEMS

Business intelligence has changed the fate of many big organizations, efficient business intelligence systems in placenhavenmadenthenmanger’snlifenlittleneasiernbynprovidingn accurate and concise information which helps to make a better decision. Vertical Information systems can provide unprecedented information to the managers by the aid of which managers are able to make thoughtful decisions that is best for the company.

Though it looks all good from the outside and with amazing success stories to back up the fact that implementing an effective business intelligence solution is beneficialnfornannorganizationnit’snnotnsansnchallengesntonimplement one functional and useful solution. Basically the challenges that come into significance are related to the business organization and how they function. If the business processes the business organization do not comply with any standard and are haphazard then it will not be possible to build a vertical information system to serve any meaningful purpose.

Due to the size and the enormous scope of the vertical information system in the form of a business intelligence solution, it requires a very high political will to implement the solution. It also involves a high degree of time, effort which equates to soaring costs to build these kinds of systems. This is a prominently evident challenge that as the costs and resource investment goes high on a system that is not yielding anything fast the management has to rethink about investing more funds and resources to the system. So the management should be committed to get a vertical information system that will eventually help to make better decisions resulting in higher turnover of the company.

Another challenge in implementing a business intelligence solution that commands high priority is that the system should be focused on the business and management aspect rather than just the IT aspect. As the system is generally owned and maintained by the IT department, but its main users are the management of the company. So making it usable and focused on the information needs of the user/business executive makes the system sustainable and easily accepted by the end user.

Data availability and data quality are two big questions to ask if a vertical information system is to be put in practice. In some Online transaction processing (OLTP) system the vendor might prohibit other systems from accessing the database. Chances are there might not be any proper system in place to get the data from. Even if there is sample data, guaranteeing data quality to get the accurate and desirable results needed by the management is a big challenge in itself.

Another practical challenge is to get over the silver bullet syndrome. Implementing business intelligence will solve all the problems of the business, it will give answer to any data and information related question one can ask is the general assumption of the user. The over expectations should be trimmed down on the realistic level, the realistic picture and the things that can be realized should be communicated. Managing this mountain of expectation of the user that business intelligence solution is the silver bullet for all business problems is a hand task to execute.

IV. Approaches to the Vertical information system

There exists approaches to build and implement vertical information system in the form of a Business Intelligence solution, the approaches basically comprise of:

1. Top Down Approach: the vertical information system is made because the management layer feels the need of the system. It is derived from the mission, objective and vision of the organization. This approach is very positive for the system because as management is committed and wants to implement the system, there is easy to implement this system and the financial factors are also coverednasnit'snthenmanagementnwhon wants the system.

2. Data Driven Approach: is based on the assumption that the data already exists in the OLTP system or some external database which can be accessed by the organization. In this approach the organization wants to analyze the existing data and derive a vertical information system to assist the business executives to make meaningful and logical decisions. In this approach the data history and forecasting should be kept in mind to give out the results demanded by the management.

3. Process driven approach: is based on the business processes. Processes that run on the organization define the organization. The success of a company can simply be measured by determining how efficient its processes are. Success largely depends on the processes being carried out, so it this approach to implement a vertical information system helps to do the things on a horizontal process level.

4. Off the shelf: seems to be the easiest approach where the business can buy a rebuilt solution and implement it. It is a tool as well as a solution. The strict financial regulation on how the financial reports should be prepared decides how the reports should be. A business intelligence solution in this approach is a benchmarking device in itself as the same solution can work out for company X and your competition, but it might not be the right solution for your business.

V. Suggested Approach for Vertical Information System Development

Inn hisn widelyn acclaimedn bookn entitledn 'Then Datan WarehousenToolkit',nRalphnKimballnstronglynarguesnthatn the job of a data warehouse manager is basically similar to the job of a publishing editor-in-chief of a magazine. As the editor-in-chief'sn jobn rangesn from basic lay outing to customer demographic studynandnmaintainingnreader'sn trust,nthenITnmanager'snjobninvolvedninnBIndevelopmentsn follows a similar path. BI developer teams need to understand the users of business area and their business processes, publish the right data at the right time, and maintain the trust of executives.

In light of these facts, we believe that the process-oriented way of BI development should be the leading approach used. In fact, the very first task in developing a vertical information system should be selecting the business process to model. We need to understand the business process of organizations. In following this approach, however, we need to understand that a business process DOES NOT mean a business function or a business department.

Tonhighlightnthisnpointnfurther,nlet'sntakenancasenofnan company that is trying to implement a VI system for its order'snmanagementnprocess.nThenfirstnquestionnwenshouldn be asking in addressing such a requirement should be what is the business process and with what source system isnitnsupported.nAnwrongnquestionntonasknwouldnben'whichn departmentnneedsnthisnverticalninformationnsystem'.nThen reason behind this is that the order management can be used by both Marketing department and Sales department. These are two different organizational units with the same interest in the final published data. If we treat them separately then there will be duplication of resources and data and inconsistencies may occur. In BI world, the best way to ensure consistency of data is to publish the right data once. A single publishing run not only reduces our technical efforts in getting the data to the user, but it also delivers the same information to all users (irrespective of their departments).

In his book, Kimball [1 pg.32] interestingly argues against the data-driven approach as follows:

> "We strongly encourage you to resist the temptation to model the data by looking at the source data files alone. We realize that it may be much less intimidating to dive into the file layouts and copy-books rather than interview a business person; however they are no substitute for user input. Unfortunately, many organizations have attempted this path-of-least-resistance data-driven approach, but without much success."

We also argue that the process-drive approach cannot succeed in its own without concerted effort by company management. Unlike operational systems that capture the nitty-gritty details of the operational business world, vertical information systems are targeted towards the business people in their posh offices in 10th floor. These people may need to exercise their political muscles to get the BI solution live. The information dealt is most sensitive, even within the organization so successful with such endeavor needs management commitment and vision. To address this we need to couple a top-down approach and convince the business executives and get their commitment in this regard.

VI. Implementation of Business Intelligence System in ETL Process using Pentaho Data Integration

BI system changes the enterprise data, e.g. operational data, transactional data or others (On Line Transactional Processing-OLTP) into the dashboard or graphical view. This application analyzes data in the past, then using dashboard or graphic to support the decision and support the enterprise plan. BI is equal to briefing books, report and query tools, and executive information System.

The sample data are GPA-PTIK data that we use to show all the implementation of Business Intelligence System and Pentaho has chosen as the open source BI application.

Extraction, Transformation, Loading (ETL) process is the important process to built data warehouse. A data warehouse is needed to store all transaction data (OLTP) before it analyzes into On Line Analytical Processing (OLAP). In General, the benefits of ETL are as follows:

1. Transform data from OLTP to OLAP (data warehouse)
2. Integration between applications
3. Clean data
4. Data migration
5. Export data

The ETL processes are inextricably linked to the data source and the database, which consists of:

1. Extract the data from an external source. An outside source can be a database, csv, excel, xml, web services, database, etc.
2. Transform (Edit) the data so as to suit the needs (can also incorporate elements of data quality).
3. Load (entering) the data into the target end. The final target could either be csv, excel, databases, XML, web services, database, etc.

Pentaho Data Integration (PDI) or Kettle ETL utility is open source under Pentaho Corp.Amerika. This was originally an initiative of Matt Casters, a programmer and consultant Business Intelligence (BI) from Belgium who has managed projects for enterprise BI.

Currently Kettle ETL is a utility that is very popular and one of the best on the market. Some advantages are as follows:

- Have a collection of data processing modules that quite a lot. More than 100 modules or step.
- Have a module that facilitates the design of the data warehouse model as Slowly Changing Dimensions Dimension and Junk.
- Performance and scalability are well-proven.
- Can be developed with a variety of additional plugins.
- Utility Kettle to be used in the integration of this data using the Spoon

In the multidimensional model, a database consists of several facts table and dimension table. A fact table contains the value of the aggregation that is the basis of measurement (measure) as well as several keys that are related to the tables of dimensions that would be the point of view of the measure.

In the process, the arrangement of the fact table and the dimension table has a standard design or schema because it is proven to improve performance and ease of translation to the OLAP system.

The schema which served as the basis for performing data warehousing. Two of the most common schema used by the various OLAP engine is the star schema (Star Schema) and snow grain scheme (Snowflake Schema).

For GPA-PTIK data, we use star schema to show the multidimensional model. Star schema based on a single fact table surrounded by one or more dimension tables as 'branches' so that look like stars. Each branching off at one of the tables of dimensions. Or in other words the table dimensions with this schema are all in the form of leaf or leaves and has no other branches.

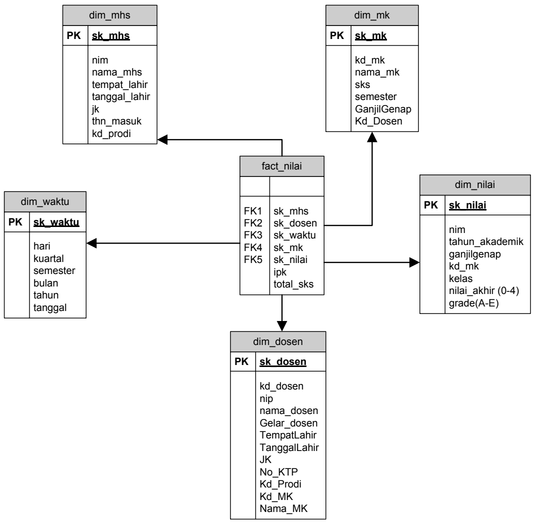

Fig. 1. Star schema of data warehouse GPA for PTIK study program

Pentaho Data Integration which called Kettle used for an integrated diversity of data available on PTIK study program in the database. In this case, MySQL is the database platform used. Below show all the dimension tables of data warehouse GPA for PTIK study program:

Dimension table Student (dim_mhs)
Data source of dim_mhs table is derived from a list of names of students who enrolled in the all semester of academic year since 2012. Where the total registered students is 1293. However, this data is only as names and NIM are stored in Excel files. While the data warehouse requires a complete student data, such as place of birth, date of birth, gender, year in, etc. Therefore, data collaboration with master data that is used in PDPT (Database University) needs to be done. The PDPT the data are the data Ms. Access. Merging the data source from Excel files and data on Ms.Access can be done using this PDI. Here is an overview transformation design to use PDI-Kettle.

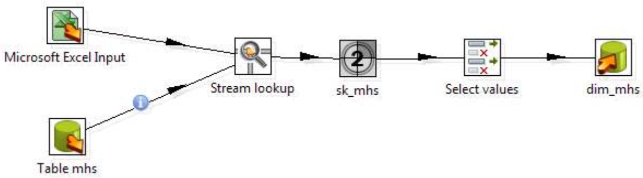

Fig. 2. The transformation to a table dim_mhs

Once the transformation is resulted student dimension tables as follows:

Table 1. Student Dimension Table

sk_mhs	NIM	Nama	TempatLahir	TanggalLahir	TahunMasuk	JK
1274	14215175	NIGEL NOLONDAM	TONDANO	19971004	2014	L
1275	14215176	DWIYANTI EKAPUTRI KANNA	PAPUSUNGAN	19961207	2014	P
1276	14215177	EUNIKE LOLENG	BAYUBOYA	19960117	2014	P
1277	14215179	IRFAN MOHAMAD ZEIN BINA	TASIKMALAYA	19960407	2014	L
1278	14215180	ANDRE E M RUMENGAN	JAKARTA	19960526	2014	L
1279	14215181	KARTIKA WIDYAYANTI TANAIJO	GORONTALO	19951115	2014	P
1280	14215182	GERALDO NDOELOE WINERUNGAN	PINALING	19960323	2014	L
1281	14215183	STIEFANUS BORONG	BUNA	19960509	2014	L
1282	14215184	KEVIN FRANSISCO KARANGAN	MANADO	19930922	2014	L
1283	14215185	HAYONO POSUMAH	MANADO	19931107	2014	L
1284	14215186	BISHY FERRY MAMENSHO	TOMOHON	19970201	2014	L
1285	14215187	RAENALDO STEFAN FEHR	TONDANO	19960923	2014	L
1286	14215188	GRAINIFE G.E MONINGKA	TONDANO	19960628	2014	L
1287	14215189	JOANFILY FREDRIKO LEMPAS	KAWANGKOAN	19960517	2014	L
1288	14215190	LEONARDI RONDONUWU	TINCEP	19960401	2014	L
1289	14215191	HERMON VAN DE MORRIS BURUMI	SERUI	19960217	2014	L
1290	14215192	INDRA DAVID PESIK	MOTOLING SATU	19950722	2014	L
1291	14215193	NELSON TENBAK	BEOGA	19930205	2014	L
1292	14215194	YOHANES EPERE	OTAKWA	19871006	2014	L
1293	14215195	ZAKEUS SEARO	PECELAMA	19871128	2014	L

Dimension table Lecturer (dim_dosen)
Data lecturer there in PTIK Study Program (Data Source) is the data stored in Excel files. Merging data is done to obtain data on the complete faculty.

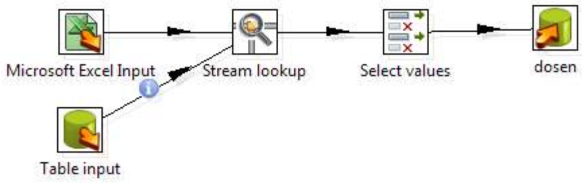

Fig. 3. Transformation for table dim_dosen

Once the transformation is resulted lecturer dimension tables as follows:

Table 2. Table data from dim_dosen

sk_dosen	[illegible]	[illegible]	Nama_Dosen	Gelar_Dosen	TempatLahir	TanggalLahir
[illegible]	[illegible]	[illegible]8321	BENYAMIN LIMBONG TAMPANG	M.Si	RANTEPAO	19530803
2	0006068403	-	GUIDO C KAINDE	ST., MM.,	Kauditan	19840606
3	0006108301	132320090	OLIVIA EUNIKE SELVIE LIANDO	S.T.	MANADO	19831006
4	0007055306	130900952	HERRY KRIEL POLII	Drs.	MINAHASA	19530507
5	0009027505	132308848	PEGGY VERONICA TOGAS	(NULL)	MANADO	19750209
6	0009077004	132206171	HERDY DJULY LIOW	S.T.	MINAHASA	19700709
7	0009108601	-	GLADLY CAREN RORIMPANDEY	ST, MISD	MANADO	19861009
8	0011116212	131786679	NONTJE JULTJHE PANGEMANAN	Dra.	MINAHASA	19621111
9	0012066206	131416580	WELMIENTJE SAHULATA	Dra.	TERNATE	19620612
10	0013017507	132326506	JANNE DEIVY TICOH	(NULL)	PANASEN	19750113
11	0013096211	131597516	RIO M ABAST	Drs.	TAHUNA	19620913
12	0014104906	130519425	JULES TICOALU TOGAS	(NULL)	MINAHASA	19491014
13	0014105106	130813536	ROBERT TJIA	M.Si	SORONG	19511014
14	0015125303	131468314	SAM DELFIE ROGAHANG	M.Pd	RATAHAN	19531215
15	0016036411	132321741	DJONI AGUSTAF	S.Pd	PALU	19640316
16	0016048304	(NULL)	VIVI P. RANTUNG	ST, MISD	(NULL)	19830416
17	0016016507	131769596	CHRISTINE T M MANOPO	Dra.	MANADO	19650116
18	0018095205	131597505	HENNIE EL MONOGINTA	M.Si	SORONG	19520918
19	0019117108	197111192008121001	RUDY HARIJADI WIBOWO PARDANUS	ST, M.Eng	Ujung Pandang	19711119
20	0020038701	(NULL)	MARIO T. PARINSI	S.Kom, MTI	(NULL)	19870320
21	0020075912	131599100	LUCKIE SOJOW	M.Pd	MINAHASA	19590720
22	0022066111	131697607	HISKIA K MANGGOPA	Drs.	KOTAMOBAGU	19610622
23	0024076604	132099012	JULIEN BIRINGAN	M.Si	MANADO	19660724
24	0025036009	131468313	JAMES JEFFREY SUMARAUW	Drs.	MINAHASA	19600325

Table Fact GPA (fact_nilai)
Fact tables are created in the transformation by combining the dimension tables that dim_dosen, dim_mk, dim_mhs, dim_nilai, dim_waktu who had previously designed.

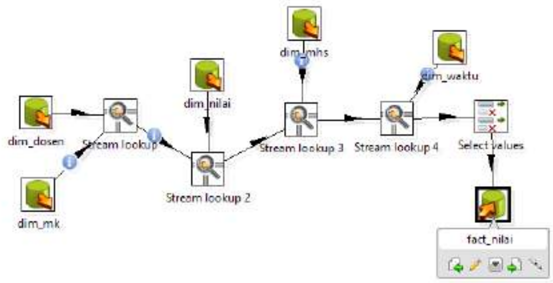

Fig. 4. Transformation ETL for table fact_nilai

Table fact_nilai after transformation is executed is as follows:

Table 3. Table Data fact_nilai

sk_mhs	sk_mk	sk_dosen	sk_waktu	sk_nilai
21	65	18	20160102	1
21	67	22	20160102	2
21	66	32	20160102	3
21	(NULL)	(NULL)	20160102	4
21	(NULL)	(NULL)	20160102	5
21	(NULL)	(NULL)	20160102	6
(NULL)	66	32	20160102	7
(NULL)	(NULL)	(NULL)	20160102	8
(NULL)	67	22	20160102	9
(NULL)	66	32	20160102	10
203	44	29	20160102	11
203	78	7	20160102	12
203	(NULL)	(NULL)	20160102	13
203	54	17	20160102	14
203	64	4	20160102	15
203	56	3	20160102	16
203	80	20	20160102	17
203	(NULL)	(NULL)	20160102	18
203	81	19	20160102	19
203	66	32	20160102	20
203	43	24	20160102	21

VII. Conclusion

Vertical information systems can help an organization address its ever-increasing uncertainties within the organization and with the outside environment. They become even more useful when the information base of an organization grows.

Several approaches can be used in designing BI systems. However, we feel the best approach is the process-oriented approach coupled with management commitment. Since management is the 'real' user of such systems and since the information at such aggregated, 'higher', levels are sensitive, full commitment of management should be harnessed to fully realize a vertical information system.

Going into the intricacies of the data oriented approach at an early stage of BI development will end up in a maze and will not address the key requirements of business people. In addition, such approach may suffer from data inconsistencies as data may be replicated by several processes.

Considering the above facts, the business oriented process provides a platform for consistent data publishing by the BI system. In addition, the business oriented approach ensures that data belonging to one business process will only be published once, irrespective of the number of organizational departments who will be using it.

Using a BI system like Pentaho for doing the ETL process to solve the problem of the diversity of data sources. After that, the data will be stored in a database called the datawarehouse. This an ongoing project for creating mdx-query, cube and dashboard of GPA-PTIK datawarehouse.

VII. References

[1] Kimball, Ralph., Ross, Margy., 2002, "The Data Warehouse Toolkit, Second Edition", John Wiley & Sons, Inc, Canada.
[2] Kimball, Ralph., Caserta, Joe., 2004, "The Data Warehouse ETL Toolkit", John Wiley & Sons, Inc, Canada. [Online].
[3] Thia Feris, 'Pentaho Knowledge Based", [online], (http://pentaho-en.phi-integration.com)
[4] [https://online.han.nl/sites/5-ICA-ALG/MasterMISD/Business%20Intelligence%20with%20OLAP/Presentations/business_intelligence.pdf
[5] http://www.rpi.edu/datawarehouse/dw-about-cmp.html
[6] Business Intelligence with OLAP study material-Data Vault 08-09. Pdf

Sensor Comparation for Smart Parking System

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***Abstract*—Parking is defined as a condition when a vehicle is temporary stored. In big cities, where parking space become more limited. In most of conventional parking system, driver enter parking area, search available parking space and park. Finding one available parking space can be frustrating, particularly during rush hour or weekend, since the driver did not know the exact available space. Some parking service provider count their available space and display it at the entry gate, but this system cannot detect vehicle that already leave the parking space until they reach the exit gate. With the system proposed, driver can reserve a parking space. When the vehicle arrives, sensor detects the vehicle and set the status to be occupied. Sensor can also detect when the vehicle leave the space so the space can be reserved for another vehicle. For the system proposed, four sensors are tested with three different positions. Sensor placement. Sensor proposed must be cheap, reliable, and require minimal maintenance.**

***Keywords*—Parallax PING, HC-SR04, Sharp GP2Y0A02YK0F, CT-SL110, parking**

I. Introduction

The growing number of car industry and increasing standard of mobilization leads to more affordable car price. From 2009 to 2013, number of cars has been increased 45%, and number of motorcycle has been increased to 61% [1]. This rapid growing number of cars causes issues in most of cities. It causes traffic congestion, air and noise pollution, and driver's frustration that can lead to bad behavior and accident. Parking issue became more serious because the limitation of parking space [2]. Price for parking spaces are rising up from time to time.

In most of conventional parking system, driver get a ticket at the entry gate, find available parking space and park, go to exit gate, present or return the ticket, pay the parking fee and get the receipt. Finding one available parking space can be frustrating, particularly during rush hour or weekend [2]. The problem become more complicated because some parking service provider still enter the vehicle even though the parking lot was already full, with expectation during the searching process, there are other users who come out so the empty space can be used by the other vehicle. Some parking service provider count their available space and display it at the entry gate, but this system cannot detect vehicle that already leave the parking space until they reach the exit gate. Available space displayed at the entry gate did not represent real situation. In big or multi-level parking lot, this can cause opportunity loss for people who want to park, as well as profit loss to the provider. While some of parking service provider offer reservation, this system only suitable for scheduled trip. Many of this system also require customer to pay deposit or charge at higher rate.

Previous research describe smart parking system based on secured wireless network using sensor communication to acquire high parking utilization and finding free parking space [2]. Further research use sensor to detect free parking space [4], and driver can reserve free space by SMS [4] [5]. To identify vehicle plate number and other identification, cameras are installed nearby parking lot [6]. Ultrasonic sensor was used to detect vehicle for parking location [7].

The basic operation for the proposed parking system is described as: When a vehicle arrive at the entry gate, a monitor shows the parking lot map. The driver then choose available parking space. System then flag selected space to 'reserved'. When the car enter the designated space, it trigger the sensor and system flag selected space to 'occupied'. When the car exit, sensor release the flag in the system so it can be used for another car. Sensor proposed must be cheap, reliable, and require minimal maintenance.

II. Sensors

A. Parallax PING))) Ultrasonic Sensor

The parallax PING (PING) sensor is an ultrasonic sensor that provide non-contact measurement. The sensor works by transmitting ultrasonic burst signal at 40 kHz (chirp).

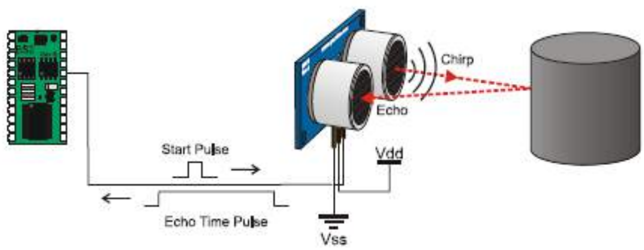

Fig. 1. Parallax Ping Sensor

This signal then travel through the air and if it hits an object, the signal would bounce back echo to the sensor [3]. The sensor uses +5 VDC power supply with 30-35 mA current.

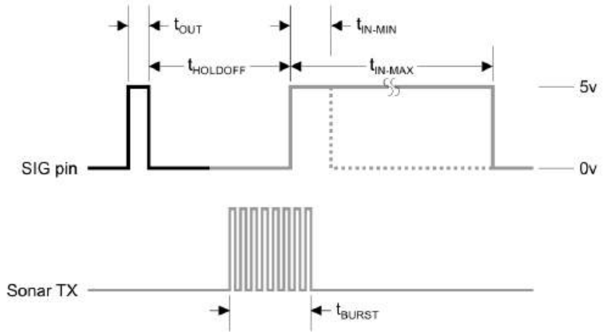

Fig. 2. Parallax Ping Clock Diagram

This sensor has a dimension of 21.3 mm by 45.7 mm and has a thickness of 15.3 mm. This dimension makes it easy to be hidden at the parking space. Price for this sensor varies from IDR.450,000 to IDR.500,000 (US$.31.5 to US$.35.0)

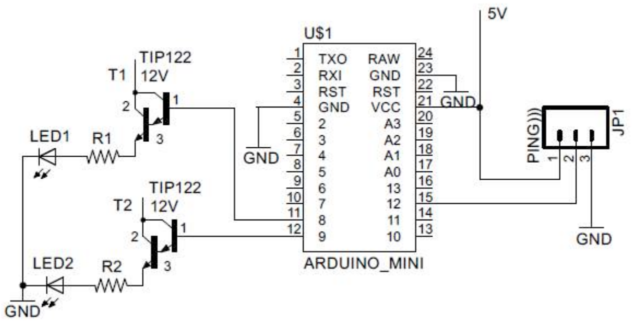

Fig. 3. Parallax Ping Schematic Diagram

This sensor, however has some limitation. Range is limited from 2 cm to 3 meters from the object. Sensor may also fail if the surface of the object detected is reflective , while positioned at shallow angle.

B. HC-SR04 Ultrasonic Sensor

Like the PING, HC-SR04 is an ultrasonic sensor that operates at 40 kHz. It also uses +5 VDC but at lower current, 15 mA [4].

Fig. 4. HC-SR04 Sensor

The sensor has identical dimension with PING, but it has additional pin for Echo. Unlike the PING, this sensor is much more affordable at IDR.17,500 (US$.1.22).

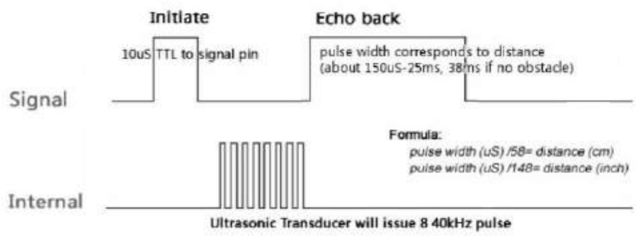

Fig. 5. HC-SR04 Clock Diagram

HC-SR04 can detect object within 2 cm to 4 meters. The sensor also shares similar limitation to detect object at shallow angle.

C. Sharp GP2Y0A02YK0F Infrared Sensor

The Sharp GP2Y0A02YK0F is a distance measuring sensor unit, composed of an integrated combination of PSD (position sensitive detector), IRED (infrared emitting diode) and signal processing circuit. The sensor adopts triangulation method so the variety of the reflectivity of the object, the environmental temperature and the operating duration are not influenced easily to the distance detection [5].

The sensor outputs the voltage corresponding to the detection distance. The sensor uses 4.5 to 5.5 VDC power supply with 33 mA current.

Fig. 6. Sharp GP2Y0A02YK0F Sensor

This sensor has a dimension of 29.5 mm by 13 mm and has a thickness of 21.6 mm. This is bigger than the two previous sensors. Price for this sensor is around IDR.175,000 (US$.12.24).

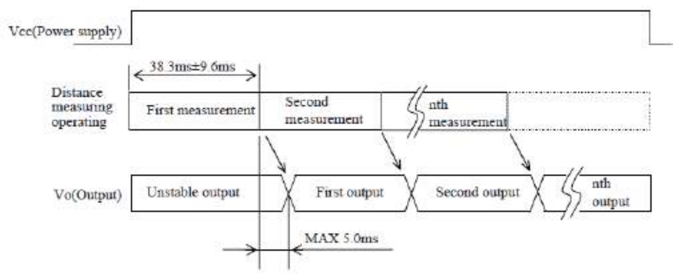

Fig. 7. Sharp GP2Y0A02YK0F Clock Diagram

This sensor uses lens that needs to be kept clean. Range is limited to 20 cm to 150 cm from the object.

D. CT-SL110 Single Vehicle Loop Detector

The device is connected to an inductive loop mounted on the road. The device can detect when a vehicle pass over the loop.This device cost around IDR.700,000 (US$.49.00).

Fig. 8. CT-SL110

The device can use 220/110 VAC or 24/12 VDC with 4.5 W power rating. The device has a dimension of 75 x 37 x 110 mm, and reaction time of 10 ms [10] [11].

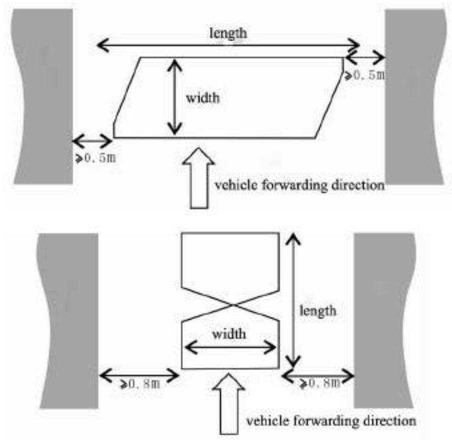

Fig. 9. CT-SL110 Detection Process

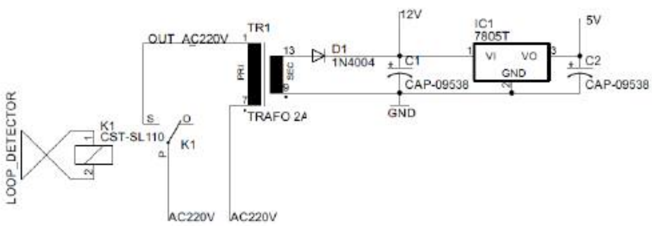

Fig. 10. CT-SL110 Shematic Diagram

III. DETECTION SCHEME

A. Parking Space

Parking is defined as a condition when a vehicle is temporary stored. Area where a vehicle park is called parking space [6]. This space, or called *Satuan Ruang Parkir* (SRP) is measured by the effective area for one vehicle to be stored, including surrounding free area and space needed to open the door [7]. Parking area for each vehicle type is regulated as follow.

Type	Parking Area (m)
Passenger Vehicle Type I	2.30 x 5.00
Passenger Vehicle Type II	2.50 x 5.00
Passenger Vehicle Type III	3.00 x 5.00
Bus/Truck	3.40 x 12.50
Motorcycle	0.75 x 2.00

For the experiment, we use Passenger Type 1 SRP (2.30 x 5.00) area.

B. Sample Car

There are 89 passenger cars from 13 manufacturers sampled. Main criteria for the sample was their availability and popularity for Indonesian market.

TABLE I. VEHICLE SAMPLE

Manufacturer	No. of Vehicle Sampled
BMW	11
Chevrolet	3
Daihatsu	6
Datsun	2
Ford	4
Honda	10
Hyundai	5
Isuzu	1
Kia	6
Mercedes Benz	8
Nissan	7
Suzuki	7
Toyota	19

1) Ground Clearance

Ground clearance is defined as the distance between the lower parts of the vehicle to the ground. The tires are not considered to this definition as they are designed to be in contact with the ground. Ground clearance needs to be considered if the sensor is placed under the vehicle.

Cars grouped into their class to determine the average ground clearance.

TABLE II. AVERAGE GROUND CLEARANCE

Vehicle Type	Average Ground Clearance (mm)
Small Hatchback	169.50
Hatchback	154.00
Minivan/SUV	183.41
Sedan	137.89

Vehicle Type	Average Ground Clearance (mm)
Full Size Sedan	149.50
Pickup Truck	202.17
Average	166.08

2) Car Height

It is possible to place the sensor above the car, so average car height is also calculated.

TABLE III. AVERAGE CAR HEIGHT

Vehicle Type	Average Height (mm)
Small Hatchback	1,530.83
Hatchback	1,518.43
Minivan/SUV	1,741.97
Sedan	1,434.59
Full Size Sedan	1,488.50
Pickup Truck	1,915.71
Average	1,605.01

Most of parking building has no more than 3 meters in height, so average distance between the roof to the car will be 1,350 mm.

C. Sensor Location

Testing was done at outdoor parking space. For each sensor, measurement was conducted 3 times at different place:

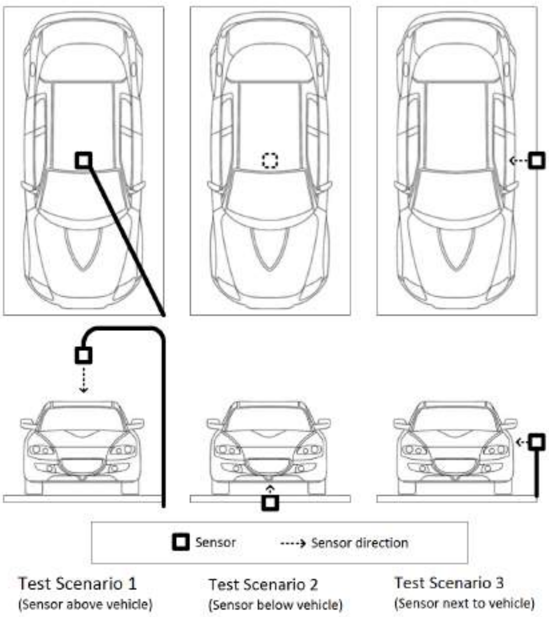

Fig. 11. Sensor Location Scenario

- Above testing vehicle, with sensor pointed down directly to the vehicle (270 degree to the horizon)
- Below testing vehicle, with sensor pointed up directly to the vehicle (90 degree to the horizon)
- Next to testing vehicle , with sensor pointed directly to the vehicle (180 degree to the horizon)

Sensor then placed at different distance with test vehicle to test its detection capability. The distance was set at 10 cm step, starting from 10 to 150 cm, except for test scenario 2 and 3, which is limited to 100 cm to the distance, since further than that, it is considered too far.

IV. TEST RESULT

Below are test result for each scenario. One indicated that the sensor could detect the car as zero indicated that the sensor cannot detect the car.

TABLE IV. TEST RESULT FOR PARALLAX PING SENSOR

No	Distance (cm)	Test Sc. 1	Test Sc. 2	Test Sc. 3
1	10	1	1	1
2	20	1	1	1
3	30	1	1	1
4	40	1	1	1
5	50	1	1	1
6	60	1	1	1
7	70	1	1	1
8	80	1	1	1
9	90	1	1	1
10	100	1	1	0
11	110	1		
12	120	1		
13	130	1		
14	140	1		
15	150	1		

TABLE V. TEST RESULT SHARP IR SENSOR

No	Distance (cm)	Test Sc. 1	Test Sc. 2	Test Sc. 3
1	10	1	1	1
2	20	1	1	1
3	30	1	1	1
4	40	1	1	1
5	50	1	1	1
6	60	0	1	1
7	70	0	0	0

No	Distance (cm)	Test Sc. 1	Test Sc. 2	Test Sc. 3
8	80	0	0	0
9	90	0	0	0
10	100	0	0	0
11	110	0		
12	120	0		
13	130	0		
14	140	0		
15	150	0		

TABLE VI. TEST RESULT FOR HC-SR04 SENSOR

No	Distance (cm)	Test Sc. 1	Test Sc. 2	Test Sc. 3
1	10	1	1	1
2	20	1	1	1
3	30	1	1	1
4	40	1	1	1
5	50	1	1	1
6	60	1	1	1
7	70	1	1	1
8	80	1	1	1
9	90	1	1	1
10	100	1	1	1
11	110	1		
12	120	1		
13	130	1		
14	140	1		
15	150	1		

TABLE VII. TEST RESULT FOR CT-SL110 SENSOR

No	Distance (cm)	Test Sc. 2
1	10	1
2	20	1
3	30	1
4	40	1
5	50	1
6	60	1
7	70	1
8	80	1
9	90	1
10	100	1

V. CONCLUSIONS

HC-SR04 and PING sensor have identical performance, but HC-SR04 is more economical in price. Both can be placed above or under the vehicle, but it is more maintenance-less if placed above the vehicle. Sensor placed under the vehicle can be blocked by trashes, prone to liquid such as oil or water from radiator/air conditioner. At indoor parking, the sensor can be attached to the roof. For outdoor parking, under-vehicle sensor considered to be more esthetically better. When using this scenario, proper housing, sealing and routine maintenance should be performed.

CT-SL110 Sensor is more robust for detecting vehicle. It still can detect the vehicle even the sensor covered with object (simulating trash / oil / water). This sensor can be used for indoor or outdoor parking. Even though, this sensor require more power and more expensive than the other sensor tested.

All the sensors tested have response time under one second, but it is recommended that when a vehicle leave the parking space, the system have a delay around 10 to 20 seconds before setting the parking space's status to free. It is necessary to make sure that the vehicle really leaves the space.

ACKNOWLEDGMENT

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REFERENCES

[1] Badan Pusat Statistik, "Perkembangan Jumlah Kendaraan Bermotor Menurut Jenis tahun 1987-2013," [Online]. Available: http://www.bps.go.id/linkTabelStatis/view/id/1413.

[2] A. A. G. Agung, "Design of Motor Vehicle Insurance Policy Management Application," *Tokyo International Conference on Engineering and Applied Science,* pp. 88-97, 2014.

[3] T. Lie Ing and H. Hadian Yanuardi, "Tinjauan Kepuasan Pelanggan Terhadap Kualitas Pelayanan Parkir Basement di Pusat Perbelanjaan Bandung Supermall," *Jurnal Teknik Sipil Universitas Kristen Maranatha,* pp. 174-193, 2007.

[4] G. Yan, W. Yang, R. D.B. and O. S., "Smart Parking: A Secure and Intellegent Parking System," *Intelligent Transportation System Magazine, IEEE,* vol. 3, no. 1, pp. 18-30, 2011.

[5] P. V. Kumar and S. T. S., "A Prototype Parking System using Wireless Sensor Networks," *International Journal of Computer Communication and Information System (IJCCIS),* vol. 2, no. 1, 2010.

[6] S. C. Khang, T. J. Hong, T. S. Chin and S. Wang, "Wireless Mobile-Based Shopping Mall Car Parking

System (WMCPS)," *Services Computing Conference (APSCC), IEEE Asia-Pacific,* pp. 573-577, 2010.

[7] H.-C. Tan, J. Zhang, X.-C. Ye, H.-Z. Li, P. Zhu and Q.-H. Zhao, "Intelligent Car-Searching System for Large Park," *International Conference on Machine Learning and Cybernetics,* vol. 6, pp. 3134-3138, 2009.

[8] M. I. Sari and R. Handayani, "Usulan Sistem Informasi Lokasi Parkir dengan Menggunakan Mikrokontroller, Sensor Ultrasonik dan Jaringan LAN," *Seminar Nasional Teknologi Informasi & Komunikasi,* vol. 2, pp. 6-8, 2014.

[9] Parallax Inc., "PING))) Ultrasonic Distance Sensor," [Online]. Available: www.parallax.com.

[10] ElectFreaks, "Ultrasonic Ranging Module HC - SR04," [Online]. Available: http://www.electroschematics.com/8902/hc-sr04-datasheet/.

[11] Sharp Corp., "GP2Y0A02YK0F - Sheet No. E4-A00101EN," [Online]. Available: https://www.sparkfun.com/products/8958.

[12] Tenet Technologies, "Vehicle Loop Sensor TLD-110/410 Series," [Online]. Available: http://www.tenetparking.com/Products/TLD110410Series.html.

[13] ShenZhen Tenet Technology Co.,LTD, "One Channel Inductive Vehicle Loop Detector for Automatic Barrier Gate TLD-110," [Online]. Available: http://www.vehicleparkingsystem.com/sale-3922422-one-channel-inductive-vehicle-loop-detector-for-automatic-barrier-gate-tld-110.html.

[14] Menteri Perhubungan, *Keputusan No. 66 Tahun 1993 Tentang Fasilitas Parkir Untuk Umum,* Jakarta, 1993.

[15] *Keputusan Direktur Jendral Perhubungan Darat No. 272/HK.105/DRJD/96,* Jakarta: Departemen Perhubungan, 1996.

Throughput Evaluation in LTE-Advanced Network Access Using Carrier Aggregation

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Abstract—**This paper proposes throughput evaluation for LTE-advanced network access in the particular service area employing carrier aggregation (CA) technique. From several kind of CA, we evaluate which one the best CA configuration in term of network throughput. 30 MHz aggregated bandwidth and two component carrier (CC) serve as a primary serving cell (PCell) and secondary serving cell (SCell) are used in the simulation. Simulation results show that the configuration of wider PCell bandwidth resulting better performance of the throughput. In particular, in CA intra-band scenario, the throughput of non-contiguous CA has better performance compared with CA contiguous. Whereas in the inter-band CA scenario, CC combination with lower frequency resulting better throughput performance.**

***Keywords*–Long Term Evolution (LTE)-Advanced, carrier aggregation, component carrier, PCell, SCell, throughput.**

I. INTRODUCTION

To meet the demand for high speed services in cellular communication, the 3rd Generation Partnership Project (3GPP) has developed the technology of Long Term Evolution (LTE). LTE has been specified starting in Release 8 (R8) in a year of 2008 and continued to Release 9 (R9) in 2009. It doesn't end there, the development of LTE has been continuing to reach the very high speed data services. In 2012, LTE-Advanced has been developed as the standard to include a new technique that is called carrier aggregation (CA) [1]-[3]. CA is a novel scheme that combines multiple frequency spectrum to increase the bit rate. CA was first introduced in LTE-Advanced Release 10 (R10). With the maximum spectrum utilization in CA technique that is 100 MHz, LTE R10 has reached 1 Gbps data rate for downlink and 500 Mbps for uplink. However, in order to keep backward compatibility with R8 and R9 UEs, the aggregation is based on R8/R9 carriers. Carrier aggregation can be used for FDD and TDD as well.

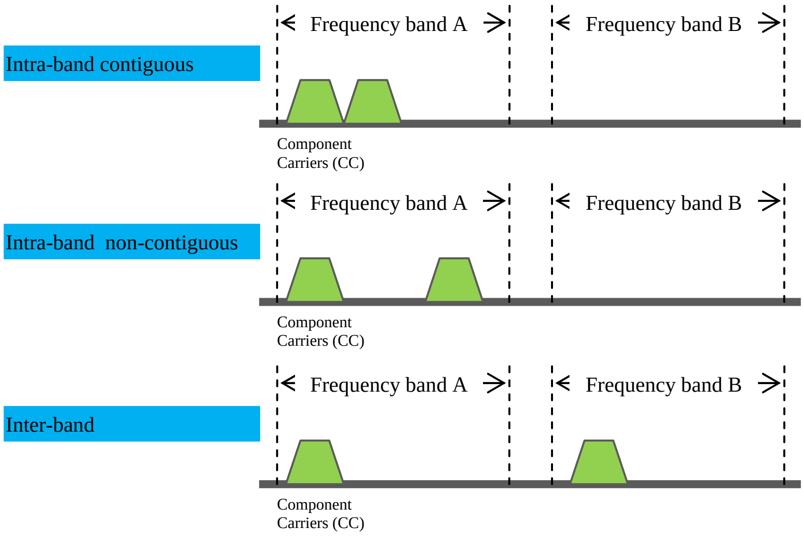
In the CA concept, each aggregated carrier is referred to as a component carrier (CC). The CC's bandwidth can vary from 1.4, 3, 5, 10, 15 or 20 MHz. The maximum number of CC can be aggregated is of five component carriers. Therefore the maximum aggregated bandwidth is 100 MHz. Furthermore, in FDD the number of aggregated carriers can be different in DL and in the UL. However, the numbers of uplink CC is always equal to or lower than the number of downlink CC. The individual CC can also be of different bandwidths. For TDD the number of CCs as well as the bandwidths of each CC will normally be the same for DL and UL. Once spectrum frequency has been allocated to the network operator, they have to combine with their existing frequency. Therefore, one of three kinds of CA might be happened, i.e. intra-band contiguous, intra-band non-contiguous, or inter-band non-contiguous. In the real network implementation we don't know which one has the best performance, for example the performance of user throughput.

There is not so much research publications revealed that discussed related to CA throughput performance. One of them is proposed in [4]. This paper deals with field experimental results on the block error rate (BLER) of uplink control information (UCI) and the downlink throughput performance of carrier aggregation (CA). Bandwidth is an asymmetric between the downlink and uplink and the center carrier frequencies of 3.92625 and 3.67125 GHz in the downlink and uplink, respectively. They obtain more than 100 Mbps of throughput based on field measurement. However, the operating frequency, the bandwidth symmetry between uplink and downlink, and the CC combination of CA might not the same in the other case. The specific service area, the number of eNB, the operating frequency, the frequency bandiwdth, and the number of users that belong to the result of LTE-Advanced network planning based on historical data were proposed in our contribution. The objective is to obtain the throughput performance among those of available CA technique. In addition to the frequency combination, we will also observe throughput based on combination of primary and secondary component carrier bandwidth. This study was based on the access network planning case study in the typical urban environment by using computer simulation.

The rest of the paper is as follows. In Section II we explain the concept of carrier aggregation in LTE advanced. Section III presents the process of LTE-Advanced network planning, Section IV presents simulation result and evaluation, and finally conclusion is drawn in Section V.

II. CARRIER AGGREGATION CONCEPT

Today in most of the country, there are almost none of a cellular service provider owns continuous spectrum of 100 MHz. This because of the spectrum scarcity and the number



of service provider relatively many in a country. Therefore, it emerges the new method called CA technique. There are three different modes of CA, namely intra-band contiguous, intra-band non-contiguous, and inter-band CA. 3GPP's radio access network (RAN) working group 4 (RAN 4) is responsible for setting performance requirements of CA technique which initially limits the aggregation up to two component carriers only. With this method, one UE is able to consume up to 100 MHz spectrum bandwidth in total.

In CA technique each carrier or channel, which is aggregated, is called component carriers (CCs). Each CCs might have bandwidth of 1.4, 3, 5, 10 or 20 MHz. The maximum number of channels that can be aggregated is five, so the maximum amount of bandwidth that can be used is 100 MHz. The number of aggregated carriers can different between downlink and uplink, but the number of uplink CCs never more than CCs downlinks [6]. Each CCs can also have different bandwidth. The CA technique is a way of how to aggregate some of the CCs. As shown in Figure 1, three techniques or types of CA on LTE-A are described and can be explain as follows.

– Intra-band contiguous CA: the method using two or more channels in one frequency band sequentially.
– Intra-band non-contiguous CA: the method to aggregating channels within a single frequency band, but there is a distance between CC's.
– Inter-band non-contiguous CA: the method to aggregating different bands of frequency channels, it is useful if there is fragmentation band.

In the CA system, there are two types of serving cell, those are Primary Serving Cell (PCell) and Secondary Serving Cell (SCell). While PCell have the task to bring Primary Component Carrier (PCC) downlink and uplink and handle the Radio Resource Control (RRC) connection, SCell have the task to bring the Secondary Component Carrier (SCC). There is only one PCC in the CA system, but the SCC could be more than one. Coverage of PCell and SCell may vary depending on the frequency it is being used.

In this work, we assume that there are several LTE-A network scenarios with FDD duplexing schemes. Two CCs and the amount of aggregated bandwidth on each CA configuration is 30 MHz. We use frequency combination that follows some CA configurations determined by 3GPP release 10, 11, and 12, i.e. 3GPP E-UTRA Carrier Aggregation (TS.36.101) [5]-[7].

III. LTE-Advanced Network Planning

A. Coverage Dimensioning

Radio network planning is an initial process in our work. The process begins with the coverage dimensioning in the particular environment, as a tested urban area to evaluate CA performance. The digital map of that area is uploaded to the network planning simulator and then all parameters are considered and taken into consideration in the simulation. Radio link calculation is a main task in the coverage dimensioning to estimate the value of the maximum allowable path loss (MAPL) and radio signal received power (RSRP) threshold. In this paper we use multiple frequency configurations, therefore several different parameters are used in the calculation. All the parameters used in the radio link budget simulation are summarized in Table 4 and Table 5. In the first step, we calculate link budget using simple formula to find the number of site required to cover the tested simulated area.

Receiver sensitivity is calculated by using the following equation and the value is in Table I.

$$S_R = -174 + 10\log\left(15.000\, x\, 12\, x\, N_{rb}\right) + NF + SINR \quad (1)$$

where N_{rb} denotes number of Resource Block, NF is a noise figure of the receiver, and SINR denotes signal to inteference plus noise ratio. From all those parameters, MAPL and RSRP threshold value can be calculated by using the following

TABLE 1. TX-RX PARAMETERS USED IN COVERAGE DIMENSIONING.

Parameters	Downlink	Uplink
Transmitter End (eNB)		
Max TX Power (dBm)	43	23
Number of Tx Antenna	2	1
Tx Diversity Gain (dB)	3	0
Feeder Loss per length (dB/m)	0.06	-
Feeder Length (m)	35	-
Feeder Loss/Body Loss (dB)	2.1	1
Connector Loss (dB)	0.5	-
Receiver End (UE)		
Noise Figure (dB)	8	4
Required SINR at Cell Edge (dB)	-4.13	-5.11
Fast Fade Margin (dB)	4.5	4.5
Interference Margin (dB)	5.5	4.5
Body Loss/Feeder Loss (dB)	1	2.1
Additional Gain (dB)	0	3

equations.

$$MAPL = P_T + G_T + G_R - L_S - S_R \tag{2}$$

$$RSPP = EIRP_{per\ subscriber} - MAPL_{\min} + shadowing \tag{3}$$

where PT denotes eNB power transmit, GT is transmit antenna gain, GR is receive antenna gain, Ls denotes an interference margin, and SR denotes receiver sensitivity. MAPL minimum of all scenarios will be converted into cell radius using Okumura-Hatta propagation model provided by (4).

$$\begin{aligned} P_L(dB) = {} & 69.55 + 26.16\log(f_c) - 13.82\log(h_{te}) \\ & - a(h_{re}) + (44.9 - 6.55\log(h_{te}))\log d \end{aligned} \tag{4}$$

where f_c denotes carrier frequency used in LTE, h_{te} denotes eNB antenna height, h_{re} denotes user antenna height, and d is the cell radius. For analytical purpose we assume a hexagon cell shape which is comprised of 3 sectors, therefore eNB cell area can be calculated by (5).

$$L_{site}(km^2) = 1.95\,x\,d^2 \tag{5}$$

and the number of calculated eNB is

$$N = \frac{L_{design\ area}}{L_{site}} \tag{6}$$

By using all those parameters in Table 1 and equations (1) to (6), we are able to calculate the required number of total site (eNB) in the tested area. Up to this step, we have done the coverage planning in the tested area so that the number of eNB within the specified parameter complies to LTE technical specification has been derived and described in Figure 2. The In the capacity planning process, we have assumed that LTE subscribers are predicted for the next 5 years. Therefore, the network throughput and cell throughput are determined using this predicted traffic calculation. The data which are needed include population data obtained from statistical municipal city office [8], the population growth in the city is assumed to be 1.1% per year [9], mobile penetration is 120% [10], LTE penetration is 3% [9], and the percentage of subscribers belong to particular provider is 42.1%.

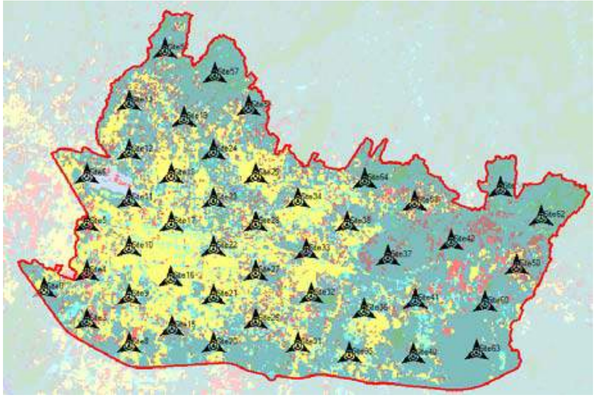


Fig. 2 eNB position on a digital map of a tested area.

In the network design process, the total site number is determined from the highest number of coverage or capacity dimensioning in order to meet both aspect. In the process of laying site, the result takes 46 eNB to cover the entire city of the tested area which can be seen in Fig. 2. Each site has three sectors, and each sector consists of two transmitters with the same azimuth, one transmitter for PCell and the other for Scell. To examine the network throughput and the network BLER in the tested area, we have to know RSRP and CINR inside the tested area.

IV. SIMULATION RESULT AND EVALUATION

The overall result prediction of RSRP and CINR can be seen in Fig. 3. Based on simulation using the above parameters, the percentage of average RSRP in the tested area is above 95% for all scenarios of A1 to A3, B1 to B3, and C1 to C6. We found the value of average RSRP is above -81.29 dBm with the minimum value of -100 dBm in all scenarios. The CINR minimum that specified in the network of LTE-A is -6.5 dB. Based upon that specification, our simulation has shown the coverage for downlink CINR prediction is 95% whereas for the uplink is 96%. From the results of these predictions, the designed network has been meet the target to be able to test the throughput and BLER value of the implementation of CA techniques in LTE-Advanced networks.

Now we are evaluating the network throughput and BLER use Monte-Carlo simulation method, which its results closer to reality because this method generate users based on the data traffic. The user follows the distribution of traffic maps as shown in Fig. 4, and the weight to be considered according to the Poisson distribution is calculated by weighting distribution in Fig. 5. Simulations performed 5 times for each scenario with the number of iterations in simulations is up to 100 times.

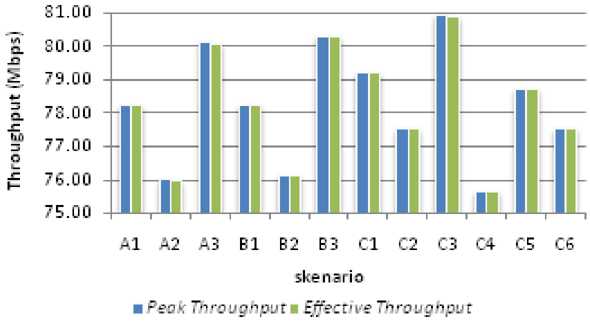


Fig. 7 Downlink throughput of whole scenarios.

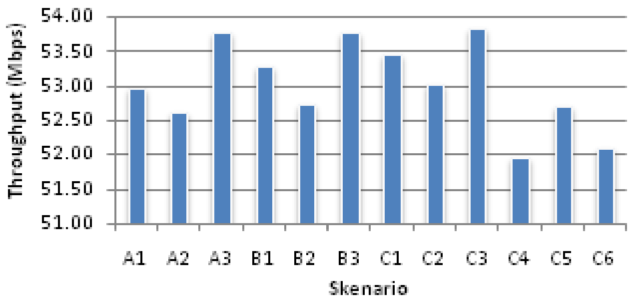


Fig. 8 Uplink throughput of whole scenarios.

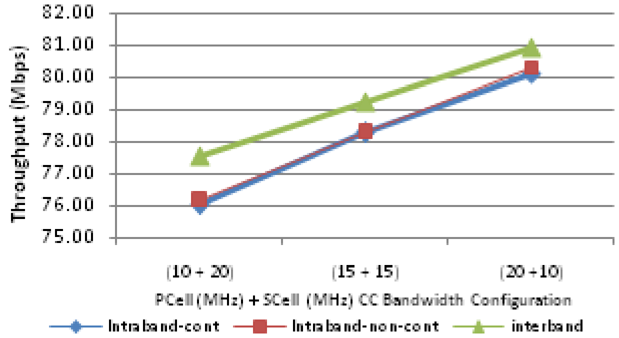


Fig. 9 Downlink throughput based bandwidth PCell dan SCell.

The results of the throughput and BLER of all the simulation will be averaged.

The results of Monte Carlo simulations for downlink BLER values shown in Fig. 6. For the uplink, BLER value is zero for all scenarios, so that the uplink effective throughput is not different from the peak throughput. In the downlink side there is a transmission error which has value ranges from 0.6% to 0.7%. In the intra-band CA only uses the 1800 MHz frequency (scenario A1-B3) downlink BLER is around 0.67% to 0.71%, while the inter-band CA with a combination of 700 MHz and 1800 MHz (scenario C1 - C3) has BLER value slightly lower, at around 0.63% to 0.65%. For inter-band CA which one of its frequency is higher than others (scenario C4-C6), its BLER value is also higher, about 0.69% up to a maximum of 0.79% at scenario C5 (CC 700 MHz and 2600 MHz), the difference reached 0.16% with scenario C2 which its downlink BLER is the lowest of all scenarios.

The difference between the effective throughput with peak throughput on the downlink is not far, highest difference is only 38 kbps, it can be seen in Fig. 7. From the simulation results in Figs. 7 and 8 are known that the downlink throughput is ranging from 75 to 80 Mbps, and uplink throughput is 51 to 53 Mbps. In CC configuration always select lower channel frequency for PCell than SCell, so that PCell coverage is larger than SCell coverage. It is known that a wider channel bandwidth have more RB, so that throughput is even higher. So, with a wider channel bandwidth and lower frequency channels, it will produce higher network throughput because larger signal coverage, as shown in Fig. 9. Thus, although the number of aggregated the same carrier, the CA configuration which has wider PCell bandwidth will result higher throughput.

Average increase in the throughput of bandwidth combination PCell and SCell 10 MHz +20 MHz to combination 15 MHz +15 MHz is 2.01 Mbps for the downlink and 459.61 kbps for uplink. While the average increase of combination 15 MHz + 15 MHz to 20 MHz + 20 MHz is 1.87 Mbps on the downlink and 539.3 kbps on the uplink. So, an increase in throughput of PCell bandwidth 10 MHz to 20 MHz in the downlink can reach 3.88 Mbps, and 998.9 kbps on the uplink. Furthermore, if based on the type of CA, The graph in Fig. 10 shows that in the intra-band CA, throughput for intra-band non-contiguous is higher than throughput for intra-band contiguous. This is due to the intra-band contiguous CA channel used to be very close together, giving rise to the presence of adjacent channel interference, which makes getting lower SINR value. The greater the SINR value, the higher the modulation type used, so the bearer channel coding rate and its efficiency higher, and resulting the higher throughput because the number of transmitted bits per symbol is more.

According to Okumura-Hatta propagation model in equation 3, the lower frequencies will have a smaller pathloss, so that at the same distance from the transmitter, it would produce a better radio channel condition for the user. The better the condition of the channel, the greater SINR and efficiency bearer resulting higher throughput. Scenario C1 and C5 both use frequencies of 700 MHz, but the frequency of 1800 MHz C5 replaced with a higher (2600 MHz). As a result, the throughput decreased 0.59 Mbps, from 79.24 Mbps to 78.75 Mbps on the downlink and 0.76 Mbps on the uplink. Furthermore, in scenario C6, the frequency of 700 MHz C1 scenario changed to 2600 MHz. The decrease in throughput from C1 to C6 is greater than C1 to C5, which is 1.68 Mbps on the downlink and 1.39 Mbps on the uplink. Throughput of inter-band CA scenario C6 even lower 0.72 Mbps from intra-band with the same bandwidth combinations, those are scenario A1 and B1 which only uses the 1800 MHz frequency band.

In PCell and SCell bandwidth combination 10 MHz + 20 MHz, there is a scenario C2 (700 MHz and 1800 MHz) and C4 (900 MHz and 2100 MHz), wich have throughput respectively 77.56 Mbps and 75.64 Mbps for downlink, and 53.03 and 51.09 for the uplink. Just like the combination of bandwidth 15 MHz + 15 MHz, this time the scenario that uses lower frequencies, ie C2, resulting better throughput than the scenario C4 that uses higher frequencies. The difference in throughput of these two scenarios on the downlink direction reaches 1.92 Mbps, and uplink up to 1.08 Mbps.

V. CONCLUSIONS

Based on the LTE network access design in this study, we can conclude the following. We found that the throughput from the network simulation, CA configuration with wider PCell bandwidth result better throughput. Then, based on type of CA in the intra-band, the throughput non-contiguous CA is better than contiguous CA. And the last, in the inter-band CA, CC combination with a lower frequency produces higher throughput. CA configuration that produces the highest throughput in the design area the city is inter-band non-contiguous CA with the combination of 700 MHz and 1800 MHz, and the bandwidth combination of the primary and secondary serving cell 20MHz + 10MHz, the throughput is 80.95 Mbps on the downlink and 53.83 Mbps on the uplink.

REFERENCES

[1] Dahlman, Erik., Stefan Parkvall dan Johan Sköld. *4G LTE/LTE-Advanced for Mobile Broadband.* Oxford : Elsevier, 2011.

[2] R. Ratasuk, D. Tolli, and A. Ghosh, "Carrier Aggregation in LTE-Advanced," in Proceeding of IEEE 71st Vehicular Technology Conference (VTC 2010-Spring), vol. 1, pp. 1-5, 2010.

[3] M. Iwamura, K. Etemad, F. Mo-Han, R. Nory, and R. Love, "Carrier aggregation framework in 3GPP LTE-advanced [WiMAX/LTE Update]," in IEEE Communications Magazine, vol. 48, issue 8, pp. 60-67, 2010.

[4] Kakishima, Y., Kawamura, T. Kishiyama, Y. Taoka, and Nakamura, T., "Experimental Evaluation on Throughput Performance of Asymmetric Carrier Aggregation in LTE-Advanced," in Proceeding of IEEE 73rd Vehicular Technology Conference (VTC Spring), vol. 1, pp. 1-5, 2011.

[5] Hua Wang, C. Rosa, and K. Pedersen, "Performance Analysis of Downlink Inter-Band Carrier Aggregation in LTE-Advanced," in Proceedings of IEEE Vehicular Technology Conference (VTC Fall), pp. 1-5, 2011.

[6] ETSI TS 136 101, v10.3.0, "LTE, E-UTRA, and UE Radio Transmission and Reception Technical Specification," June 2011.

[7] Ran Zhang; Zhongming Zheng; Miao Wang; Xuemin Shen; Liang-Liang Xie, "Equivalent capacity analysis of LTE-Advanced systems with carrier aggregation," in Proceedings of IEEE ICC Conference, Pp. 6118 - 6122, DOI: 10.1109/ICC.2013.6655582, 2013.

[8] Website BPS Kota Bandung [Online]. Available : http://www.bandungkota.bps.go.id/

[9] Denny Kusuma Hendraningrat et al., "Refarming of Frequency 700 MHz Analysis for Long Term Evoltion (LTE) in Indonesia Using Link Budget Calculation," in International Conference n Electrical Engineering and Informatics, 2011.

[10] Website APJII [Online]. Available: http://www.apjii.or.id/v2/index.php/read/page/halaman-data/9/statistik.htm.

PAPR REDUCTION BY DATA SUBCARRIERS AND NULL SUBCARRIERS SWITCHING

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Abstract-**One of OFDM main problem is amount of PAPR (Peak Average Power Ratio). The magnitude of PAPR caused the summation of power by the wave in phase so that maximum power compared to average power becomes high. One method to reduce the amount of PAPR is switching null subcarrier with the data subcarrier. This method has advantages over other methods that do not damage the transmitted signal. BER is not increasing and does not require any side information to restore the signal to its original data. The problem in the switching method is the complexity of the system. In this paper will offered a method based on switching data subcarrier null subcarrier. The new method will provide lower complexity and the PAPR results lower than original OFDM but still higher than switching the data subcarrier null subcarrier method.**

Keyword : OFDM, Null subcarrier data subcarrier switching, Data subcarrier null subcarrier switching, PAPR

I INTRODUCTION

The communication now is growing very fast. telecommunication must be able to transmit bigger dan faster data. Data communication in large quantities and fast called broadband. Broadband today has become the people's needs. Therefore, emerging new technologies that support through wireline broadband such as ADSL, HDSL, etc. as well as wireless broadband such as 3G, WiMAX, LTE, WIFI were developed. Wireless broadband becomes a promising technology because it has the advantage of easy to implement and requires less cost for installation and penetration than wireline. The example of wireless broadband that implemented in household right now is WIFI. However, wireless also has a disadvantage because it is transmitted through the air then should susceptible to noise and fading.

One of the technique used in wireless broadband is Orthogonal Frequency Digital Multiplexing (OFDM). This technique is a development of the Digital Frequency Multiplexing. In Frequency Digital Multiplexing total signal frequency band is divided into N non overlapping frequency[1]. But in OFDM signals divided into several subcarriers at a lower speed and send it with overlapping subcarriers. This signal combined by providing the requisite orthogonal. Orthogonal indicates that there is precise mathematical relationship between the frequencies of the system[1]. When the subcarrier truly orthogonal there is no interferences between subcarrier

An OFDM can be written as :

$$\mathrm{s(t)} = \sum_{i=-\frac{N_s}{2}}^{\frac{N_s}{2}-1} \mathrm{d_i} + \frac{\mathrm{N_s}}{2}\exp\left(j2\pi\frac{i}{T}(t-t_s)\right), \mathrm{t_s} \le \mathrm{t} \le \mathrm{t_s} + \mathrm{T}$$

$$s(t) = 0, t < t_s,\ \ t > t_s + T \qquad (1)$$

However, as is everything in the universe nothing is perfect. There would be a weakness of OFDM. One of the OFDM weakness is the high Peak Average Power

Ratio (PAPR). PAPR is the ratio of signal power amplitude value with an average amplitude value. More subcarrier in the signal would make the PAPR higher. High PAPR produced when N signal of OFDM subcarriers added in the same phase[2]. The peak of the power could be N times larger than average power. In mathematical can be written as

$$\text{PAPR(dB)} = 10\ log_{10} \frac{\max|X[n]|^2}{E[|X[n]|^2]} \quad (2)$$

When PAPR is high then the signal needs an amplifier with a wide dynamic range to avoid non-linear distortion. A key component is the high power amplifier (HPA)[3]. Wide dynamic range HPA owing to cost, design and most power efficiency consideration[3]. High PAPR also decreases the efficiency of the amplifier and increases the complexity of the Analog to Digital Converter (ADC) and Digital to Analog Converter (DAC). Therefore it is necessary to develop techniques to reduce the high PAPR values.

In OFDM there is a state in which no energy is transmitted or called null. This technique uses one or more null carriers with the intention that the data considered an subcarrier[4]. With the change in the input where data subcarrier that have switched to the null subcarrier that no energy transmitted will make the differences in IFFT input. Differences IFFT input will have different results on the output IFFT and PAPR itself.

method switching null subcarrier data subcarrier which proposed by Wong KT, B Wang and J Chen[4].This method use null signal on IFFT which use as a guard band to switch with data subcarriers. For further if there is an OFDM signal with L IFFT which mean has L subcarriers$\{fe\ , l = 1, \dots, L\}$ with ascending frequencies S = $\{l = 1, \dots, L\}$. And N is Null subcarriers with N = $\{g_n\ , n = 1, \dots, N\}$ and ascending frequencies $\{f_{gn}\ , n = 1, \dots, N\}$. D is data subcarriers $\{h_d\ , d = 1, \dots, L - N\}$ and of course uses frequencies indexing $\{f_{hd}\ , d = 1, \dots, L - N\}$.\

Without modifying L, D and N in OFDM the method switches S number of null subcarriers with data subcarrier. This switches have one condition that $f_{hs} < f_{hs+1}$and $f_{gs} < f_{gs+1}$. With this method the signal will have $\binom{S}{L-n} = \left(\frac{(L-N)!}{S!(L-N-S)!}\right)$ number different of switching possibilities. As an example if we have IEEE 802.11 OFDM signal with 64 subscarriers. Its consist of 54 data subscarriers and 12 null subcarriers[5]. With one switching null subcarriers data subcarriers there is exist $\left(\frac{(L-N)!}{S!(L-N-S)!}\right) = \left(\frac{(64-12)!}{1!(64-12-1)!}\right) =$ 52 possibilities of signal. And for two switches there is $\left(\frac{52!}{2!50!}\right)$= 1326 signal possibilities.

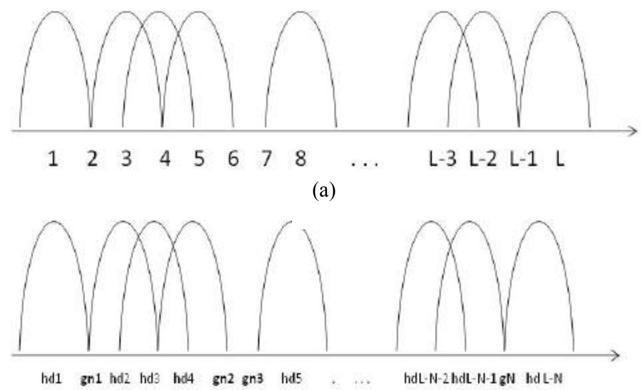


Fig. 1(a) OFDM L subcarriers
(b) OFDM L subcarriers with N null subcarriers and L-N subcarriers

With different possibilities the signal has different input of IFFT which will give the different output and different PAPR possibilities. To reduce the PAPR of the signal the method should try all of the possibilities and choose the minimum PAPR. The advantages of the method are there is no modifying on the signal which means the method not increasing Bit Error Rate (BER), and no side information is needed for this method. By the condition $f_{hs} < f_{hs+1}$ and $f_{gs} < f_{gs+1}$ the receiver will know which null subcarriers were switched and the system will un-switch before detector. Without side information means that the method is unlike SLM method which usesome subcarriers to send side information and reduce the data rate. Another advantage of the method could use simultaneously with another PAPR reduction scheme such clipping, SLM, Partial Transmit Sequence (PTS).

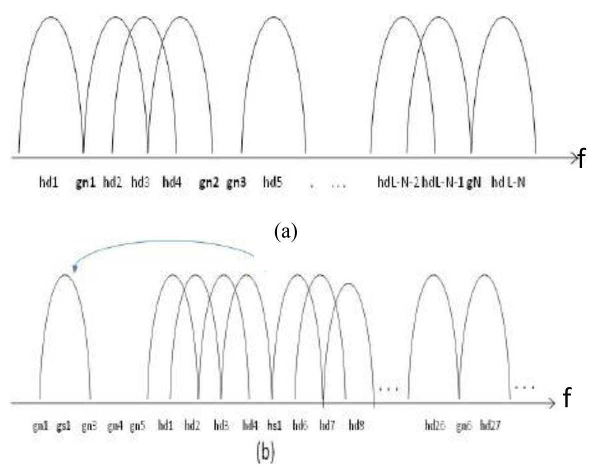


Fig. 2 (a) Data and null subcarriers at OFDM 64 subcarriers
(b) 1st switch gs1 with hs1

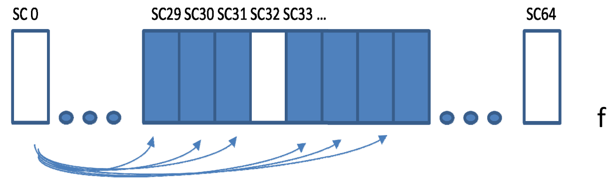
II PROPOSED METHOD : PAPR REDUCTION USING SWITCHING DATA SUBCARRIERS AND NULL SUBCARRIERS

A. BASIC CONCEPT

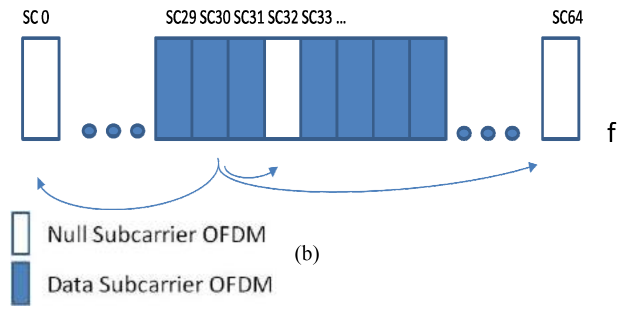
By using the same method with a switching null subcarrier data subcarrier can be used a method for find the lowest PAPR value by combination of input using switching data subcarriers with null subcarriers. Of course this proposed method will be different with previous method . When the method of switching null subcarrier data subcarrier, the subcarrier for switching is the null subcarrier but for the proposed method the data subcarrier is the one to be switched (fig. 3). The numbers of possibilities available in this proposed method would be less because generally the number of null subcarrier in an OFDM signal is smaller than the number of data subcarriers.

For example, in IEEE 802.11 signal which has 64 subcarrier the number of data subcarrier is 52 and 12 for null subcarrier.subcarrier[5]. In order to get the lowest PAPR value with switching null subcarrier data subcarrier method,12 null subcarrier should be exchange with 52 of data subcarrier fig.3(a). While in the proposed method, the 52 data subcarrier will be exchanged with 12 null subcarriers fig 3(b). Hence for 1 subcarrier switching the combination of proposed methods are $\left(\frac{(L-N)!}{S!(L-N-S)!}\right) = \left(\frac{(64-52)!}{1!(64-52-1)!}\right) = 12$ possibilities of signal. And for two switches there is $\left(\frac{12!}{2!10!}\right)=$ 66 signal possibilities.This means that the proposed method has less combination then the previous method.

Because the subcarrier already switched the output of IFFT will change too, hence for 1 switching on 802.11 OFDM signal there are 12 combinations of signals with different PAPR value. Obviously in the 12 combination will have both higher and lower PAPR compared with original signal. This combination will be chosen the lowest PAPR value in order to get the right signal to be sent. To avoid the side information needed the propose method use the same condition with switching null subcarrier data subcarrier method.



(a)



(b)

Fig. 3(a) Switching null subcarriers data subcarriers
(b) Switching data subcarriers null subcarriers

B. COMPUTIONAL COMPLEXITY REDUCTION

One of the goal of proposed method is to reduced complexity of the system. It could be proof by system calculation. Both of the system base on scrambling technique. Which mean the method search all possibility null and data switching position in order to find the lowest PAPR. Then choose the lowest PAPR and send the data series to the HPA. The complexity both system shown at table 1.

From table 1 show that the for 64 IFFT size proposed method gives 12 possibilities in order to find the smallest PAPR with 1 switching subcarrier. That was lower computional complexity compare with switching null subcarrier data subcarrier which have 52 possibilities. When 2 subcarrier were switched proposed method only need 66 possibilities compare with 1326 possibilities when using previous method.

TABLE 1 computional complexity proposed method and previous method

Method	Switching possibilities	64 IFFT size			
		Null SC	Data SC	Number of possibilities	
				1 SC switching	2 SC switching
Switching null subcarrier data subcarrier	$\left(\frac{(L-N)!}{S!\,(L-N-S)!}\right)$	12	52	52	1326
Proposed Method	$\left(\frac{(L-D)!}{S!\,(L-D-S)!}\right)$	12	52	12	66

III. SIMULATION AND RESULT

In order to test the method, in this thesis will simulating the proposed method in the computer. Simulation only in OFDM transmitter in order to find the less PAPR of the signal by computing every possibilities signal. Pseudo code of the simulation shown below

Algorithm proposed method
begin
Input : jumlah_bit, jumlah IFFT= {64, 128, 256},jumlah_switch={1}
Output : nilai_PAPR_final, out_IFFT_final
Jumlah_frame=jumlah_bit/jumlah_IFFT
For i=1 to jumlah_frame do
For j=1 to jumlah_switch do
For k=1 to jumlah_null do
Switch hdj with gp(jumlah_null)
Apply IFFT and calculate PAPR
Save nilai_PAPR(j,k), out_IFFT(j,k)
End for
End for
Choose minimum PAPR, nilai_PAPR_final(i), out_IFFT_final(i)
End for
End

Tabel 2 Simulation Parameters

IFFT size	64
Number of Data Subcarrier	52
Number of switched subcarrier	1
Modulation	BPSK
Total number of ODFM symbol	4.000

C. SIMULATION RESULT

The simulation results are described in the form of CCDF . To see how PAPR with 1 switching produced PAPR and compare the PAPR with PAPR of original signal and PAPR with previous methods can be seen in fig. 4,5 and 6.

At fig. 4 shows that for 64 IFFT size the proposed method gives lower PAPR than the original signal but still higher than PAPR with previous method. For PAPR more than 6 dB original signal has probability 0.5 (50 percent) and previous method gives probability 0.3 (30 percent) and the proposed method gives 0.35 or 35 percent.

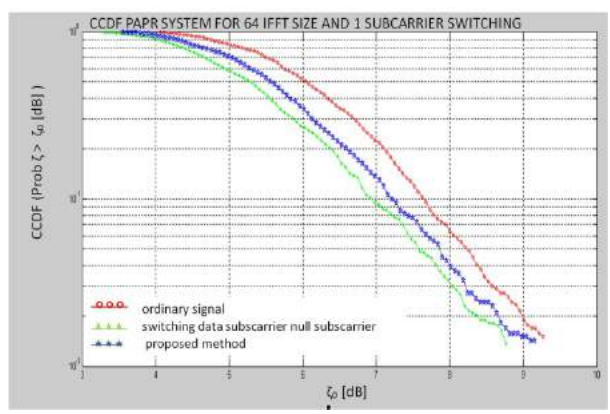


Fig. 4 CCDF of PAPR simulation with 64 IF IFFT and 1 subcarrier switching

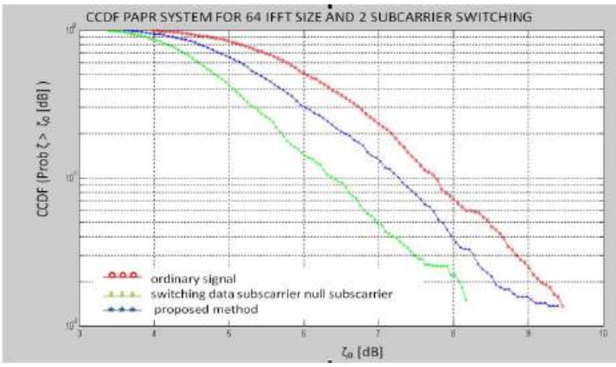


Fig. 5 CCDF of PAPR simulation with 64 IF IFFT and 2 subcarrier switching

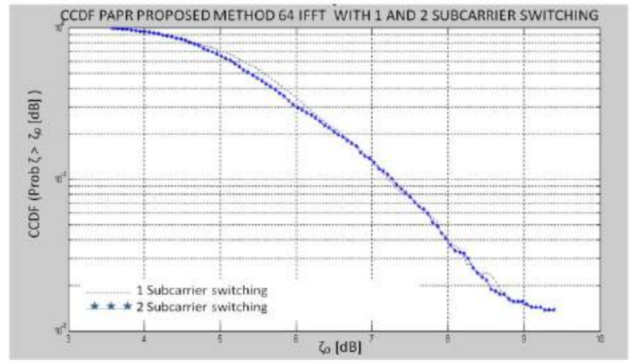


Fig. 6 Reducing PAPR using proposed method with 1 and 2 subcarriers switching

In the second chart which proposed method simulating with 2 subcarrier switching the PAPR still lower than the original signal but still higher compare to PAPR using previous method (switching null subcarrier data subcarrier). It can be seen probability PAPR more than 6 dB with 2 switching is 0.15 (15 percent) using previous method and 0.3 (30 percent) using proposed method fig. 5. This probability is decrease compare to 1 subcarrier switching especially compare to previous method. Also shown on both graphs that 2 subcarrier switching gives lower PAPR than 1 subcarrier switching. This simulation result according to hypothesis that PAPR of the proposed method will be lower than original signal but still higher than previous method. That because the possibility of the signal in proposed method less than possibility signal using previous method. From table 1 can be seen that for 1 subcarrier switching proposed method has 12 possibilities signal compare to 52 possibilities using previous method. and for 2 subcarriers switching proposed method only has 66 possibilities compare to 1326 using previous method.

Reducing PAPR using proposed method with 1 subcarrier switching and 2 subcarriers switching shows at figure 6. The graph shows that differences of PAPR with 1 witching and 2 switching is very little between them, for 64 IFFT the biggest differences only in 5.5 db. When using 1 subcarrier switching probability PAPR more than 5.5 db is 0.5 then using 2 subcarrier switching the possibilities decrease which only has 0.45. This result does not like previous method that gives significant differences.

IV CONCLUSION

A new method to reducing PAPR based on previous method (switching data subcarriers null subcarriers) is proposed. From calculation show that number of possibilities signal lower than previous method. that mean computational complexity of the system is reduced. Hence from simulation show that proposed method could reduce PAPR compare to original signal with consequence PAPR proposed method still higher compare to previous method. another consequence is additional subcarrier switching can not give big impact on PAPR reducing.

[1] Prasad, Ramjee, OFDM for Wireless Communication System. s.l. : Artech House, 2004.

[2] Richard Van Nee, Ramjee Prasad, OFDM for Wireless Communication. s.l. : Artech House, 2000.

[3] Litsyn, Simon, Peak Power Control in Multicarriers Communications. s.l. : Cambridge University Press, 2007.

[4] Wong KT, B Wang, JC Chen, "PAPR Reduction by Switching Null Subcarriers and Data Subcarriers", s.l. : IEEE, 2011, Vol. 27.

[5] **IEEE.** Information Technology. Telecommunications and Information Exchange Between Systems-Local and Metropolitan Area Networks Specific Requirements-part 11 : Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Spesification : High Speed Physical. *IEEE Std 802.11a.* 1999.

Analysis of Interference from Wireless Traffic Light Controller upon Remote Keyless Entry for Vehicles

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Abstract-This paper aims to analyze the interference between wireless traffic light controller and remote keyless entry for vehicles. Both systems are located closely to each other at the frequency 433-434 MHz. During the analysis, broadcast variant time delay on wireless traffic light versus distances between and RKE determined to ensure their interference. The interference impact is evaluated in a scenario where the wireless traffic light and RKE for vehicles operate on frequency 434 MHz and 433.9 MHz respectively. Three time delays 0 s, 0.2 s and 0.5 s were considered during the simulations. The results show that time delay 0 s and 0.2 s cause a very high probability of interference. Whereas time delay setting of 0.5 s provide probability of interference about 2 % at distance 5 m.

Keyword: Adjacent Chanel, Interference, probability of interference, Remote Key Less Entry, Time Delay.

I. INTRODUCTION

In recent time, the automobile industry demonstrates a significant progress. Several important functions are enhanced by means of electronic supports. Remote Keyless Entry (RKE) based on the use of radio frequency signals is an example of such advances. The performance of remote keyless entry systems under the influence of conducted Radio Frequency (RF) signals was investigated [1]. The RF signal is induced through a Bulk Current Injection (BCI) test in the frequency range of 1 MHz – 400 MHz into an RKE receiver with the 314.9 MHz frequency and -110 dBm of sensitivity requirements.

In [2], the robustness of two RKE systems to electromagnetic interference is investigated. Measurements are performed in an anechoic chamber to estimate the robustness of the RF receivers. Results show that the selectivity of both systems are poor and they are easily jammed. Field strengths as low as 0.1 V/m can already jam the wireless link over a wide frequency bandwidth of 4 MHz.

The development of wireless technology is very fast and its applications in many systems are ubiquitous. Wireless technology is also used in traffic light controller for replacement the previous used underground cable networks. Wireless traffic light controller is an electronic module used for easy and effective control a road junction with a wireless remote control [3]. The system can be driven in one of two modes, manual or automatic. In the manual mode, an operator can change manually the control of traffic light in the master module, and it sends signals to slave controller modules for changing the traffic lights. In automatic mode, the traffic light controller board will change the light sequence according to the preset patterns and time delay. An operator can certainly change the pattern at any time using the remote. The propose system help the policeman to control the junction by himself and be able to change conditions of the traffic flow dynamically.

In this work we observe a RKE system with 433.9 MHz which might be interfered by a wireless traffic light controller working at a much closed frequency position of 434 MHz. We observe the potentially existing interference from the wireless traffic light controller to the RKE system as a victim, by varying the time delay setting and distance from the wireless traffic light controller to the RKE system.

This work is structured as follows. Section II briefly describes the interference, Wireless traffic light systems and Remote keyless entry for vehicles. Section III shows the simulation scenario adopted. Section IV presents the mathematical modeling for the observation of interference here. The results and analysis are presented in Section V, and finally, the conclusion in Section VI.

II. INTERFERENCE PHENOMENON FROM WIRELESS TRAFFIC LIGHT CONTROLLER UPON REMOTE KEYLESS ENTRY

Interference is defined as the effect of unwanted energy due to one or a combination of emissions, radiation, or inductions upon reception in a radio communication system, manifested by any performance degradation, misinterpretation, or loss of information which could be extracted in the absence of such unwanted energy [4]. The origin of the interference may be at any of three locations [5]: A transmitter may generate out-of-band signal components (such as harmonics) that fall in the band of the desired signal. Meanwhile, the front end of a receiver may overload or produce spurious responses. And in the channel, nonlinearity in the transmission medium may cause undesired signal components in the band of the desired signal.

In respect to spectrum positions of the interfering and interfered signals, interference is distinguished by co-channel interference (CCI) and adjacent channel interference (ACI). CCI is defined as interference resulting from two or more simultaneous transmissions on the same channel [4]. Whereas, ACI is the interference from a signal upon a signal in an adjacent channel. ACI may be caused by inadequate filtering, such as incomplete filtering of unwanted modulation products in frequency modulation (FM) systems, improper tuning, or poor frequency control, in either the reference channel or the interfering channel, or both.

Wireless traffic light controller is an electronic module used for easy and effective control a road junction with a wireless remote control [3]. The system can be driven in one of two modes, manual or automatic. In the manual mode, an operator can change manually the control of traffic light in the master module, and it sends signals to slave controller modules for changing the traffic lights. In automatic mode, the traffic light controller board will change the light sequence according to the preset patterns and time delay. An operator can certainly change the pattern at any time using the remote. The propose system help the policeman to control the junction by himself and be able to change conditions of the traffic flow dynamically.

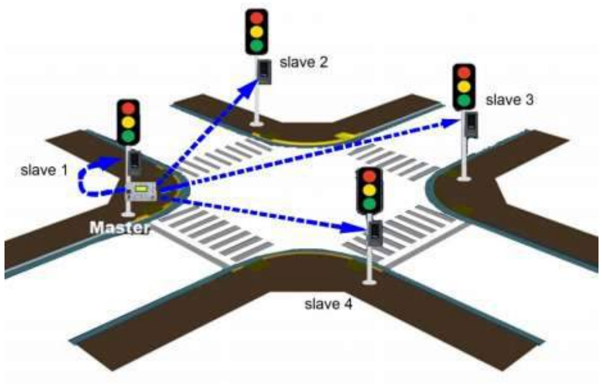


Fig. 1. Wireless traffic light controller.

A wireless traffic light controller mounted at a street crossing is illustrated in Fig.1. The module consists of microcontroller-based master and slave modules. The modules are connected to each other wirelessly. In [3], a 2.4 GHz wireless system is used. In [6] an RF transceiver ChipCon CC1000 916 MHz is applied. In this work, the master and slaves modules are connected by a wireless system with carrier frequency of 434 MHz.

A remote control device for vehicles or remote keyless entry (RKE) is an electronic device became a popular feature for vehicles. The system consists of a handheld subsystem (“the key”) and the electronic structure embedded in the vehicle, as depicted in Fig.2. The RKE module includes the following features locking, unlocking, remote start, window closures, and activation of an alarm. Fig.2 describes how the remote keyless entry works. In “the key”, a sequence counter together with an encryption key is combined to get a secure encrypted data. The data is sent by an RF transmitter to the on-the-vehicle mounted receiver. The RF transmitter and receiver used in the RKE works on UHF (ultra-high frequency) range, 315 MHz or 433 MHz [9].

III. OBSERVATION SCENARIO

In this work, an assessment on the effect of interference from wireless traffic light controller upon a remote keyless entry is carried out. The observation is performed in an open area, i.e. the wireless traffic light controller module, the remote keyless entry and the measurement instrument (a spectrum analyzer) are located in an outdoor environment, as shown in Fig.3.

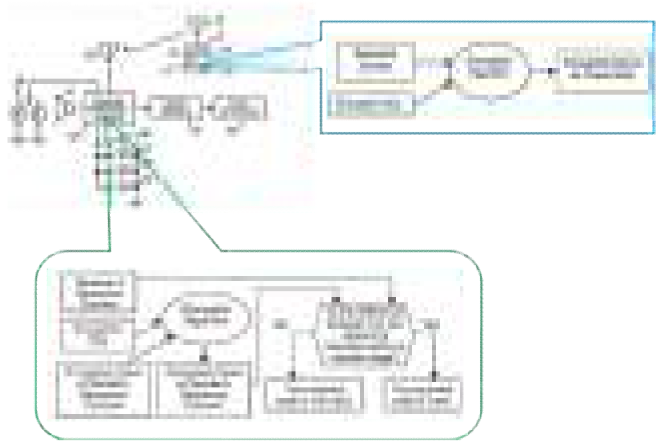


Fig. 2. Remote keyless entry for vehicles.

Regarding to the scenario in Fig.3, the received signal powers are measured by varying the time delay setting (0s, 0.2 s and 0.5 s) at the microcontroller in the master module of the wireless traffic light controller. By a certain time delay, for example by time delay 0.2 s, the effects of interference is studied by varying the distance from the wireless traffic light controller to the spectrum analyzer. The received signal is designated as the interference signal (iRss).

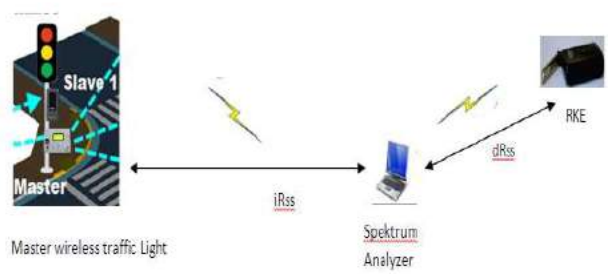


Fig. 3. Observation Scenarios.

At the same time, the signal sent by the remote keyless entry from a fixed distance of 1.0 meter is measured. The received signal is designated as the desired signal (dRss). This data will be useful for the analysis of probability of wireless interference obtain traffic light on the remote key less entry. Additional information in this research are, the remote keyless entry used is remote keyless entry of Xenia car, which works at frequency about 433.9 MHz, and the following table should give parameters of the master module of the wireless traffic light controller.

TABLE I
PARAMETER OF THE MODULE WIRELESS TRAFFIC LIGHT CONTROLLER

Carrier Frequency	Interface	Modulation	RF power	Impedance
434 MHz	RS 232 / RS 485/ TTL	GFSK	≤ 10 dBm	50 Ω (antenna port)

IV. MATHEMATICAL MODELING

This section provides the mathematical description to quantify the interference level occurring in cases observed in this research. The quantity used here is the probability of interference from the wireless traffic light controller upon the remote keyless entry (RKE) for vehicles. The probability of interference used here is as defined in [7]:

$$P_I = 1 - P_{NI} \tag{1}$$

where P_I is the probability that the wireless traffic light controller interferes the RKE, P_{NI} is the probability that wireless traffic light controller does not interfere the RKE, with

$$P_{NI} = \left(\frac{dRss}{iRss} > \frac{C}{I} \middle| dRss > sens_{RX} \right) \tag{2}$$

dRss and *iRss* are as defined in previous section, the desired received signal strength and the interference received signal strength, respectively, (*C/I*) is the signal-to-interference ratio and $sens_{RX}$ is the receiver sensitivity. Eq. (2) is a conditional probability that the ratio of the desired received signal strength to the interference received signal strength bigger is than the signal-to-interference ratio given that the desired received signal strength bigger is than the receiver sensitivity.

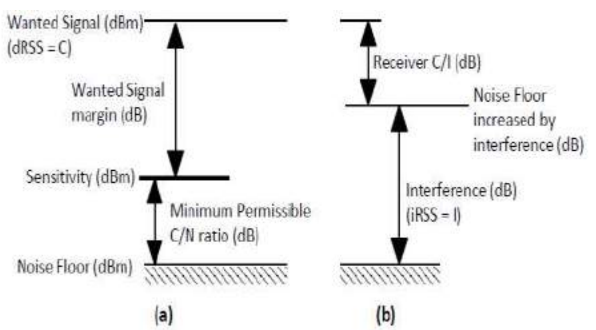


Fig. 4. The signal levels used to determine whether or not interference is occurring [8].

Fig.4 (a) represents the situation when there is no interference and the victim is receiving the desired signal with some margin. In this case the victim's Signal level is given by the sum of the Sensitivity and wanted signal margin l. Fig.4 (b) illustrates what happens when interference occurs. The interference adds to the noise floor. The difference between the wanted signal strength and the interference signal, measured in dB, defines the Signal to Interference ratio. This ratio must be greater than the required C/I threshold if interference is to be avoided. The spectrum analyzer checks for this condition and records whether or not interference is occurring. This is figure illustrated further in Fig. 5.

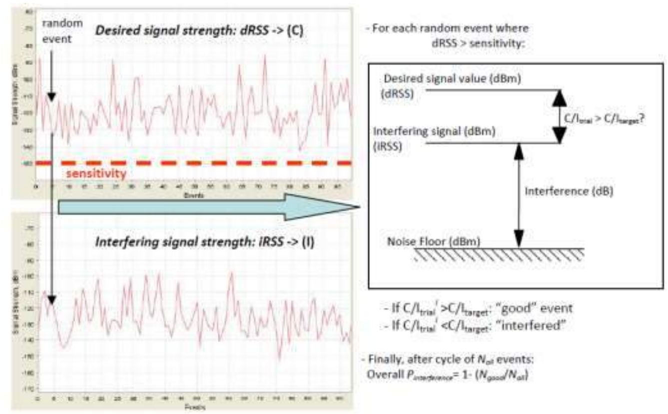


Fig. 5. Illustrative summary of the interference criteria computation [8].

In this research, for each parameter setting, i.e. certain time delay setting and distance from wireless traffic light controller to the spectrum analyzer, we conducted 100 times measurements each event and counted the number of two situations which could happen. In this way, the conditional probability should be approximated numerically by

$$P_I = 1 - \frac{N_{good}}{N_{all}} \tag{3}$$

where

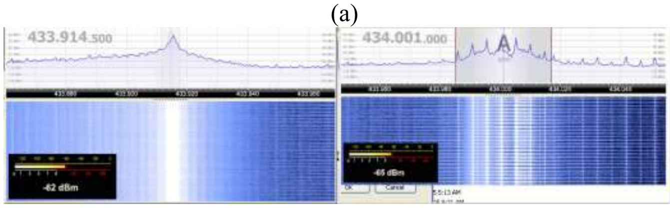
- If $C/I_{\text{trial}} > C/I_{\text{target}}$ state : " good " event
- If $C/I_{\text{trial}} < C/I_{\text{target}}$ state : " interfered "

The capacity of the wireless system is dependent on the value of *C/I*. The smaller the value of interference or noise that happens then the performance of the remote keyless entry for vehicles becomes better. In this research the value of C/I_{target} is defined from the experiments, in which ratio the RKE can still work under interference by the wireless traffic light controller. The Value C/I_{target} above 0 dB does not interferer RKE

V. RESULTS

Due to the almost overlapping spectrum interval between the wireless traffic light controller and the remote keyless entry (RKE), we belief the continuous operation of the wireless traffic light controller, i.e. time delay setting 0 s, can interfere the RKE. However, too large value of time delay can degrade the performance of the traffic light controller, and hence can affect the traffic flow significantly. In this work, we would like to study, how the traffic light controller can interfere the RKE by a certain time delay setting at a certain distance.

(a)



(b)

Fig. 6. (a) captured measurement for time delay 0.2 s, (b) captured measurement for time delay 0.5 s.

Fig. 6 shows the received signals from the RKE at the distance 1.0 m, for case (a) from the wireless traffic light controller at 1 m by a time delay 0.2 s, and case (b) from the wireless traffic light controller at 10 m by a time delay 0.5 s. We see for case (a), the received signal from the RKE is about -65 dBm at the frequency 433.9005 MHz, and the received signal from the wireless traffic light controller about -60 dBm at the frequency 434.001 MHz, which yields the approximated signal to interference ratio of about C/I trial = -5 dB. This value is smaller than the C/I target, so that this case is categorized as interfered condition. In the case (b) we received the signal from RKE -62 dBm at 433.9145 MHz and from the wireless traffic light controller -65 dBm at 434.001 MHz, which leads to C/I trial = 3 dB. The second case belongs to the "good" event. In this measurement use variasi variant FM with WFM (Wide FM) maximum 48 KHz at spectrum analyzer to get maximum peak power signal WTLC and RKE .

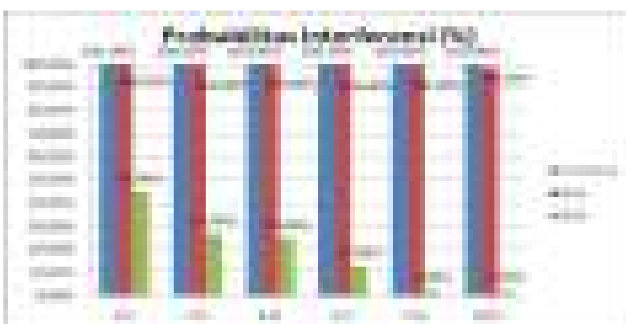


Fig. 7 Probability of Interference Wireless Traffic Light upon the RKE.

We repeat now such observations hundred times for different time delay settings 0 s (continuous case), 0.2 s and 0.5 s at the different distances from the wireless traffic light controller to the spectrum analyzer, whereas the distance from the RKE to the spectrum analyzer is kept constant by 1.0 m. By counting the 'good' event for each setting, we can approximate the probability of interference by eq. (3).

Fig. 7 summarizes the observation results. We have no chance to avoid interferences for the continuous case. Also for the time delay setting of 0.2 s, the interference is still too high for the RKE system, even for large distance at 10 m we still have very strong interference (99 %).

The setting of time delay to 0.5 s changes the condition significantly. At the same distance to the spectrum analyzer as the RKE system, i.e.1 m, the RKE system delivers more power to the spectrum analyzer than the wireless traffic light controller does, so that the probability of interference shows a value of 45 %. By moving the spectrum analyzer away from the controller, the probability of interference decreases monotonic, and shows the value about 2 % at the distance of 5 m. This observation results could be as a recommendation for choosing a parking location, especially for crowded places like in downtown.

VI. CONCLUSION

In this research we conducted an interference analysis from the wireless traffic light controller (WTLC) upon the remote keyless entry (RKE) system. We made observation by different settings of time delay (0 s, 0.2 s and 0.5 s) and at different distance (1 m to 10 m) between the wireless traffic light controller to the spectrum analyzer. By taking the C/I between the signal from the RKE and the interference from the WTLC to the spectrum analyzer, we counted the number of the interference cases. The observation gives the results, that the time delays 0 s and 0.2 s lead to very strong interferences. However the time delay 0.5 s causes smaller interferences. At the distance of 1 m the probability of interference is 45%, whereas for the distance 5 m, the probability of interference decreases to a very smaller value of 2 %.

REFERENCES

[1] Rostamzadeh, C., Pavatich, F., On Determination of Conducted RF Immunity Test Methodology for Automotive Remote Keyless Entry Receivers, Electromagnetic Compatibility, 2008. EMC 2008. IEEE International Symposium on, 18-22 August 2008, pp. 1-6.

[2] van de Beek, S., Vogt-Ardatjew, R., and Leferink, F., Robustness of Remote Keyless Entry Systems to Intentional Electromagnetic Interference, Proc. of the 2014 International Symposium on Electromagnetic Compatibility (EMC Europe 2014), Gothenburg, Sweden, September 1-4, 2014.

[3] Thatsanavipas, K, Pongabunchoke, N, Mitatha, S and Vongchumyen, C., Wireless Traffic Light Controller, Procedia Engineering 8 (2011) pp. 190-194.

[4] Federal Standard FED-STD-1037C – Glossary of Telecommunication Terms, 1996, http://www.its.bldrdoc.gov/fs-1037/fs-1037c.htm (accessed August 2015).

[5] Couch, Leon W, II, Digital and Analog Communication Systems, 8th ed., Pearson, New Jersey, 2013.

[6] Tubaishat, M, Qi Qi, Shang, Y., Shi H., Wireless Sensor-Based Traffic Light Control, Consumer Communications and Networking Conference (CCNC), Las Vegas, NV, 2008, pp. 702-706.

[7] Więcek, D., & Wypiór, D,.New SEAMCAT Propagation Models: Irregular Terrain Model and ITU-R P. 1546-4. Journal of Telecommunication and Information Technology, No. 3, 2011, pp. 131-140.

[8] European Communications Office, SEAMCAT Handbook. 2010, www.cept.org (accessed August 2015).

[9] Francillon, A., Danev, B., Capkun, S.: Relay attacks on passive keyless entry and start systems in modern cars. Cryptology ePrint Archive, Report 2010/332, 2010.

Comparative Analysis of LLQ Traffic Scheduler to FIFO and CBWFQ on IP Phone-Based Applications (VoIP) Using Opnet (Riverbed)

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Abstract- Nowadays, world is on going to "Convergence Era", where data, voice, and video overcome to pass through same network, like IP network. Unfortunately, the fast growth of application technology has not been supported yet by default IP network in the case of quality of service. As we know, that voice over IP services have sensitivity to delay, jitter, latency, and requirement of different bandwidth which is also requiring some service guarantee. How comparing queue mechanism of VoIP attributed to QoS? In this case, need to give guarantee of QoS relied on operation of data packages to be delivered along with its scheduling. Scheduling represent queue mechanism from an applying of network QoS's, that is an arrangement process of an exit and entry of assumed data packages queuing exceed CIR (Committed Information Rate) specified. Regulation of this data exit and entry can be formed with specified classification. This research compare queue method of First In First Out (FIFO) with two algorithm type of WFQ scheduling which at most used, that is Class Based Weighted Fair Queuing (CBWFQ) and Low Latency Queuing (LLQ). Pursuant to Delay, Jitter, and MOS (Mean Opinion Score) obtained, LLQ give performance of network in a better way.
Keyword: VoIP, QoS, CIR, LLQ, Riverbed.

I. INTRODUCTION

In supporting growth of a sustainable development, a development of friendly environment technology is urgently needed. Technological environmental friendliness earn is also marked with existence of a better technology development giving effectiveness. Among other is development of technology giving the increases of our network performance.

Human internet activity is being progressively in time. Taken from survey, obtained at just 2011, the activity in most business sector is often by to send and accept e-mail equal to 97.69%, while the lowest percentage activity is hotel promotion equal to 0.03%, teleconference through VoIP its self equal to 13.54% [16]. This percentage is indicate that the requirement of VoIP technology need excellence of good network.

With an early hypothesizing that queue mechanism of LLQ have better performance [16], [13], this research conducted to know as for how comparing VoIP queue mechanism to it's QoS? In this case, by comparing traffic scheduler of VoIP with algorithm of Low Latency Queuing (LLQ) with First In First Out (FIFO) and Class Based Weight Fair Queuing (CBWFQ).

As for limitation, this research concern on several things, that network only focused at two router and also in application of VoIP, then this research use simulation software of Riverbed Academic Edition in it's perception.

II. THEORY

Information for the classification process in the IP header field or in other words the process of classification on layer 3 of the OSI standard, there are two kinds, namely IP Precedence and Differential Service Code Point (DSCP). This classification information specified in the first three or six bits of the field Type of Service (ToS) in the IP packet header.

Based on the timeline-based categories, there are 3 Network QoS parameters, namely Delay, Jitter and Response Time [15]. In here, we are only use the Delay and Jitter parameters because these parameters have the greatest influence on the Network. This research are also used mean opinion score (MOS) to analized the performance of network.

A. Delay

Delay is the time it takes a bit of data to pass through a network from a node to another node. There are three pieces of delay, namely

- Transmission Delay
- Propagation Delay
- Delay processing

Performance is considered good if the value of the resulting delay getting smaller. To compute a delay can be formulated as follows:

$$\text{Delay} = \text{Time of Arrival} - \text{Post time} \quad (1)$$

In OPNET tutorial outlined on its website[18], the delay represented by End-to-end delay, ie a total delay of voice packets (analog-to-analog) mathematically written:

$$\text{Packet end-to-end network delay} = \text{delay}_{\text{encoding}} + \text{delay}_{\text{decoding}} + \text{compression delay} + \text{decompression delay} + \text{delay} + \text{delay dejitter\_buffer} \quad (2)$$

B. Jitter

Jitter is a variation of delay caused by long queues in a data processing time. There are two types of jitter, which is one way jitter and inter-arrival jitter. Mathematically written as follows.

$$average\_delay = \frac{total\_time}{packet\_arrival} \qquad (3)$$

The ITU-T standard value of jitter is still tolerable is 30 ms.

C. Mean Opinion Score (MOS)

Mean opinion score ITU Recommendation P.800 is used to measure the performance of a multimedia communications over a network based on a view of the end user.

In the multimedia communication network (such as audio, video, or voice telephony), especially when the codec used to compress the bandwidth required for the communication, it takes MOS to measure the extent to which the quality of the communication is based on the perspective of the end user. End users will provide an assessment of the range of numbers 1 -5 wherein, figure 1 means very poor quality and number 5 is very good quality.

D. Traffic Scheduller

1) First In First Out (FIFO) traffic scheduller

FIFO queuing technique refers to the FCFS (First Come First Serve), the data packets are processed first come first. Data packets that come out first in the input into the FIFO queue, and then issued in accordance with the order of arrival. FIFO queuing technique is suitable for medium-sized networks with bandwidth of 64kbps but spend enough processor and memory resources.

Fig. 1. FIFO queue [6]

2) Weighted Fair Queuing (WFQ) traffic scheduller

WFQ consists of two types of algorithms, ie CBWFQ and LLQ as described later. WFQ mathematically expressed by SN (Sequence Number) in the equation:

$$SN = Previous\_SN + (Weight * New\_Packet\_Length) \qquad (4)$$

Then,

$$Weight = 32384/(IP\ Precedence + 1) \qquad (5)$$

CBWFQ in initial mode on an interface separates traffic into flows, determining the transmission speed of each queue, and then weighting the priority of each queue. From the perspective of WFQ, there are two categories of data flows: a session high bandwidth and low bandwidth session. Low-bandwidth traffic effectively given priority over high-bandwidth traffic, and high bandwidth traffic transmission service proportionally dividing joint according to a predetermined weighting.

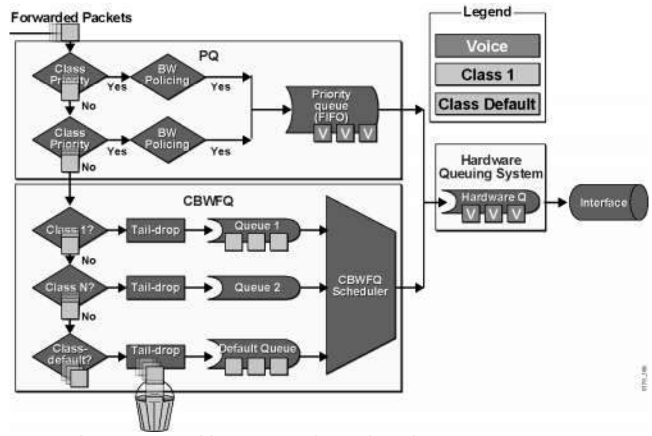

Fig. 2. LLQ Architecture (Implementing Cisco QoS, 2006)

III. RESEARCH METHODS

The research method used is the application and measurement of simulation network using Opnet/riverbed software. As for the mind set of simulation/research is can be depicted at picture below.

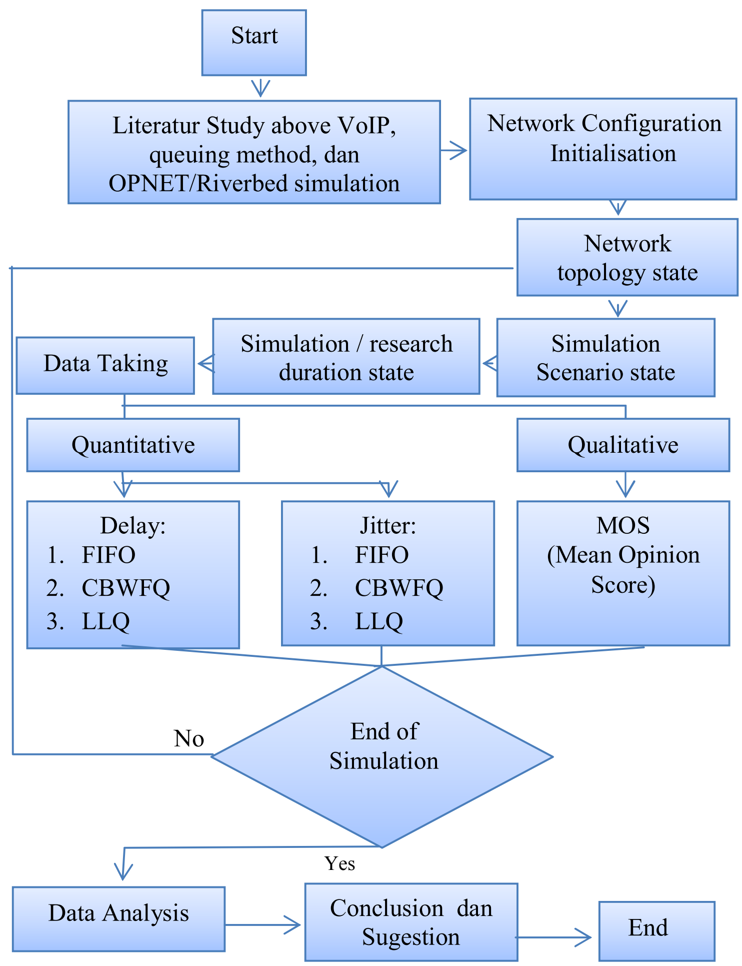

Fig. 3. Research Design

Scheme of simulation used to design network system to be used at this research, as well as designing topology for the VoIP nodes. Making of simulation, that is to make simulation in OPNET/Riverbed Academic Edition based to devices which have been made before. The measurement done with measuring performance from three queue method of FIFO, CBWFQ and LLQ at parameter of Packet End to End Delay, Jitter, and Mean Opinion Score (MOS).

Data retrieval is steps to be taken after the results of the measurements obtained. The data obtained from simulations using OPNET/Riverbed Academic Edition. From the results of the processing of this data will be created statistics from the data obtained. Then, do the analysis and drawing conclusions on all results of data retrieval.

As for application settings and configurations can be done as option following, pursuant to topology settings and of the following atribute.

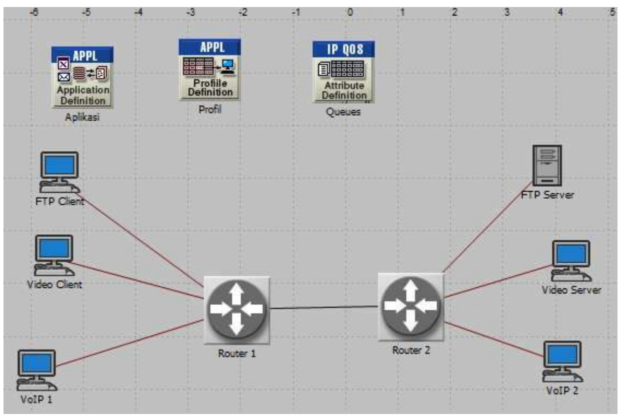

Fig. 4. Network Topology [9]

Network topology above consist of: 4 workstation 10BaseT, link 10BaseT, 2 Router, FTP Server, Video Server, and PPP DS1 link with capacities of link 1,544 Mbps with average packet size 200 byte. That workstations is:

- FTP Client is conduct transmission of FTP with FTP Server, by 50% command put and 50% command get, inter request time generated with constant distribution, and mean outcome is 10, file size 500 kbps. With Type of Service: Best Effort.
- Video client, by default consist of 1 user asking for video service of video server
- Voice over IP - VoIP 1, to get communications voice with VoIP 2.
- Voice over IP - VoIP 2, to get communications voice with VoIP 1.

In the network there are 3 applications: FTP Application (Parameter for the application of this FTP is: High Load with Type of Service (ToS) is Best Effort (0)); Video of Application (Parameter for the video application is: VCR Quality Video, with ToS Streaming Multimedia (4)); Voice Application (Parameter for the application of this FTP is: PCM Quality Speech, with ToS Interactive Voice (6)).

Profile configuraton in network above is: FTP Profile (Application: “Aplikasi FTP”, with Start Time Offset : constant (5), and Repeatability: Once at Start Time, Operation Mode: Simultaneous, Start Time: constant (100), duration: end of simulation); Video Profile (Application: “Aplikasi Video”, with Start Time Offset: constant (5), and Repeatability: Once at Start Time, Operation Mode: Simultaneous, Start Time : constant (100), duration: end of simulation); Voice/VoIP Profile (Application : “Aplikasi Voice”, with Start Time Offset: constant (5), and Repeatability: Once at Start Time., Operation Mode: Simultaneous, Start Time: constant (100), duration: end of simulation).

IV. Research Results

A. Voice Traffic Average FIFO, CBWFQ, and LLQ

Voice Traffic Sent to be obtained from FIFO, CBWFQ, and LLQ in the measurement have same values. This means that all of the method of queuing is observed, ie FIFO, CBWFQ and LLQ, experiencing the same conditions.

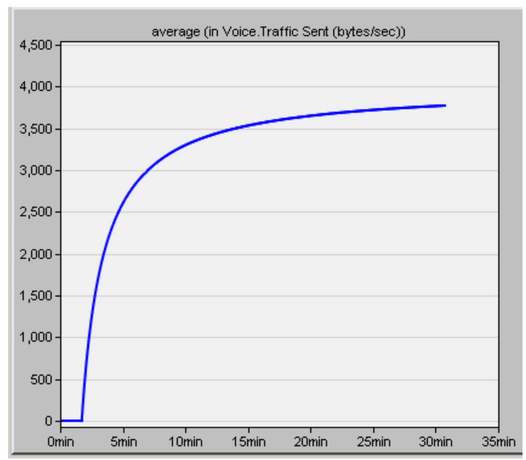

Fig. 5. Average in Voice Traffic Sent of FIFO

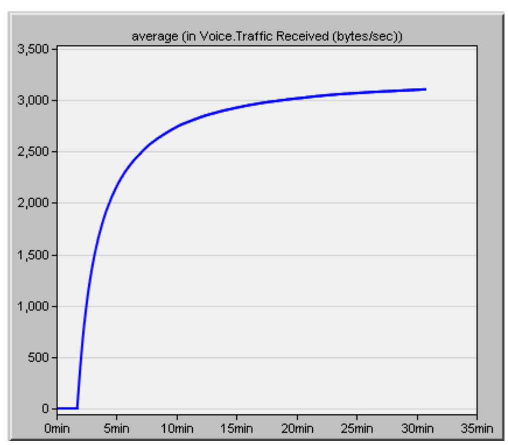

Fig. 6. Average in Voice Traffic Received of FIFO

Voice Traffic Received to be obtained from result is that maximum number reach 3,105kbps.

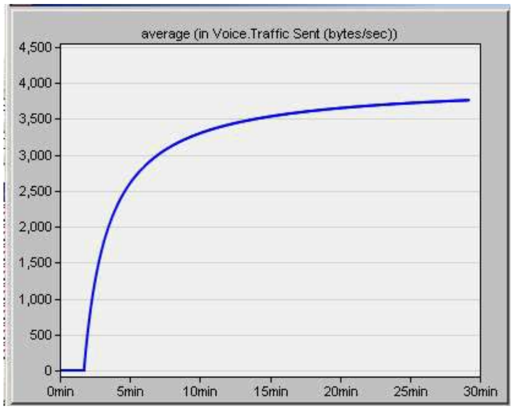

Fig. 7. Average in Voice Traffic Sent of CBWFQ

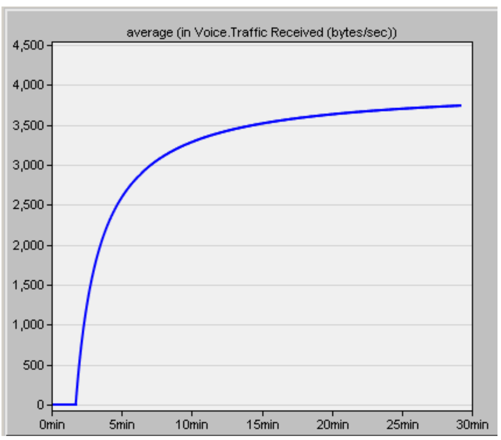

Fig. 8. Average in Voice Traffic Received of CBWFQ

Voice traffic received to be obtained from result that maximum number reach 3,741.4kbps.

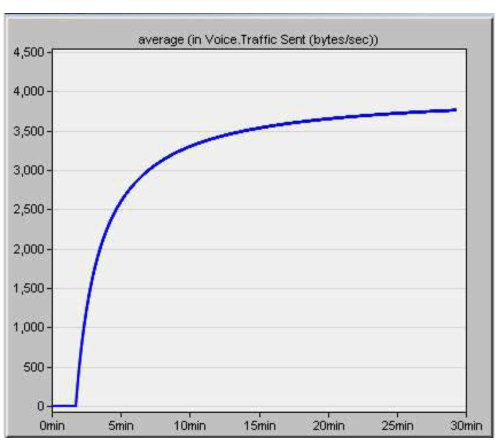

Fig. 9. Average in Voice Traffic Sent of LLQ

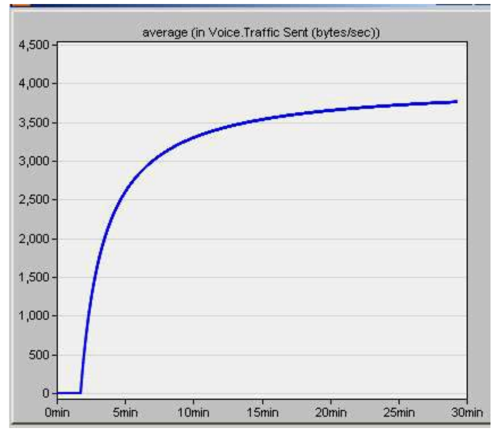

Fig. 10. Average in Voice Traffic Received of LLQ

Voice traffic received to be obtained from result is that maximum number reach 3,741.7kbps.

B. Voice Packet End-to-End Delay of FIFO, CBWFQ, and LLQ

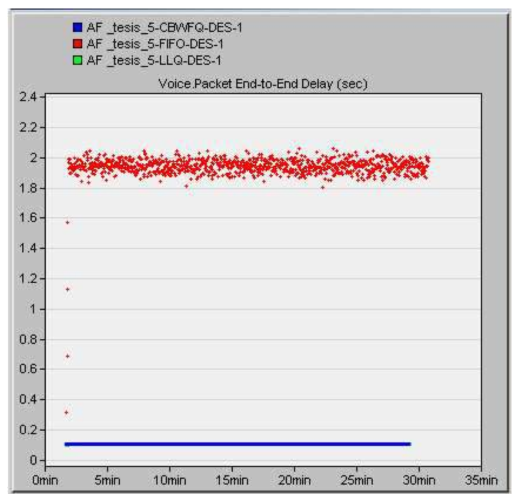

Fig. 11. Voice Packet End-to-End Delay of FIFO, CBWFQ and LLQ

At FIFO, delay happened until number 2s, while CBWFQ and LLQ delay at spanning 0,01s. LLQ data and CBWFQ relatively have same value, some this LLQ data below CBWFQ is equal to 105ms. But some other CBWFQ delay data doesn't. It's means that this LLQ can degrade delay compared to CBWFQ but relative unstable yet.

At some data, LLQ have smaller delay than CBWFQ, that is about 105ms. Both queue method, that is CBWFQ and LLQ, have value of delay below 150ms. Its mean can be well accepted by user. This matter refers to recommendation of ITU-T G.114 concerning delay at one way communication. The application network delay of VoIP, including best range 0-150 ms (acceptable for most application).

C. Jitter of FIFO, CBWFQ, and LLQ

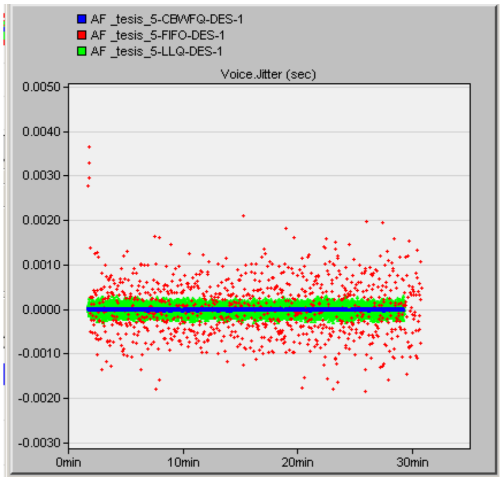

Fig. 12. Voice Packet End-to-End Delay of FIFO, CBWFQ and LLQ

From the above results, it can be concluded that the result of the jitter value obtained is within the limits of the ITU-T standard value of jitter is still tolerable is 30 ms. The graph also shows that for every scenario jiiter greatest value obtained in the FIFO scenario, this is due to the instability of the network delay when using the FIFO than CBWFQ and LLQ. So, Higher jiiter values to be got at FIFO scenario, this matter because of instability of delay when using FIFO compared to CBWFQ and LLQ.

D. Mean Opinion Score (MOS) of FIFO, CBWFQ, and LLQ

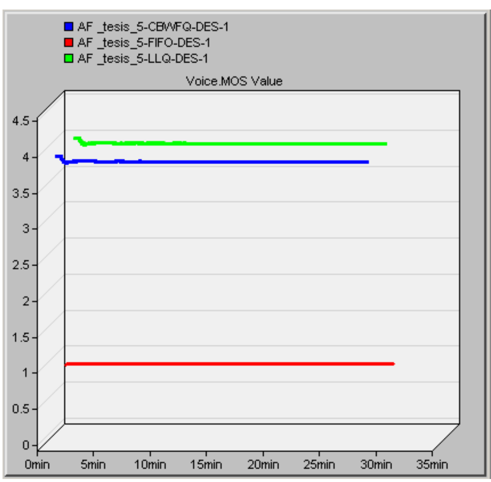

Fig. 13. MOS of FIFO, CBWFQ and LLQ

By refers to ITU-T P.800 recommendation, from result of average value of MOS for CBWFQ obtained value “4”. While at LLQ obtained by value at “4” also. Differ from FIFO at value “1”. Thereby, CBWFQ and LLQ have “Good” in opinion, while FIFO have “very bad” in opinion.

Related research conducted by Miftah Rahman [16]. In this study, Miftah Rahman analyzed the network quality in the network that integrates VoIP by comparing various kinds of encoding such as G.711 and G.729. Any study found that the congestion management method LLQ produce better delay (time delay is more stable) than CBFWQ method. It is based on Priority Queuing features adopted by the LLQ.

Related to this research, analysis of network quality in the network that integrates VoIP, based on parameters of delay, jitter, and MOS, obtained that some of the LLQ voice data congestion management method transfer quality produce better, characterized by a time delay which is smaller than the CBFWQ method. Then, in the qualitative parameters, MOS (Mean Opinion Score) of LLQ and CBWFQ at better than FIFO

V. Conclusions

The conclusion of this research is that by using LLQ scheduling technique, Quality of Service condition can be better, so that for the voice application which is sensitive to delay still fulfill voice delay standard even network in a overload state. Besides, with same configurations, LLQ have very good QoS for voice more than FIFO and CBWFQ.

References

[1] Aboelela, Emad. Network Simulation Experiments Manual Second Edition. ISBN: 978-0-12-373974-2. Morgan Kaufmann Publishers is an imprint of Elsevier. USA. 2008.

[2] Bates, Regis, J.Bud. Broadband Telecommunications Handbook, Second Edition. Mc-Graw Hill Companies. ISBN 0-07-139851-1. USA. 2002.

[3] Rao, K. R. et.al. Introduction to multimedia communications : applications, middleware, networking. ISBN 13 978-0-471-46742-7. John Wiley & Sons, Inc., USA. 2006.

[4] Santiago, Alvarez.”QoS for IP/MPLS Networks”. USA: Cisco Press. 2006.

[5] Bevilacqua, D.M. et.al. Modeling the Performance of Low Latency Queueing for Emergency Telecommunications. Proceedings of the 2007 Winter Simulation Conference 1-4244-1306-0/07 (pp. 2266-2275). 2007.

[6] Dekeris, B. et.al. Analysis of QoS Assurance Using Weighted Fair Queueing (WFQ) Scheduling Discipline with Low Latency Queue (LLQ). Intl.Conf. Information Technology Interfaces ITI, 19-22 Juni di Cavlat Croatia. (pp. 507-512). 2006.

[7] Gang Su, et.al. QoS guarantee for IPTV using Low Latency Queuing with Various Dropping Schemes. International Conference on Systems and Informatics (ICSAI 2012) 978-1-4673-0199-2/12 IEEE (pp. 1551-1555). 2012.

[8] Kristof, Richard (Co-Publisher) et.al. Administering Cisco QoS for IP Networks. Syngress Publishing, Inc. USA. 2001.

[9] Velmurugan, T. et.al. Comparison of Queuing Disciplines for Differentiated Services using OPNET. International Conference on Advances in Recent Technologies in Communication and Computing. IEEE Computer Society 978-0-7695-3845-7/09 (pp. 744-746). 2009.

[10] Akhtar, S et.al. Performance Analysis of Integrated Service over Differentiated Service for Next Generation Internet. JCIT, ISSN 2078-5828 (print), ISSN 2218-5224 (online), volume 01, issue 01, manuscript code: 100717 (pp. 95-101). 2010.

[11] Fischer, M.J. et al. Approximating Low Latency Queueing. IEEE Computer Society 978-0-7695-3162-5/08 (pp. 188-194). 2008.

[12] Guo, Zirong dan Huaxin Zeng. Simulation and Analysis of Weighted Fair Queueing Algorithms in OPNET. IEEE Computer Society 978-0-7695-3562-3/09 (pp. 114-118). 2009.

[13] Sczilagyi, S. Analysis of the Algorithms for Congestion Management in Computer Networks. Carpathian Journal of Electronic and Computer Engineering 6/1. ISSN 1844 – 9689 (pp. 3-7). 2013.

[14] Vegesna, Srinivas."IP Quality of Service". USA: Cisco Press. 2001.
[15] Prasetyawan, Hendra."Analisa Kinerja VoIP Client Dengan Menggunakan Modul RTP Terenkripsi Dan SRTP Pada Jaringan VoIP". Master Thesis at Universitas Mercubuana: Jakarta. 2014.
[16] Rahman, Miftah. Analisa Quality of Service (QoS) IP Telephony dengan metode Low Latency Queuing (LLQ).Master Thesis at Universitas Mercubuana. Jakarta. 2013.
[17] Svensson, Tommy and Alex Popescu. OPNET Modeler: Development of laboratory exercises based on OPNET Modeler. Master thesis of Electrical Engineering on Telecommunications and Signal Processing. Blekinge Institute of Technology. 2003.
[18] www.riverbed.com
[19] www.point-topic.com

Blood Provision and Service Application for People in Manado Using Mobile Web Technology

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Abstrack— the development of technology is increasing rapidly. These developments have a huge impact on all areas, including The Health Field. Blood is one very important component in supporting the health of the human body. The lack of information about blood stock availability in hospitals and the Indonesian Red Cross (PMI) makes people often have difficulty in getting the blood they needs. Today many people are willing to donate blood to help people who in needs for blood, but time and cost became their main obstacle, because they need to find the nearest hospital or Indonesian PMI to do so. This application is built to be able to help people get information about existing blood stocks and provide media that can deliver information to people who want to donate their blood. This application is built using Hypertext Preprocessor programming language (PHP). The methodology use is Rapid Application Development (RAD) and using UML (Unified Modeling Language) as a modeling tool. The test result showed that this application is properly function and this application can assist the people in ordering the blood it needs and the people who wants to donate their blood without having to go to the nearby hospital or PMI. Keywords: Blood, Blood Stock, Mobile Web.

I. INTRODUCTION

The development of today's world demands a very high technological progress in various fields, one of the field is health. Information technology is needed by everyone to improve the provision of information so as to facilitate the public in obtaining information in the field of health. Blood is a vital component that is necessary for human health.

The human need for the availability of blood stock both in hospitals and in the Indonesian Red Cross (PMI) is higher than ever, because of the increased in accidents and sickness that need blood.

People often have difficulty in finding blood stock, due to the availability of stock of blood in hospitals and in the PMI were limited. The supply of blood is very limited, this is not caused by the unwillingness of people to donate, but more to they don't know where to go or what are the requirement to be able to donate their blood. On the other hand people who need bloods don't have the information about the bloods availability or know where to go to find the specific blood they need. Those problems are prove to be not efficient and effective because it will take a lot of time and money.

Based on the description of the problems that exist, then there is a need to build an application service and the provision of blood, especially for people in the city of Manado which can be accessed on mobile phones that have a browser that will be built based on mobile web that can help get information about the availability of blood stock in the house sick and PMI, as well as to facilitate the public to get media that can distribute information blood donation, so as to help meet the needs of blood stock that is often lacking in the city of Manado.

Based on the description of the problems above, then there is a need to build an application that can be accesses via mobile phone through browser, which can help society by providing information about blood stock in Hospital or PMI and also information about where people can donate their blood, and eventually will help solving the shortage of blood stock in Manado.

II. OBJECTIVE

To create an application that can be access from mobile and web that can help the society in Manado to get information about place, requirement, and stock of blood in hospital and PMI in Manado.

III. LITERATURE REVIEW

A. Web application

Web applications originally built only by using a language called HTML (HyperText Markup Language), and the protocol used is called HTTP (HyperText Transfer Protocol). In the next development, a number of scripts and objects were developed to expand the capabilities of HTML. Those scripts are PHP and ASP and object are Java applet. The web application itself can be divided into two, namely static and dynamic web [1]..

B. Blood

Blood comes from the word hemo or hemato derived from the Greek word haima which means blood. Blood is a liquid that serves to send substances and oxygen needed by the body tissue, transporting chemicals products of metabolism, as well as the body's defense against viruses or bacteria [2].

Human blood is a liquid body tissue, its main function is to carry oxygen which is needed by cells throughout the body. In detail, the function of blood is as follows:

1. Respiration : oxygen and carbon dioxide carrier
2. Nutrition: the carrier of nutrients
3. Excretion, i.e. transport of urea to be excreted
4. Plays a role in maintaining the body's acid-base balance
5. Regulate the balance of water between plasma and tissue fluid
6. Regulation of body temperature: heat formed by the metabolism in tissues is carried by the blood to the body surface to be emitted to the outside.
7. Protection against infections: antibodies, leukocytes
8. Transport of hormones and enzymes.

C. Blood Type

Blood type is a special characteristic of blood from an individual because of different types of carbohydrates and proteins on the surface of red blood cell membranes. Two types of blood classification which is important are ABO and Rhesus (Rh factor) [3].

D. Rhesus

Rhesus System was divide into two groups, namely:

1. People with Rh-positive (Rh +), mean blood has Rh antigen as indicated by the positive reaction or erythrocyte clumping occurs at the time of the tests with anti-Rh (Rh antibodies).
2. People with Rh-negative (Rh-), mean blood does not have Rh antigen as indicated by the negative reaction or clotting does not occur when testing with Rh (Rh antibodies) [4].

Blood was collected in sterile plastic bags and anti-blob containing material for blocking blood clots and also preservative to ensure red blood cells keep alive. Blood can be stored up to 42 days and must be treated and cooled properly [5].

E. System Development Methodology

Development of this application will follow the stages in accordance with the methodology of Rapid Application Development (RAD). According Soetam (2011), RAD (Rapid Application Development) is a software implementation method that prioritizes speed and flexibility in it. This method aligned with the development of programming languages which is leads to the concept of reusable and visual programming. Of course, this method is very difficult to implement if application is developed with the "old-fashioned" programming language like Assembly language [6].

IV. ANALYSIS AND DISCUSSION

F. Identifying target User

Table 1. List of Targeted User

List of the target user	Role	Responsibility
Application Administrator	*Admin*	- Controlling all existing data in the application - See reports of blood stock in the hospital, PMI. - Confirm acceptance of new users in the application.
Hospital	*User*	- Entering information about blood stock in Hospitals. - Can view blood stock information in hospitals and Red Cross. - Provide confirmation for people who ordered the blood that is in the hospital.
PMI	*User*	- Entering information about blood stock in PMI. - Can view blood stock information in hospitals and Red Cross. - Provide confirmation for people who ordered the blood that is in the PMI.
General Public	*User*	- Entering data about blood donation - Can view blood stock information in hospitals and Red Cross. - Provide confirmation for people who ordered the blood. - Provide information regarding the urgent need for blood.

G. Use Case Diagram

This stage contains a clear view of the interaction between user and the system using use case diagram.

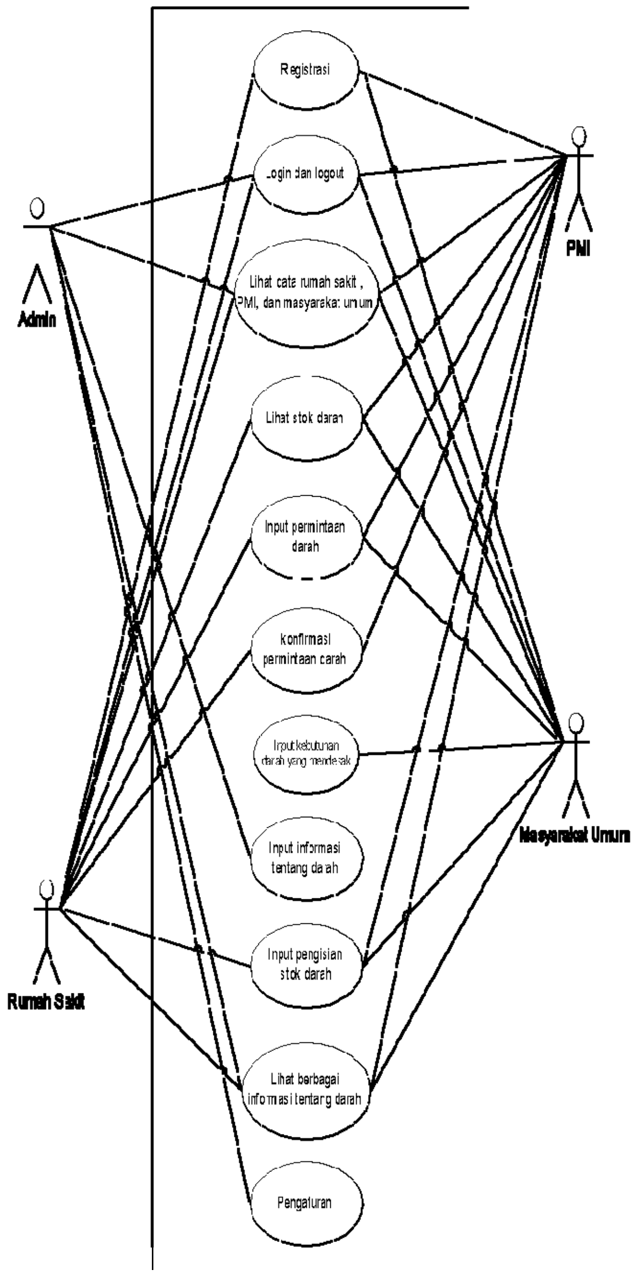

Figure 1. Use Case Diagram

H. User Interface

In this figure below is the implementation of the user interface from finished application.

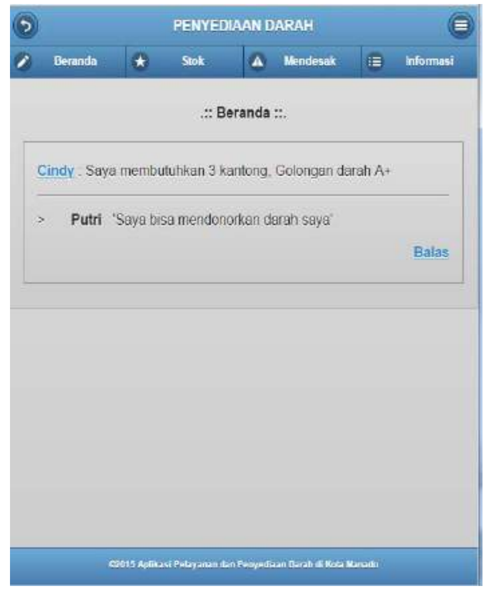

Figure 3. Home Interface of the application

This figure shows the interface for donating blood.

Figure 4. Interface for Blood donating page

This figure shows the interface to enter blood stock in hospitals.

Figure 5. Interface for adding blood stock for hospital.

V. CONCLUSION AND SUGGESTION

I. CONCLUSION

Based on the applications that have been created, the author can draw the conclusion as follows:

1. This application can provide convenience for the general public in obtaining information about blood stocks in hospitals, PMI, or from the general public who wish to donate blood.
2. This application can also help to deliver information to people who urgent information about blood need stock because the blood required is not available in the hospital or at PMI.
3. This application can help general public to order blood without have to go to hospitals or PMI.

J. SUGGESTIONS

For further development, the authors suggest a few things:

1. Payment feature can be added to each blood reservation
2. Can give information about the health condition of donor and people who need blood.
3. Can address a maximum limit for the storage of blood in hospitals and PMI.

BIBLIOGRAPHY

[1] Hidayat, H. (2011). *Cara Instan Menguasai Pemrograman Web*. Jakarta: Agogos *Publishing*.

[2] dan [3] Nadia, B., Handayani, D., Rismiati, R. (2010). *Hidup Sehat Berdasarkan Golongan Darah A,* Jakarta: Dukom *Publisher*.

[4] Azhar, F. N., Madona, P., Tianur. (2014). Alat Pembaca Golongan Darah dan Rhesus. *Jurnal Teknik Elektro dan Komputer*. Vol. 2, No.2, pp 146-147.

[5] Toh, S. C., Ong, A. S., Huat, L. M. (2010). *Bilingual Express Science Form 3*. Malaysia: Pelangi.

[6] Soetam, R. (2011). *Konsep Dasar Rekayasa Perangkat Lunak*. Jakarta: Prestasi Pustakarya.

Artificial Immune Wireless Intelligent Sensor and Actuator Network (WISAN) for More Electrical Aircraft Performance Monitoring System (study case : 80 Passenger Aircraft)

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***Abstract*— The issue of this paper is about computer development in the field of aviation. This is in accordance with the technological development of aircraft that shifted from manual to computer-based conditions. Many things can be learned about computerization in the aviation world especially associated with the control mechanism of the aircraft or aircraft communications system. The research related to control of aircraft by utilizing wireless technology and computers. This control technology embeds a special system on another system-computer based. Intelligent system is a system that is filed with artificial immune algorithm. This algorithm is expected to be known to prevent all potential failures. This control technology is intended to be implemented on electrical-based aircraft. Redundancy became a matter of support in obtaining reliable and robust performance of system . Sensor system with the support of the microprocessor as a data processor becomes major system of this control system. Setup and configuration of sensors with certain consideration, good data communication systems and realtime applications become necessary to obtain the expected outcomes. Wireless data communication based on certain setting and proper topology is important thing to support the aircraft control system. System to be built also considers the existing standard computer on airplanes**

Keywords—component; artificial immune, wireless communication, aviation , sensor

I. INTRODUCTION

At this time the aircraft into a means of transportation is very important and widely used. Indonesia is an archipelago with large population . The need for air transportation is very large with various destinations. To fulfill this requirement, the national aircraft producer will manufacture aircraft with medium size is around 80 to 100 passengers. This aircraft will be made in the hope of flights to many places without having to wait long because the quota has not been fulfilled. The problems that will be faced by aircraft are inadequate runway condition, generally short runway, and weather . This situation demands endurance and toughness aircraft. Many events such as accidents caused by the poor condition of the aircraft.

Several things to consider in planning the manufacture of the plane with the goal of efficiency in aircraft performance and passenger comfort are modern technology of energy-saving and automated systems. The selection of technology used is expected to improve the performance of aircraft, including flight safety. Safety-critical systems is part of airworthy that has to pursue

The situation of the world that begins to enter the era of energy crisis is also affecting the availability of air transport. Nowaday aircraft generally using turbo engine or jet propulsion which uses liquid fuel combustion system. With shortages of fossil fuels then people look for a system that can support these conditions, using an electric-based machine jet engine apart. Since the addition of an aircraft engine system is electricity base then known term "More Aircraft Electrical " (MEA). Modification not only in the primary system in the engine system, but also on the monitoring system. The monitoring system improvement relates to the performance of the machine and other system will use wireless network as a medium, sensor as a monitor device and actuator as controller device.

The next generation aircraft system which is dominated by electricity should have a better system reliability.This aircraft technology development requires extensive and complex study . For further the air transport known as a high risk transportation system . It can be cultivated with the monitoring mechanism and good control. Many things can be assessed both in terms of flight control systems, and avionics etc.

The system complexity has led to an increased desire for aircraft system surveillance. The sensor and actuator base monitoring system can be propose as a solution. A number of sensors of varying types can be mounted on the engine, or inside the engine to sense various physical parameters.Actuator can be mounted to support the aircraft system. Many concepts of technology can be apply in sensor and actuator systems, all intended to the reliability and stability of the system in an aircraft. This was also followed by the use of devices that support. Working methods and algorithms in wireless networks based monitoring system will determine the quality of the performance of the system. This system is called wireless intelligent sensor and actuator network (WISAN).

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Plane crash that occurred because of the bad condition of the aircraft is the focus of a problem here. Several accidents that happened in Indonesia, August2, 2009 Merpati Airlines ,the case is uncontrolled flight into terrain , November 30, 2004, the case is runway overshoot. Build a wireless network based monitoring system into the proposed solutions to solve these problems. It becomes important things to be discus. Selection of the technology will be related to the type of aircraft being built. Large and medium size aircraft will be different in terms of complexity management. In determining the sensor and actuator devices and their accompanying technologies will be affected by this type of aircraft. In this paper will be developed monitoring system which embed in other systems in aircraft electrical based. The implication of the proper management of monitor and control is important due to aircraft system robustness

WISAN usage with good management can help to realize the reliable condition. The network management scheme uses a particular method or algorithm. In terms of alertness, it can be selected as a method of fault detection, this is in accordance with the algorithm of artificial intelligence, namely the immune system (ie a system that is adapted from the physiology of the human body)

II. Problem Statement

A. *Safety Flight*

- The major issue is how to obtain good performance of 80 passenger aircfraft related to the safety flight ? Because it is important to assure that this new technology will guarantee the safety and comfort . Therefore, it will be used a robust monitoring and controlling system based on wireless networks that utilize sensors and actuators with certain method
- What kind of method and algorithm will be propose related to robust and reliable monitoring and controlling system?
- What proper parameters will assist in obtain the desired performance that is robust and reliable system

III. Research objective

The aims of this research is increasing the reliability of more electrical aircraft by using robust performance monitoring system. The Objectives of this research is increasing the reliability of more electrical aircraft by using robust performance monitoring system WISAN based with artificial immune algorithm

The objective and the scope are:

- To propose a technique called Wireless Intelligent Sensor and Actuator Network to improve the More electrical aircraft system
- To develop monitoring system based on the proposed technique with artificial immune algorithm
- To evaluate and analyze the performance of the proposed technique.

Delimitation of the reseach

- not discuss improvements to the main system MEA
- discuss subsystem MEA
- do not discuss the security system monitoring and control

IV. Literatur View

The condition of the world oil crisis pushed aircraft manufacturers switch on the new technology. The new aircraft technology is replacing mechanical devices, hydraulic, pneumatic becomes more electrically based. Therefore, this concept in aircraft industry has tried to considerably reduce the impact of these trends, by increasing the implementation of electrical components in aircrafts for generating, distributing and utilizing electrical power. Thus, the more electric aircraft (MEA) concept impacts, significantly on aircraft electrical power system design,can reduce size and weight, and improve fuel efficiency .

A. Conventional aircraft concept in aircraft

- AC power Generation

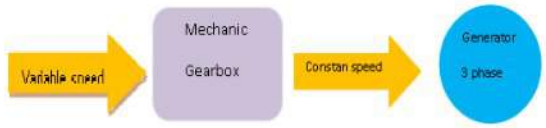

Fig 1. AC Power generation Conventional aircraft

Mechanical gearbox creates a constant speed shaft from a variable speed input, then Constant speed shaft drives the Generator.

- Several subsystem and power source in conventional aircraft :

Electrical :

Avionics

Cabin (lights, galley, inflight entertainment etc)

Lights, pumps, fans,115V, 400Hz AC

Pneumatic

Cabin Pressurization

Air conditioning

Icing protection

Hydraulic

Flight control surface actuation

Landing gear extension/retraction and steering

Braking

Doors

Mechanical

Fuel and oil pumps local to engine

B. More Electric Aircraft

- Concept of More Electrical Aircraft
  - Removal of hydraulic system, reduced system weight and ease maintenance.
  - Bleedless engine, improved efficiency

- The benefit characteristics of electrical Systems
  - controllability
  - power on demand
  - reconfigurability
  - maintain functionality during faults
  - advanced diagnostics and prognostics
  - more intelligent maintenance
  - increased aircraft availability

- AC power generation

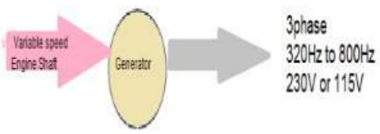

Fig 2. AC power generation More Electrical Aircraft

Generator provides variable frequency supply and controls variable voltage from variable speed engine shaft. Direct connection between generator and power bus

- **More Eelectrical Aircraft**

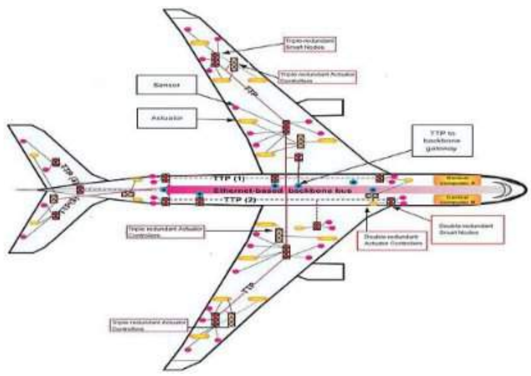

Fig 3 More Electric Aircraft Sub system

Source: aerospace Engineering Magazine

Conventional Aircraft Subsystem Architecture:

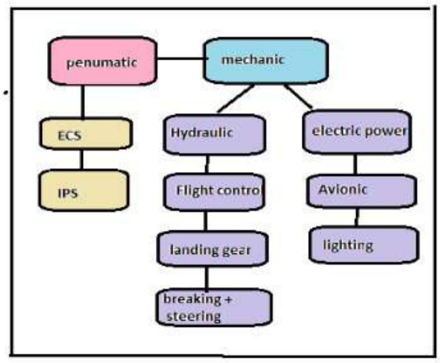

Fig 4. Conventional Subsystem aircraft architecture

IPS : Ice Protection System

ECS: Enviromental Control System

More Electrical Aircraft Subsystem Architecture

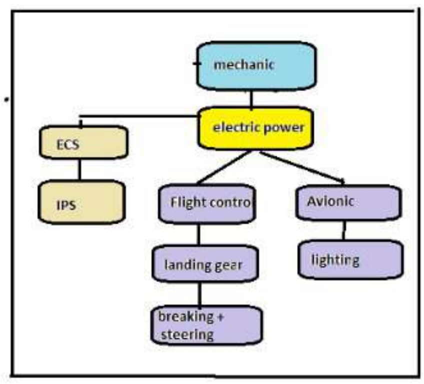

Fig5 .MEA Subsystem Architecture

On conventional aircraft energy conversion flow long enough, while at the MEA relatively short. overall the condition can reduce the use of power. Subsystem on aircraft includes many things, among others, to control the flight consisting of aileron, rudder, elevators etc. The device is in the workings of uses pressure, and the motor. In addition, for many environmental monitoring assisted by the sensor.

Subsystem work with certain patterns and mechanisms based on the concept of appropriate work. Subsystems support device that initially work more mechanical hydraulic or pneumatic mechanical then begin to shift in the electrical mechanic. Therefore a monitoring system will also be adjusted

SENSOR & ACTUATOR

Aircraft subsystem uses many sensors and actuators as a monitor or control device. Generally the physical parameters not only one kind but various types , eg electro-mechanical (EMA), or hydraulic –mechanical

C. Actuation system

Conventional actuators like EMA continue to improve. Smart materials actuators such as piezo-ISA are considered to be mature, and are being used more and more in small equipments.

- **Actuator arrangement redundancy management concept :**

1. Fault tolerant actuator by actuator redundancy

One weakness of electromechanic that have to be considered as credible failures is jam failures. To anticipate this failure, it is important to design redundancy into actuation systems of such safety critical applications. We use actuator redundancy concept to ensure safe operation of primary flight control . Using a conventional actuator topology, any failure can make other system inoperative. By applying redundant control channels and feedback sensors , the system still has to be capable to localize and isolate a failure for continued safe operation.

2. Optimizing redundancy

The required number of redundant components in safety critical actuation systems can be determined by combining the quantified failure modes of mechanical and electrical components . At the time of making the device design redundant on, then previously pursued forecasts the possibility of failure occurs while the system is still able to overcome. The approach can be used with mapping permutations. At the time of making the device design redundant on, then previously pursued forecasts the possibility of failure occurs while the system is still unable to overcome. The approach can be used with mapping permutations, hence even there is fail the system can operate,this called fail safe. It is the result of system optimization.

3.Safety analysis

Safety of a system is very important. A safety-based analysis begins with knowing the condition of what should happen, then also know how the conditions were not good. In addition, safety analysis also associated to address the safety automatically.

D. Sensor System

High-Precision Node Design

High-precision means the testing random error is small . Precondition of strain measurement has low-voltage and varying-load features, the series reference scheme is selected for better initial tolerance, low temperature coefficient and power dissipation , high sensitivity . The quality of sensor can be modificate by add amplifier to the sensor instrumentation amplifier. Since strain gauges are usually adopted to monitor static signals, a low-pass filter is designed to eliminate the high frequency noise. The voltage follower is adopted to output the filtered voltage signal.

State-of-the-art in WISAN node platforms

- MSB430,
- ESB
- TmoteSky
- Mica2Dot
- Imote2

E. Wireless Intelligent Sensor Network

Wireless Intelligent Sensor and Actuator Network (WISAN) as a scalable wireless platform.Wireless sensor and actuator network are comprised of large numbers of minimal capacity sensing, computing, and communicating devices and various types of actuators . This network use protocols to collect, transport, and perform sensor fusion of data and react with control actions. The instrument that support WISAN have to conform IEEE802.15.4 standard. Network works by using a specific algorithm intelligently. By studying the conditions based on the processing of raw data in order to obtain a symptom or conclusions which later became a particular pattern. This system is capable of delivering real-time streams of sensor data . WISAN can work with heterogeneous sensors and actuators and combines with analog or digital interfaces.

- WISAN characteristic:

1. WISAN can be utilized worldwide in 2.4Ghz ISM frequency band and coexists with WiFi and other devices.
2. The sensor nodes perform energy-efficient distribution of computational load by performing on-
3. board compression, signal generation for actuators and evaluation of fuzzy rules.
4. Each node on the network is synchronized with other nodes on the order of microseconds. The synchronization is periodically updated. This guarantees data acquisition by all nodes at the same time.

5. Bandwidth utilization is maximized through bandwidth scheduling allowing to increase number of nodes in the network by factor of five.

6. Hardware design of the nodes allows to perform self-localization of the nodes, that is establish relative position of each node to others.

- **WISAN Performance Parameter**

1. Energy Awareness:

Self-sustaining power sources and minimization of energy consumption by using low power consumption sensor. Center of data processing use ultra low power microcontroller

2. Self-localization of sensor nodes**.**

For mobile or static sensor nodes in large networks Tracking device placement can be quite challenging. Selflocalization of sensor nodes using Radio Signal Strength Indicator (RSSI) method .

3. Fault-tolerant

WISAN is based upon IEEE802.15.4-compatible transceiver which addresses these issues. This system should coexist with another wireless communication system. in order to support such diversity, the important thing is to pay attention to interoperability system indicated by the level of fault tolerance

4. Error detection energy spectrum and usage

Wireless sensor should also be equipped with an intelligent subsystem that supports error detection, especially in the distribution and use of energy..

ARTIFICIAL IMMUNE SYSTEM

Artificial immune is an adaptive system inspired by human immunology which can efficiently detect changes in the environment or deviations from the normal system behavior in complex problems domains .Work patterns of these system are data analysis, fault and anomaly detection. This system is used to anticipate bad things happen to a full system with detection and if it is found that there is anomaly process then the full system can be reconfiguration or error mitigation.

- **Comparison Immune System between Human and Sensor network**

In human body , immunity component B cells produce antibody. Tcells recognize and kill virus. B cell like sensor nodes , and T cells similar to rate control. Antibody act like data sensor with certain density or frequency rate

Artificial Immune System (AIS) algorithm is also compliance for aircraft fault detection, as an extension to a intelligent flight control (IFC). Since the IFC approach uses fault accommodation with no detection, causes degradation in handling qualities. If the failure can be identified, this can befurther minimized to ensure fault accommodation and better handling qualities.

The approach knowledge of the normal operational behavior of the aircraft from sensory data, and probabilistically generates a set of pattern detectors that can detect any abnormalities , indicating unsafe in-flight operation.

- Fault detection method immunity based

Once the fault is detected and identified, an adaptive control system would use this detection information to stabilize We experimented with data sets collected under normal and various simulated failure conditions. The reported results are from a collection of test cases that reflect the performance of the proposed immunity-based fault detection algorithm.

Various Fault detection method immunity based e.g:

· Real Value Negative Selection Algorithms

· Clonal Selection Algorithm

· Discrete Immune Network Models

1. Real-Valued Negative Selection (RNS) Algorithm

This method uses rules of comparing a condition with other conditions that are known to be wrong. If the closer / similar symptoms occur, the system is said to fail and must be quickly addressed before it becomes catastrophe.

Stages in the working mechanism of negative selection:

a. Define self as a collection some way models the normal operation , Generate a set *detector randomly* , each of which fails to match .

b. The radius of a particular detector is defined in terms of Euclidian distance to its nearest neighbor in the training dataset

c. A greedy algorithm is used to select a new population of detectors,

d. If any detector ever matches, then an abnormality is known to have occurred, because the detectors are designed not to match any similar sample

2. Clonal Selection Algorithm

Randomly initialise a population , for each pattern in Antigenic then select base on affinity leveland certain mutation rate to form part number of random new ones.

3. Discrete Immune Network Models

a. create an initial network from a sub-section of the antigens

b. determine its stimulation level (based on antigenic and network interaction)

c. eliminate cell with a low stimulation

d. select mutated clones and integrate

- **Intelligent Flight Control**

Flight control architecture has been investigated for both flight and propulsion control the Intelligent Flight Control (IFC) system . Fault detection using real valued data set as input , and will be extract by applying data fusion and normalization techniques.

- **Aircraft Landing System**

As the plane was landing a lot of things that must be considered, related to the weather, the physical condition of the field, aircraft weight, adequate air devices etc. So that the parties involved are also quite a lot, pilots, air traffic control, sensor as the input source. Rules and certain count also accompany the decision-making plane landing.

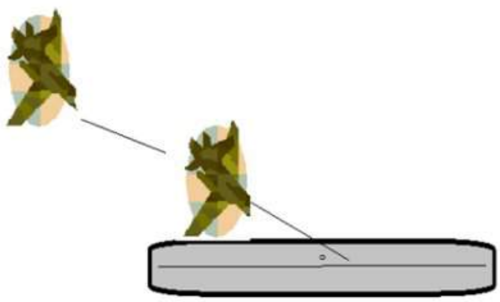

Figure.8 Landing

Equation:

Minimum Runway = Actual landing distance/ 60 %

Minimum Runway when WET =

Actual landing x (115%/60%)

V. CONCEPT

Several recommendations were made to facilitate the development mechanism for example : building system model, using an estimator with a variety of approaches and algorithms, using the simulator and emulator so that the picture of the actual system may be obtained with a certain degree of deviation. System will be design with fault detection artificial immune based and negative selection algorithm , in addition recent condition of aircraft e.g landing system .

On this paper will be submitted a monitoring and control system using a wireless-based sensor as data takers in real time and actuators, wherein the data management follows the rules-based negative selection and artificial immune. The system also noticed the problems that often occur in the form of an accident while landing. Systems embedded in Intelligent Flight control also makes the condition of the runway into considerations when landing aircraft in order to survive.

A.Analyze and Design

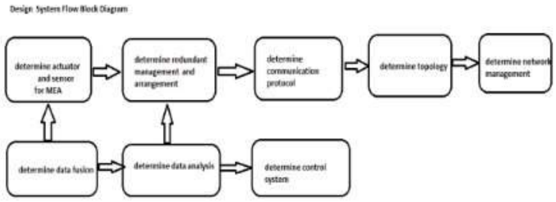

Fig 9. Design System Flow Block Diagram

For the medium category aircraft, selection and use of sensors and actuators to be adjusted. Use of IEEE 802.15.4 protocol for communication with devices such as sensors that support Tmote, realtime operating system and software application support or electronic flight bag system may be a wireless network of sensors and actuators are pretty good. Sensor systematically assembled redundant and do not all optimization measures and modifications resulting in maximum performance. The use of algorithms in data processing as well as preventive and accompanied by techniques of artificial immune-based fault detection negative selection will make the system work faster. Pilot parameters as input source is also a determinant of the accuracy and precision in decision making. Efforts to increase the safety of landing to anticipate the accident which occurred to the conditions when landing airfield into an added value and novelty proposed system to be built

- Wisan model system

Wisan modeling as a means to monitor and control based on the objectives to be achieved, namely robust system in which the system is expected to be aware of any deviations or errors which in turn will be able to be fixed on the main system. Work patterns of artificial immune approach where the sensor is not only receive data, but also have the appropriate intelligence desired criteria because the sensor is equipped with a microcontroller. System is also equipped with a microprocessor that can process data and make decision

The research in this proposal aims to develop a Wisan based medium size aircraft system design. For 80 passenger aircraft ,a small-size, high-precision is designed for multi-channel strain gauge Signal conditioning and monitoring. To address the need for low-power consumption, timeliness, and scalable operations, a cluster-star network topology protocol is adopted and researched.

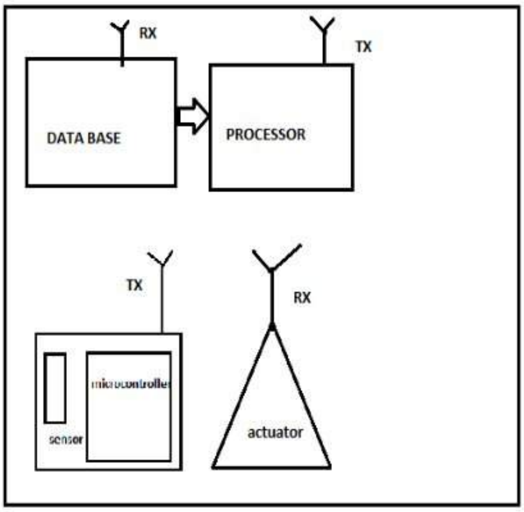

Fig 10. WISAN block diagram

- **Integration System**

An aircraft engine wisan includes a number of wireless sensor communication nodes and a central engine control unit (i.e., engine controller) in operable communication with each other. Each wireless sensor communication node can communicate directly with the engine controller, or through one or more other wireless sensor communication nodes in the network.

- **Actuator System**

Primary Flight Control Actuation System consist of flight surface on the airplane,use smart actuator e.g :

- aileron servo actuator

-flaperon control & modul

-elevator servoactuator

-rudder servoactuator etc

The electromechanical actuator (EMA) – an electric motor,gearbox and mechanical actuator has the potential to be more compact than an electro-hydraulic device, due to the absence of a reservoir EMA is a higher efficiency actuator but only relatively recent developments in motor and power electronics have made the EMA just for high power density and safety concerns over jamming.

The actuator reliability can be determine by fault analysis, concern about electronic component reliability that is indicated by MTBF. Inspite to estimate the failure probability of a gearbox, it is assumed that bearing failures . To mitigate against electronic drive failures, inspite of single lane dual-lane fault-tolerant electric drive architecture can be incorporated into the EMA. To eliminate the gearbox, direct drive has been considered in EMA extension

Fault-tolerant electric drive configurations for an EMA

In EMA the variable speed nature necessitates the use of power converter-fed drives; line start induction machines, induction and reluctance motor drives. The fault tolerance perspective have to be analyze in partitioning the machine into independent fault-tolerant units which eliminate magnetic coupling between units. This is aparticular problem when short-circuit faults occur.

Sensor System

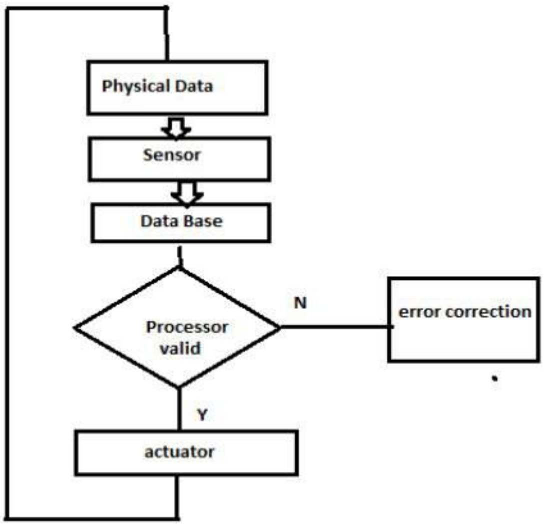

Figure 11.Sensor System

The MEA use high quality of sensors, regard to sensitivity, linearity, and size relate to wight reduction. The sensor section includes one or more sensors e.g.,temperature sensors, pressure sensors, vibration sensors, proximity sensors, and position sensors, etc is linked by appropriate signal.

Sensor node classification and configuration .

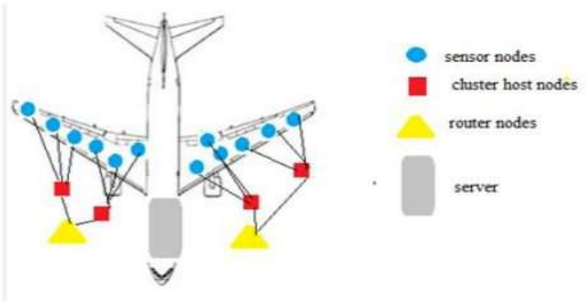

Figure 12. Sensor Network Topology in Aircraft

The desired outcome of this research is the realization of MEA that has good performance, so it can be used comfortably and safely. As an initial step MEA research is aimed at aircraft with only 80 passengers. Good performance of the MEA is not a simple matter, because it must be viewed from many angles. In this study will be evaluated from the construction of performance monitor and control system of some subsystem of the MEA, which although not exhaustive but contributes to the improvement of MEA technology

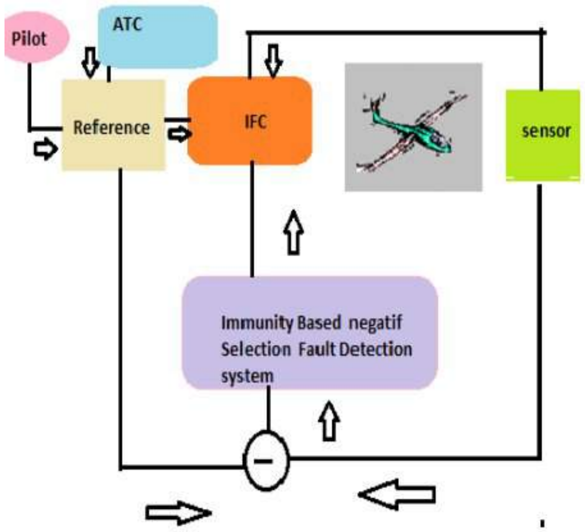

Figure 14. Block diagram Artificial Immune Wireless Intelligent Sensor and Actuator Network (WISAN) for More Electrical Aircraft Robust Performance Monitoring System

VI. CONCLUSION

For the medium category aircraft, selection and use of sensors and actuators to be adjusted . Use of IEEE 802.15.4 protocol for communication with devices such as sensors that support Tmote, realtime operating system and software application can build a wireless network of sensors and actuators. Sensor systematically assembled redundant and do not all optimization measures and modifications resulting in maximum performance.

The use of algorithms in data processing as well as preventive and accompanied by techniques of artificial immune-based fault detection negative selection will make the system work faster. Pilot parameters as input source is also a determinant of the accuracy and precision in decision making. Efforts to increase the safety of landing to anticipate the accident which occurred to the conditions when landing airfield into an added value and novelty proposed system to be built

REFERENCE

[1] DIGITAL AVIONICS HANDBOOK, THIRD EDITION.CARY SPITZER, UMA FERRELL, THOMAS FERRELL

[2] Integrating Subsystem Architecture Sizing and Analysis into the Conceptual Aircraft Design Phase, Cooperation between GA Tech's Aerospace Systems Design Laboratory (ASDL) and PACE, ImonChakraborty Senior Graduate Researcher, David Trawick Graduate Researcher, Dimitri Mavris Boeing Regents Professor of Advanced Aerospace Systems Analysis & Director, Mathias Emeneth Senior Business Development Manager, PACE America Alexander Schneegans Managing Partner & Co-Founder, PACE GmbH, President, PACE America

[3] Wireless Intelligent Sensor and Actuator Network (WISAN): a scalable ultra-low-power platform for structural health monitoring , Edward Sazonov*a, Ratneshwar Jha b, Kerop Janoyanc, Vidya Krishnamurthya, Michael Fuchsc, Kevin Crossba Dept. of Electrical and Computer Engineering, b Dept. of Mechanical and Aeronautical Engineering ,c Dept. of Civil and Environmental Engineering Clarkson University, 8 Clarkson Ave, Potsdam, NY, 13699

[4] Network platforms, Devices and testbeds ,Eitan Omiyi, Kaan Bür and Yang Yang . Dept of electronic and electrical engineering , University College London

[5] Immunity-Base Aircraft Fault Detection System,D.Dasgupta,K.Khrisna Kumar,D.Wong,M.Berry

[6] http://www.intelligent-systems.info/wisan.htm

Air Pollution Detection Application Using Micro-controller

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***ABSTRACT* — Air pollution can affect air quality. It makes bad impact for humans and other creatures. Therefore, it is necessary to measure air pollution levels can determine the condition of the air in an environment that humans maintain air quality. Measurement of levels of pollution in the air hasn't be done by humans without the use of a facility. Limitations of the human senses is a factor that can affect the accuracy of measurement results of air pollution. The research aim is to measure the levels of carbon monoxide (CO). Measurement of carbon monoxide gas sensor using MQ-7 will produce carbon monoxide gas values in ppm and converted by DFRduino Uno V3.0 microcontroller to be sent to the software and read the measurement result can then be saved. The methodology used is prototyping methodologies with UML (Unified Modeling Language) as system design. According to the results of testing, the application can measure the levels of carbon monoxide (CO) in an environment.**

Keywords: Air Pollution Application, Microcontroller, Sensor MQ-7, Pollution

I. INTRODUCTION

A. Background

Air is a natural resource which very influential for human life and other creatures, so it needs to be preserved, maintained, and guaranteed the functions. To maintain the quality of air, control becomes very important to do (Soedomo 2001)

One way to maintain the air quality is to reduce air pollution caused by natural sources and human activities. There are various types of air pollutants, but the most dominant is carbon monoxide or Co. (Utami 2011).

To determine the level of air pollution, a tool that can monitor air quality is needed. However, commonly used tools have not been complemented by an application that can be operated using computer. Therefore, there is a need to build an Application Detection Levels of Air Pollution and a tool that can measure the level of air pollution by using components DFRduino UNO microcontroller and
sensor MQ V3.0-7

B Statement of Problem

Based on the existing problems, the statement of problem as follows: How to build an application that can detect levels of air pollution and designing tools that can measure levels of Co?

C. Scope of Problem

1. Type of pollution that can be detected by the application is gas Co.
2. The tools that built isn't give a warning on air conditions change.

II. LITERATURE REVIEW

A. Microcontroller

A microcontroller is a digital electronic device that has inputs, outputs, and control programs can be written and erased in a special way (Winarno and Deni 2011)

B. DFRduino Uno V3.0

DFRduino Uno V3.0 is a simple microcontroller board that compatible with Arduino UNO R3 and Arduino IDE. DFRduino types of AVR microcontroller uses different colors for features I/O ports. Red for power, blue for analog I/O, green for the digital I/O. Board on Atmega 328 has 14 digital input/output pins, 6 PWM output pins, 6 analog inputs, 16 MHz crystal oscillator, a USB connection, power jack, ICSP header, and a reset button. All pins are needed to support the microcontroller (DFRobot 2014)

Figure 1. DFRduino Uno V3.0 (DFRobot 2014)

C. Sensor MQ-7 (Carbon Monoxide)

Sensor MQ-7 is a gas sensor for detecting carbon monoxide (CO) which can detect concentration of gas carbon monoxide from 20 to 2000 ppm. The sensor module is equipped with a

potentiometer to adjust the sensitivity.

Figure 2. Sensor MQ-7 (Futurlec 2013)

D. C and C# Language

According to Siswoutomo (2005: 202), C is a programming language that built in the early 1970s by Erick Thompson and Denis Ritchie for the UNIX operating system. The C language is a type of programming language that is closer to assembly language so it also known as medium-level language.

According to Kurniawan (2010), C # is a modern programming language that is general-purpose, object-oriented and can be used to create a program Microsoft .NET Framework architecture.

III. DISCUSSION

A. User Analysis

This tool and its application can be used widely. They are useful not only for organizations but also individual who want to get information about the level of air pollution in a certain place.

Tabel 1 User tasks

No.	User	Tasks
1.	Organization/Individual	Measures the level of carbon monoxide gas using the component in the carbon monoxide gas measurement tool. Results will be shown in the air pollution detector application.

B. Data Analysis and Communication

This step analyses communication data that occurred in this system. Process of the system performance from the input and output is presented in Figure 3.

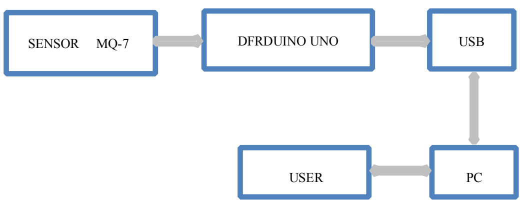

Figure 3 System Performance

Functions of the each block

Censor gas MQ-7 Block as input/the measured gas data.

Microcontroller DFR duino Uno V3.0 Block as the converter of sensor data.

PC Block as the control to begin and end the measurement as well as to display the detection results.

C. Description of the System Performance

Prior to use this application, the user is required to access the application in the computer. First, the user selects the *start* button to begin the measurement and the *stop* button when the time needed is sufficient.

Next, censor MQ-7 will detect carbon monoxide gas in the air. If carbon monoxide (CO) component is found in the air, the censor will send the received gas value to the microcontroller DFR Uno V3.0.

Furthermore, the microcontroller will record data of carbon monoxide value from the censor and transform it into the serial communication mode so as to be sent to the computer. In the display of air pollution detector, there will be measurement results of the carbon monoxide value measured before. These results provide information about the quality of air in the environment. Importantly, these results can be retained by the user.

D. Interface Display

This stage shows development of basic interface of the carbon monoxide (CO) air pollution detector application

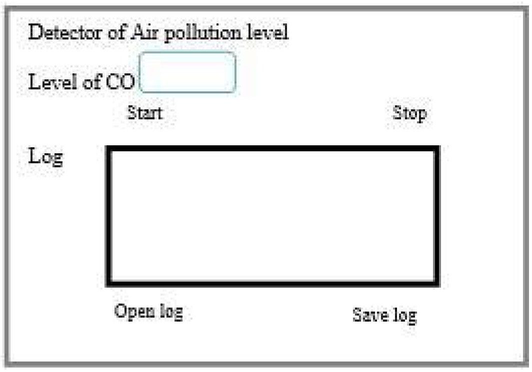

Figure 4 Display of Design Air Pollution Detector

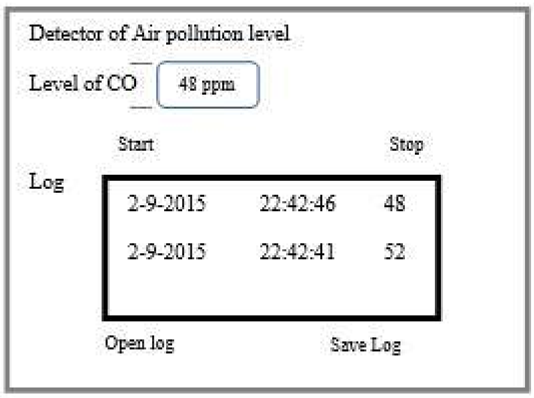

Figure 5. Display of Air Pollution Detector

E. Design of the Hardware

This section explains development of carbon monoxide (CO) air pollution detector: MQ-7 (CO) censor and

DFRduino Uno V3.0 microcontroller which connected to the computer.

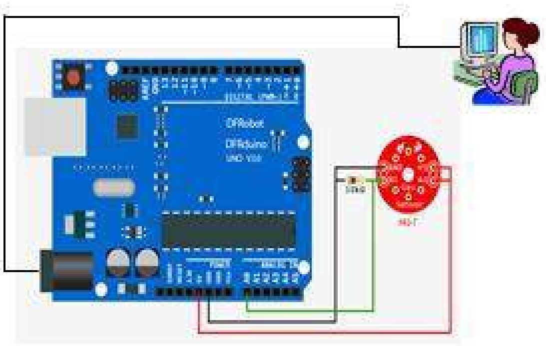

Figure 6 Hardware system

Figures 6 is the system of carbon monoxide (CO) measurement tool in which DFRduino Uno V3.0 microcontroller is connected to MQ-7 censor and computer using USB. When the user open the application and begin the measurement by pressing the *start* button, instruction is sent by the microcontroller and start the measurement of the carbon monoxide gas. Furthermore, results from the censor measurement that is in the form of analog value is taken by the microcontroller and converted to decimal and further display in the detector application of the air pollution. Results from monoxide gas measurement can be shown in ppm and time. These results can also be kept by the user.

F. Implementation of the Tools Design. This section describes design of the carbon monoxide gas (CO) tool design with microcontroller DFR duino Uno V3.0 specification and MQ-7 censor.

Figure 7. Sensor Microcontroller Circuit

G. Interface Implementation

This stage explains the application of air pollution level detector.

Figure 8 Page of the Air Pollution Level Detector

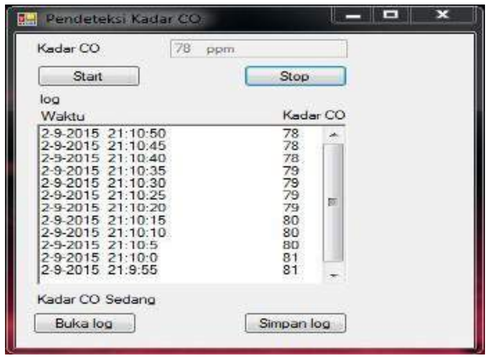

Figure 9 Results of the CO level measurement

In the page of air pollution level detector, there is a start button to begin detection process and a stop button to end the measurement. Store button will keep the all results. Next, log opener is to display output that is results and time measurement of carbon monoxide by the MQ-7 censor.

H. Implementation of the Test

Day/date: 23 July 2015
Venue: Perumahan Holly Lestary
Time: 21.37
Results: carbon monoxide gas level in the chosen place is 50. The index of the air pollution standard is 0-50. This result indicates that the air condition can be categorized as good.

Figure 10 Results of A Test

Day/Date: 23 July 2015
Venue: Jalan raya Kairagi 1 Kombos
Time: 21.10
Results : carbon monoxide gas level in this street is

92. The index of air pollution is 51-100. This means that the air condition is in the medium category.

Figure 11 Results of B test

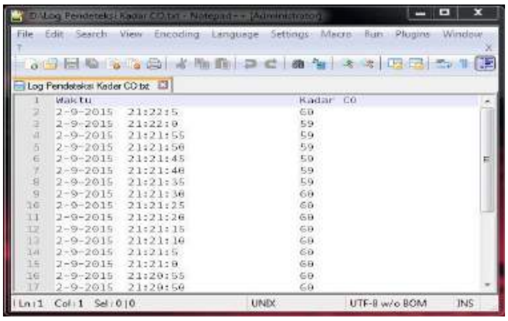

Figure 12 *Log* of CO Level

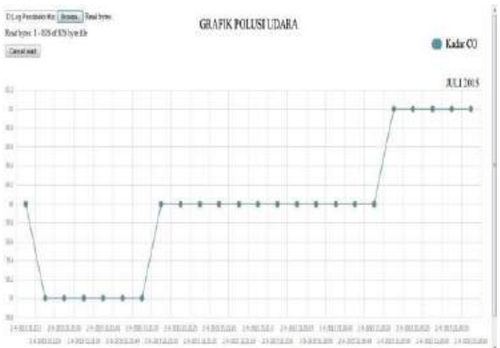

Figure 13 The graph of Air Pollution

Figure 12 shows *tracking* performance system that kept the CO level and time of the measurement per 5 second within 24 hours. Figure 13 is the display of air pollution graph which taken from *log* level CO.

I. Analysis of the Test Results

Based on the test using this application, some conclusion can be taken as follows.

1. All components developed in this study can be used appropriately.
2. The application can be well accessed.
3. Revealed results of air pollution measurement for carbon monoxide gas level and the results can be retained.
4. Measurement, display, and storing functions are well functioning.
5. This application can be used only if carbon monoxide gas measurement tool has been well installed connected to computer.

IV. CONCLUSION AND SUGGESTIONS

A. Conclusion

Based on results found in this study, it can be concluded that:

1. The application and the air pollution detector can be used as the air pollution detector and measured CO gas level in a certain environment.
2. This application can provide conclusion of air condition and show graphs as the visualization of the CO level results.
3. This application can assist user to detect the level of air pollution, particularly for carbon monoxide (CO) gas. Detecting the level of air pollution can help stakeholders to improve and maintain healthy environment..

B. Suggestions

Several suggestions to improve the system are as follows.

1. The function of this application can be enhanced by making the message sent automatically to the users who are registered in a particular area. These users will get notification about the air condition in their area.
2. To improve its function, several gas censors can be added to the tool design so as to be able to detect other pollutant gasses.

REFERENCES

1. DFRobot. (2014). DFRduino Uno V3.0. Available at http://www.dfrobot.com/2014/04 [Accessed 12 Maret 2015]
2. Futurlec. (2013). Sensor MQ-7. Available at www.futurlec.com/gas_sensors.shtml [Accessed 15 Maret 2015]
3. Kurniawan, E. (2010). *C#. Cepat Mahir ASP.NET 3.5 untuk Aplikasi*. Yogyakarta: Kawan Pustaka.
4. Siswoutomo, W. (2005). *Rahasia Para Programmer Profesional.* Yogyakarta: Elex Media Komputindo.
5. Soedomo. (2001). *Pencemaran Udara.* Bandung: ITB Bandung.
6. Utami, B. (2011). *Polusi Udara.* Jakarta: Grasindo.
7. Winarno dan Deni. (2011). *Bikin Robot Itu Gampang*. Jakarta : PT Kawan Pustaka.

A Text Mining Application of Emotion Classifications of Twitter's Users Using Naïve Bayes Method

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***Abstract*— Twitter is one of social media with more than 500 million users and 400 million tweets per day. In any written tweet of twitter users it contains various emotions. Most research on the use of social media classifies sentiments into three categories that are positive, negative, and neutral. However, none of these studies has developed an application that can detect user emotions in the social media, particularly on Twitter. Hence, this research developed a text mining application to detect emotions of Twitter users that are classified into six emotions, namely happiness, sadness, anger, disgust, fear, and surprise. Three main phases of the text mining utilized in this application were preprocessing, processing, and validation. Activities conducted in the preprocessing phase were case folding, cleansing, stop-word removal, emoticons conversion, negation conversion, and tokenization to the training data and the test data based on the sentiment analysis that performed morphological analysis to build several models. In the processing phase, it performed weighting and classification using the Naïve Bayes algorithm on the validated model. The process for measuring the level of accuracy generated by the application using 10-fold cross validation was done in the validation phase. The findings showed that this application is able to achieve 83% accuracy for 105 tweets. In order to get a higher accuracy, one requires a better model in training data.**

Keywords—text mining; Twitter; emotion; classification; naïve bayes.

I. INTRODUCTION (*HEADING 1*)

Social networking websites, like Twitter and Facebook, create enormous opportunities for users to communicate with one another without having to worry about differences in moral and social values. They also enable mutual learning and sharing of valuable knowledge with no regard to geographical distance, time barrier, and language skills. People thus join and engage in various communities and discussion groups that best suit their needs.

Twitter has at least 500 million users and 400 million tweets posted in its site every day [1]. Tweets are written messages in the form of texts that contain opinions, expressions, and emotions of users. Data in Twitter's site is inherently unstructured because users do not care about spelling and grammatical construction of sentences when posting their tweets. With a large amount of user-generated data on Twitter every day, extracting logical emotional patterns of this data with accurate information from such unstructured form is considered as a critical task to perform.

Text mining can be used to overcome this problem as it provides computational intelligences of multidisciplinary disciplines like information retrieval, artificial intelligence, statistics, database systems, and others [2]. Application of text mining techniques on social networking sites can further reveal results related to human thinking patterns, group identification and recommendation, and also opinion about any specific topics of interests. In the previous studies, many researchers classified emotions into two classes that are 'positive and negative' or 'happy and unhappy'. Hence, this excludes user's other basic emotions like anger, fear, disgust and surprise [3]. Furthermore, it was noticeable that those existing techniques of emotion prediction in text just work on articles which contain only the direct emotional words and neutral words. In addition, the existing research papers do not particularly target the datasets in social networking sites and they hardly mention the pre-processing phase that is important to simplify the text mining process [4].

Based on the problems mentioned above, this study will try to develop an application that is able to extract data from Twitter and then classify the data into six categories of emotions, such as happiness, sadness, fear, anger, surprise, and disgust, using Naïve Bayes classification method [5]. Classification is "the process of learning a set of rules from a set of examples in a training set. Text classification is a mining method that classifies each text to a certain category." [2, p. 5]. This study is divided into three phases to extract information from datasets and transform it into an understandable structure for further use. The three phases are pre-processing phase, text mining phase, and results validation phase that help this study to conduct a thorough analysis on how to discover groups and structures in the data set.

II. RESEARCH QUESTION AND OBJECTIVES

A. *Research Question*

How to build a text mining application that can classify emotions of Twitter's users into six categories, such as happiness, sadness, fear, anger, surprise, and disgust, using Naïve Bayes classification?

B. *Research Objectives*

Objectives of this research are as follows:

1. To be able to extract data from Twitter that is large, unstructured, and dynamic.
2. To organize the collected data into pre-defined categories that will be used for text analysis by performing pre-processing process.
3. To construct suitable models based on the datasets by assessing how well the model fits, adjusting some models, and finally selecting the best model from among those that have been tried.
4. To present the classified data accordingly to the emotion categories in a useful format, such as a graph or table.

III. LITERATURE REVIEW

A. *Definition of Text Mining*

According to Berry and Kogan, text mining is a technique used to discover textual patterns from unstructured, big, and dynamic data [6]. Text mining focuses on categorizing texts, grouping texts, extracting concepts, and reducing texts. Text mining is an extension of data mining technique. Irfan *et al.* pointed out that data mining is a technique to extract logical patterns from structured database [2].

B. *Phases in Text Mining*

As an extension of the data mining technique, text mining uses phases in data mining to extract unstructured text that can be further used for finding opinion about certain subjects, patterns, and other purposes. The phases are as follows:

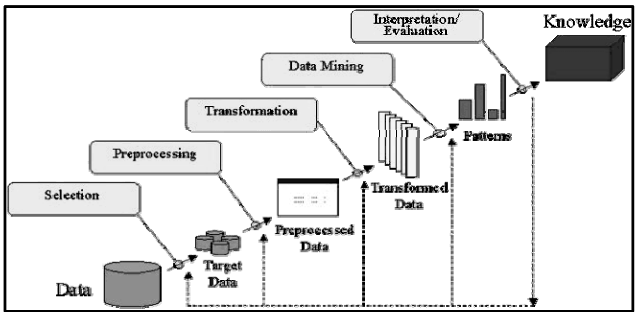

Figure 1. Phases in Data Mining [7]

- Data selection

  Data selection from the collection of operational data needs to be done before the stage of extracting information in Knowledge Discovery in Databases (KDD) begins. Data resulted from the selection in the data mining process, is stored in a file, separated from operational databases.

- Pre-processing/cleaning

  Before the process of data mining can be implemented, it is necessary to perform data cleansing that is the focus of KDD. Cleaning processes include checking inconsistent data, removing duplication of data, and correcting errors in the data like printing errors.

- Transformation

  Is the process of transformation on the data that has been selected to fit the process of data mining. The coding process in KDD is considered as a creative process and is highly dependent on the type or pattern of information to be searched in the database.

- Data mining

  Is the process to find patterns or information in the selected data using techniques or methods that are varied widely. Selection for suitable methods or algorithms to use is fully dependent on the purpose(s) and KDD processes as a whole.

- Interpretation/evaluation

  Information patterns resulted from the data mining process must be presented in the comprehensive form that can be understood by the involved parties. This phase will check whether or not the pattern or information found is conflicting with the previous facts or hypothesis.

C. *Classification*

Data mining algorithms are basically divided into supervised learning, unsupervised learning, and semi supervised learning. Classification is a common example of supervised learning approach whereby a given data set is typically divided into three parts that are training, validation, and testing data sets with known class labels. Classification is the process of providing an algorithm with records in which an output variable of interest is known and the algorithm "learns" how to predict this value with new records where the output is unknown.

Gundecha and Liu explained that "supervised algorithms build classification models from the training data and use the learned models for prediction. To evaluate a classification model's performance, the model is applied to the test data to obtain classification accuracy" [8, p. 2]. Several methods that can be used for supervised learning are decision tree induction, Naïve Bayes, Support Vector Machines (SVM), and k-nearest neighbors. In light of this, this particular research will use Naïve Bayes classification method to classify emotions of Twitter's users.

D. *Naïve Bayes Classification Method*

Naïve Bayes is one of the fastest and simplest Bayesian Learning methods. It is derived from Bayes theorem and the hypothesis of freedom, producing statistic classification based on opportunities. This is a simple technique and should be used before attempting a more complex method. Barry and Kogan explained that chances of a message d in class c, P cd, calculated as [6]:

$$P(c|d)\ P(c)\Pi P(tk|c)mk=1 \quad \text{(i)}$$

where (|) is the conditional probability of a feature that occurs in the classroom message c. P(c) is probability from the previous message happened in class c. P(tk|c) can be used to measure how much evidence the contribution that c is the correct class. In the email classification, classes from the messages are determined by finding maximum a posteriori (MAP) classes that are mostly defined by:

$$cmap=a\ gmaxc\ {*c}\ cs+\ P(c|d)=a\ gmaxc\ {*c}\ cs+\ P(c)\ \Pi P(tk|c)=1 \quad \text{(ii)}$$

Formula 2 involves multiplication many conditional probability, one for each feature, then the resulting calculations are in floating point underflow. In practice, multiplying the opportunities are often converted into an additional logarithmic probability and therefore, to maximize the equation can use the following alternative.

$$cmap=a\ gmaxc\ {*c}\ cs+,\ P(c)\ \Sigma\ P\ tk\ c=1 \quad \text{(iii)}$$

All parameters of the model, the probability distribution classes and features, can be estimated with relative frequency of training data D. For example, when the given class and messaging features do not occur together in the training data, the estimated probability based on the appropriate frequency will be zero, which would make the right side of Formula 3 is undefined. This problem can be overcome by incorporating some corrections as Laplace smoothing in all probability estimates, so the chances of each feature can be calculated by the following equation.

$$P(t|c)=\ T\ 1\Sigma\ T\ V=\ T\ .\Sigma T\ V \quad \text{(iv)}$$

where | | is the number of terms in vocabulary.

E. Emotions

Emotion is defined as a reaction to certain situations that can be seen from the face or body [9]. According to Ekman, there are six categories of emotions such as happiness, sadness, fear, anger, disgust, and surprise [5]. In extending this viewpoint, Robbs made a wheel of feelings as shown below.

Figure 2. Wheel of Felling [10]

This wheel of feelings depicts the six emotions that thus are formed in the center of the wheel and the outside is synonymous more complex (e.g. loving, remorseful, and alienated). This helps to narrow down the best word used to express the emotional state at this time.

F. Twitter

Twitter offers a form of micro-blogging social network that allows users to make updates of writing texts with a maximum length of 140 characters and read messages [11]. Twitter updates are known as tweets. Several features provided in Twitter's site that are following, followers, mentions, favorites, hash tag (#), direct message, list, and more.

Tweets can be seen by the public, but the sender can restrict delivery of messages to their friends list only. Users can see the tweets of other users, known as followers (followers).

IV. DESIGN AND IMPLEMENTATION

A. Requirements

The following will enlist the specification requirements for the text mining application of emotion classifications of Twitter's users.

1. Applications can extract data from Twitter online using the Twitter API (Application Program Interface).
2. Applications can search tweets by username or keywords entered in real-time using the Twitter API.
3. Applications can clear the data that has been extracted.
4. Applications can perform training of the data that has been extracted.
5. Applications may classify tweets or sentences into six emotional categories, namely happiness, sadness, fear, anger, surprise, and disgust.
6. Applications can validate and provide accuracy values obtained from the training data and testing data.

B. Design Modeling

This particular section will briefly explain the modeling of the text mining application by showing all the activities and datasets used in the application. Once the tweets data have been successfully extracted from the Twitter API, these data will go to the pre-processing phase where there are numerous activities performed including case folding, cleansing, stop-word removal, emoticon conversion, negation conversion, and tokenization. When the pre-processing is done, it later moves to the training part where one should use the training dataset to generate models based on the rules set on the data. The models are evaluated using the Naïve Bayes algorithm that results. After the best model is selected, it can automatically classify the emotions and thus display the results by showing the level of accuracy of classifying emotions that exist in the tweets. Of note here is that test data are used only at the end of the model building and selection process to assess how well the final model might perform on additional data.

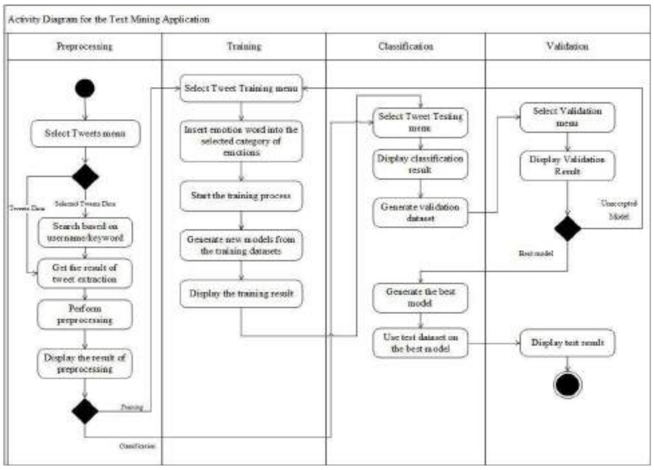

Figure 3. Activity Diagram for the Text Mining Application

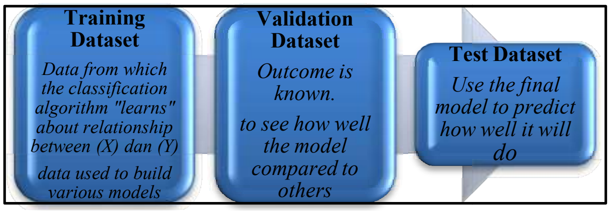

Figure 4. Datasets for the Text Mining Application

Note:
X = input variable (independent variable)
Y = output variable (dependent variable)

In the design process, the data model is divided into three datasets that are training dataset, validation dataset and test dataset. A series of rules are set on the training dataset. This dataset is obtained when entering into the application to determine the category of each tweet that is later weighted for each word in that particular tweet. Several models are emerged in this training process. Another dataset, also known as the validation dataset, is thus applied to these models. This is to compare which model works the best using the Naïve Bayes algorithm. Once the best model has been selected, the test dataset is used to predict how well it can classify the emotions contained in those tweets by showing the accuracy results.

C. Development Phases

Figure 5 shows steps in developing the text mining application.

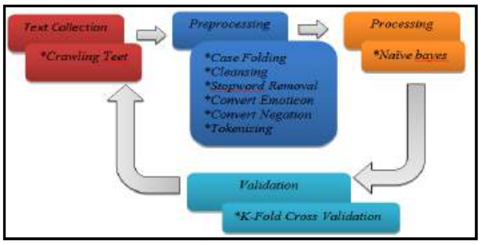

Figure 5 Development Phases for the Text Mining Application

1. Text Collection
   It is done using the streaming API and Twitter Search with additional filters based on username/keyword as shown in the figure below.

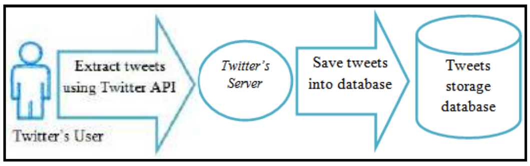

Figure 6. Process of Text Collection

2. Preprocessing:
   Once the entire data tweets are successfully retrieved, then one needs to separate the data into two datasets that are the training data and data testing. Furthermore, the second part of the data is then performed text preprocessing of the data. The stages of preprocessing, among others:
   1. Conversion to lowercase.
   2. Removing URL (e.g. http://bit.ly/mEibnV).
   3. Removing mention (e.g. xoxo).
   4. Delete a character other than a- z.
   5. Eliminate stop-word.
   6. Convert the symbols/emoticons into texts.
   7. Convert the negation and combine it with the word after negation.
   After the training data and testing are clean, the next step to do is to apply the Naïve Bayes classifier algorithm in the training process to build a model of the probability of training data.

3. Processing:
   Using Naïve Bayes algorithm to classify tweets in the application.
   Naive Bayes classification on every tweet represented in a pair of attributes $<a_1,a_2,a_3,\ldots,a_n>$ where a_1 is the first sentence, a_2 is the second sentence, and so on. V is the class set. At the time of classification, this method will produce a category/class of highest probability (V_{MAP}) by inserting attributes $<a_1,a_2, a_3,\ldots,a_n>$. Formula for V_{MAP}:

$$V_{MAP} = \underset{v_j \in V}{\operatorname{argmax}} P(v_j \mid a_1, a_2, a_3, \ldots, a_n) \qquad \text{(v)}$$

   By using the Bayes theorem, the equation (v) can be written as:

$$V_{MAP} = \underset{v_j \in V}{\operatorname{argmax}} \frac{P(a_1, a_2, a_3, \ldots, a_n \mid v_j) \times P(v_j)}{P(a_1, a_2, a_3, \ldots, a_n)}$$

(vi)

$P\ (a_1,a_2,\ a_3,\ldots,a_n)$ is a constant value for all, V_j, so that the equation (vi) can be expressed into the equation (vii).

$$V_{MAP} = \underset{v_j \in V}{\operatorname{argmax}} P(a_1,a_2,a_3,\ldots,a_n|v_j) \times P(v_j) \quad \text{(vii)}$$

Naive Bayes algorithm simplifies this equation by assuming that every attribute is conditionally independent of each other for each category. In other words,

$$P(a_1,a_2,a_3,\ldots,a_n|v_j) = \prod_i P(a_i|v_j) \quad \text{(viii)}$$

If the equation (vii) substitutes to the equation (viii), the result is:

$$V_{MAP} = \underset{v_j \in V}{\operatorname{argmax}} P(v_j) \times \prod_i P(a_i|v_j) \quad \text{(ix)}$$

P(v_j) and word probabilities for each category P($a_i|v_j$) counts in training process. Where,

$$P(v_j) = \frac{docs_j}{training}$$
$$P(a_i|v_j) = \frac{n_i + 1}{n + kosakata} \quad \text{(x)}$$

$$P(a_1,a_2,a_3,\ldots,a_n|v_j) = \prod_i P(a_i|v_j) \quad \text{(xi)}$$

Where $docs_j$ is the total of documents in j category and training is the total of documents used in the training process. n_i is the total of words appeared in v_j category, n is the total of glossaries existed in each category v_j and glossary is total of unique words in all of training data.

4. Validation: using 10-fold cross validation to determine the accuracy of classifying the emotions existed in the tweets generated by this application.

$$Accuracy = \frac{Total\ of\ True\ Classification}{Total\ of\ Testing\ Data}\ x\ 100\% \quad \text{(xii)}$$

D. Implementation of User Interfaces

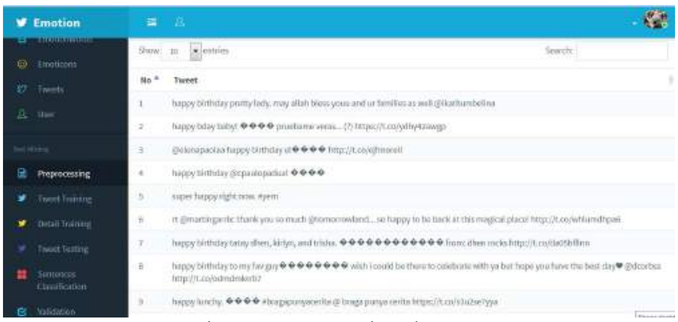

Figure 7. Preprocessing Phase

Figure 8. Processing Phase

Figure 9. Validation Phase

E. Implementation of Database

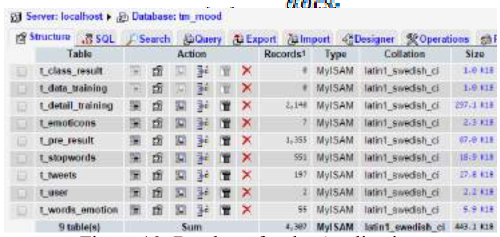

Figure 10. Database for the Application

V. TESTING

A series of testing performed on the application to measure the accuracy of classifying emotions of Twitter's users. The testing is divided into three parts as follows:

1. Using 10-fold cross validation

Table 1. Testing Performed Using 10-fold Cross Validation

Number of unique words	1.366 words
Training data	423 data
Test data	105 words
Accuracy	83.1034482759%

2. Based on the number of training data
   a. Number of unique words : 557 words
   b. Test data : 152 data

Table 2. Testing Based on Training Dataset

	Training Data	Accuracy
Test #1	116	71,3 %
Test #2	229	60,45 %
Test #3	280	62,04 %

3. Based on unique words and training dataset
   Number of dataset: 152 data

Table 3. Testing Based on Unique Words and Training Dataset

	Unique Words	Training Data	Accuracy
Test #1	1.164	280	63,7 %
Test #2	1.362	359	77,55 %
Test #3	1.495	404	77,14 %

The results of testing performed using 10-fold cross validation and training data showed that the ratio of the first and second test resulted in decreased levels of accuracy. Meanwhile, the second and third tests yield increased accuracy. Testing based on the number of unique words and training dataset revealed that the accuracy of classifying the user emotions is better as can be seen from the first, second, and third testing in Table 3. In summary, the number of unique words and training datasets have significant impacts on the level of accuracy achieved as it may increase or decrease the accuracy to certain levels.

VI. CONCLUSION AND RECOMMENDATION

A. Conclusion

1. This text mining application can successfully extract data from Twitter online using the Twitter API.
2. This application can classify emotion words and examples of emotion sentences that are suitable to be used to express the emotions of Twitter's users.
3. The application can classify emotions into 6 categories such as happiness, sadness, fear, anger, surprise, and disgust.
4. It has managed to build a model to classify tweets based on sentiment and categories using Naïve Bayes algorithm.
5. The test results showed that unique words and a larger training data will lead to a higher accuracy for the identification of emotions because it can provide a better and wider coverage of the emotional moments in our daily lives.

B. Recommendation

1. To conduct further research on this area of interest by developing this application using other classification methods such as K-Nearest Neighbor and Support Vector Machine.
2. Can add #hashtags as one of the components for the classification of emotions.
3. To automatically delete or omit a tweet that contains no emotion.
4. In the preprocessing process, one can add functionalities to detect and eliminate duplication of tweets (spams) as well as to reduce and eliminate the number of letters in a row.
5. The language used in the stop-word is not only English but could use Indonesian, local language or other foreign languages.
6. Add the Feature Extraction in particular for the Semantic Analysis and syntactical Analysis. Included in the syntactical Analysis is POS tagging and parsing, Semantic Analysis can detect concepts, events, and relationships between them.
7. Add the Feature Selection in particular for frequency-based feature selection, latent semantic indexing (LSD), and random mapping to eliminate irrelevant information and information which is repeated from the tweet.

References

[1] D. Farber, "Twitter Hits 400 Million Tweets Per Day, Mostly Mobile," 06th June 2012. [Online]. Available: www.cnet.com/news/twitter-hits-400-million-tweets-per-day-mostly-mobile/. [Accessed 10th September 2015].

[2] R. Irfan, C. K. King, D. Grages, S. Ewen, S. U. Khan, S. A. Madani, J. Kolodziej, L. Wang, D. Chen, A. R. Rayes, N. Tziritas, C. Z. Xu, A. Y. Zomaya, A. S. Alzahrani and H. X. Li, "A Survery on Text Mining in Social Networks," *The Knowledge Engineering Review,* pp. 1-24, 2004.

[3] T. Patil and S. Patil, "Automatic Generation of Emotions for Social Networking Websites using Text Mining," *Journal of IEEE Computer Society,* vol. 4, p. 31661, 2013.

[4] M. Adedoyin-Olowe, M. M. Gaber and F. Stahl, "A Survey of Data Mining Techniques for Social Media Analysis," 17th December 2013. [Online]. Available: http://arxiv.org/abs/1312.4617.

[5] P. Ekman, Emotions Revealed: Recognizing Faces and Feelings to Improve Communication and Emotional Life, New York: Times Books, 2003.

[6] M. W. Berry and J. Kogan, Text Mining Application and Theory, United Kingdom: Wiley, 2010.

[7] J. Han, M. Kamber and J. Pei, Data Mining: Concept and Techniques, Waltham: Morgan Kaufmann Publishers, 2012.

[8] P. Gundecha and H. Liu, "Mining Social Media: A Brief Introduction," INFORMS, Catonsville, USA, 2012.

[9] S. W. Sarwano, Pengantar Psikologi Umum, Jakarta: PT. Raja Grafindo Persada, 2010.

[10] "Dramatic Languages," 2014. [Online]. Available: https://robbsdramaticlanguages.wordpress.com/2014/07/31/. [Accessed 10th August 2015].

[11] D. Juju and F. Sulianta, Broading Promotion with Social Network, Jakarta: PT. Elex Media Komputindo, 2010.

[12] M. Yassine and H. Hajj, "A Framework for Emotion Mining from Text in Online Social Networks," *Journal of IEEE Computer Society,* vol. 10, pp. 1136-1142, 2010.

Development of Decision Support System for Manado's BAPERJAKAT using DAD and AHP

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***Abstract*—Decision support system is one of the systems that utilize computer technology which can facilitate proses of making a decision. In this case, the system will be used for advisory board positions and ranks of Manado city government or for short BAPERJAKAT to assess employee suitability to be promoted into structural position. Using decision support system method analytical hierarchy process (AHP), the system will consider and judge employee thus appoint them into suitable structural position. The system will be develop using software engineering method of Disciplined Agile Delivery (DAD) and it is expected to assist, minimize error and facilitate government authority to consider employee rank and position.**

Keywords*—*analytical hierarchy process; disciplined agile delivery; decision support system;

I. INTRODUCTION

As part of Manado city government contained Advisory Board Position and Rank or Baperjakat which is headed by Manado's city secretary. It is authorized to guarantee the quality and objectivity of the appointment civil servants (PNS) in and out of structural position start from echelon II position and down. In performing its functions, Baperjakat should cooperate with the Employment and Trainning Agency (BKD) as the agency in charge of implementing the planning, organizing, coaching, controlling and making policy of employment. Moreover, structural position is a position that indicates the duties, responsibilities, authority and rights of a civil servant in order to lead an organizational unit of the state.

Problems that occur at this time is Baperjakat, which have tasked to provide suggestions and advices to the Mayor of Manado on the prospective employee who will be lifted in and out of structural position, still use manual form made by BKD to be filled. Baperjakat then manually analyze the form thus it is susceptible to error in judgment and the absence of valid method in carrying out its assessment could lead to partiality. Given that echelon II position in Manado's government agency is a very important position then there are certain criteria or requirements that must be owned by the candidate for the post.

As the solution to the problems, authors plan to create a computerized decision support system using one of its methods which is Analytical Hierarchy Process (AHP) in order to process more objective assessment of candidates and the position will fall on the right and appropriate employee. This system will be developed using software engineering methodology of Disciplined Agile Delivery (DAD) and built using Java programming language.

The goal is to design a system for Baperjakat in Manado city government and aim to help Baperjakat team in conducting an assessment of employee that will be appointed in structural position of echelon IIb and produce a system that will rank the employee from the assessment result of employee who have the highest score to the lowest.

II. THEORITICAL FOUNDATION

A. Decision Support System

Decision Support System (DSS) is a system capable of providing problem-solving ability and support business and organizational decision-making activities. This system is used to assist decision-making in situation of semi-structured and unstructured situations, where no one knows for sure how the decision should be made [5]. DSS aims to provide information, guide, provide predictions and lead to user information in order to make better decision. Sprague and Watson [4] defined DSS as the system that has five main characteristics, which is (1) computer based system, (2) used to assist decision-maker, (3) to solve the complex problems that is impossible to do with manual calculations, (4) by a way of interactive simulation, and (5) data and analytical models as the main component.

B. Analytical Hierarchy Process

Analytical Hierarchy Process (AHP) is one of decision support model developed by Thomas L. Saaty. According to Saaty[3] the decision support model will elaborate multi-factor problem of a complex multi-criteria hierarchy. Hierarchy is defined as a representation of a complex problem in a multi-level structure where the first level is a goal, followed by level factors, criteria, sub-criteria, and so on down to the last level of alternatives. By hierarchy, a complex problem can be decomposed into groups and then organized into hierarchical form thus the problem would appear more structured and systematic.

In solving problems with AHP there are some basic principles that must be understood, namely:

- Establish hierarchy. Issues that needed to be resolved are broken down into its elements, which consist of criteria and alternatives, then organized into hierarchical

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structure. In order to obtain accurate results, the decompositions of elements made up cannot be parsed again.

- Assessment of criteria and alternative. Create judgment about the relative importance of two elements at a certain level with regard to the level above. Criteria and alternatives assessed through paired comparison. According to Saaty[3], for a variety of problems, the scale of 1 to 9 is the best scale to express opinions.

- Prioritization. For each criteria and alternatives, there should be a paired comparison. Pairwise comparison matrix determined from the weight vector, to obtain local priorities. Then the global priorities determined by the synthesis between local priorities. The value of comparison is then processed to conclude the relative rankings of all alternatives.

- Consistency measurement. Not all criteria assessment is consistent. These inconsistencies can be caused by an error at the time of assessment or due to lack of information and lack of consideration. In making a decision it need to know how much consistency there, so that the resulting decision are based on considerations with good consistency.

According to Kusrini [2], the AHP procedures are consist of (1) defining the problem and determined the desired solution then draw up a hierarchy of problems faced. Preparation of the hierarchy is to set goals which is the target system as a whole at top level; (2) determining the priority elements. The first step in setting priorities is to make paired comparison of elements and comparing elements in pair according to the criteria given. Then pairwise matrix is filled using numbers to represent the relative importance of one element against other elements; (3) synthesis. Considerations for pairwise comparison were synthesized to obtain overall priorities. Things to do in this step are add up the value of each column in the matrix, divide each column with the total value of the column in question to obtain a normalization matrix, and add up the values of each line and dividing by the number of elements to obtain an average value; (4) measuring consistency. In decision-making, it is important to know how well the consistency exists because we do not want decisions based on considerations with a low consistency. Things to do in this step are first, multiply each value in the first column with the relative priority of the first element, the value in the second column with the relative priority of the second element and so on. Second, add up each line. Lastly, the result of summation lines divided by relative priority of element in question; (5) Calculate consistency index (CI) with formula CI = (λ Maks-n)/n, where n = number of elements; (6) Calculate consistency ratio (CR) using the formula of CR = CI/IR, where CR = consistency ratio, CI = consistency index and IR = random consistency index. The random consistency index is shown in table 1 bellow.

TABLE I. RANDOM CONSISTENCY INDEX

Uk. Matriks	IR
1,2	0.00
3	0.58
4	0.90
5	1.12
6	1.24
7	1.32
8	1.41
9	1.45
10	1.49
11	1.51
12	1.48
13	1.56
14	1.57
15	1.59

(7) Check the consistency of hierarchy. If the value is more than 0.1, then the assessment data must be corrected. But if the consistency ratio less than or equal to 0.1, then the calculation results can be expressed properly.

C. Employee Affair

Civil servant is every citizen of Indonesia which has been determined eligible, appointed by the competent authority and entrusted with the task in public position, or entrusted with other country related task and paid based on legislation in force[6]. Structural position is a position that indicates the duties, responsibilities, authority and rights of a civil servant in order to lead an organization unit of the state. Table 2 show echelon structural position and rank levels in accordance with the decree of Kepala Badan Kepegawaian Negara No. 13 of 2002.

TABLE II. ECHELON STRUCTURAL POSITION AND RANK LEVELS

NO	ESELON	JENJANG PANGKAT, GOLONGAN/RUANG			
		TERENDAH		TERTINGGI	
		PANGKAT	GOL/ RUANG	PANGKAT	GOL/ RUANG
1	I a	Pembina Utama Madya	IV/d	Pembina Utama	IV/e
2	I b	Pembina Utama Muda	IV/c	Pembina Utama	IV/e
3	II a	Pembina Utama Muda	IV/c	Pembina Utama Madya	IV/d
4	II b	Pembina Tingkat I	IV/b	Pembina Utama Muda	IV/c
5	III a	Pembina	IV/a	Pembina Tingkat I	IV/b
6	III b	Penata Tingkat I	III/d	Pembina	IV/a
7	IV a	Penata	III/c	Penata Tingkat I	III/d
8	IV b	Penata Muda Tingkat I	III/b	Penata	III/c
9	Va	Penata Muda	III/a	Penata Muda Tingkat I	III/b

D. Advisory Board Positions and Rank (Baperjakat)

To ensure the quality and objectivity in the appointment of civil servant from the structural position of echelon II and below then formed advisory board position and ranks that consists in capital city, regional and district/ town. Baperjakat has the task of providing advice to the trustees personal officer in appointment, transfer, dismissal from position in echelon II and below. Baperjakat also consider promotion to the structural positions, showed work performance or discover new discoveries that will benefit the country. Moreover, baperjakat has the task of extent the retirement age for civil servant occupying the post of echelon I and II and appoint the secretary of the provincial/district/city.

III. System Design

The design phase of the system is intended as preparation for designing the system, sketching how the system is set up and provide a clear picture to developers to build the system. To design Baperjakat desicision support system, will use the methodology of Disciplined Agile Delivery (DAD) where consist of three phases of development[1].

A. Inception

This phase involves the analysis of business processes and process of mapping the sub processes up to the level of activities. Figure 1 show hiring procedure of employee for structural position and candidates' searches conducted by BKD from 2 sources which are employee database and candidates that being put forward for consideration from SKPD. Once there are several candidates then BKD will collect the data raging from technical and administrative requirements and team of Baperjakat will assess the candidate which later will be proposed to the Mayor of Manado.

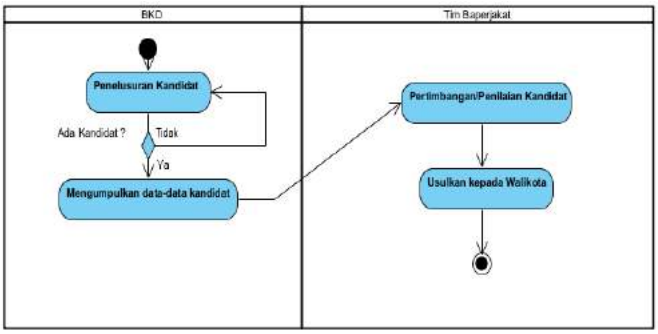


Fig 1. Manual business process

The system developed using similar process as shown in figure 2 with some improvement from the manual process. After using the system, assessment process of candidate for the position will become easier and the results are more objective and accurate.

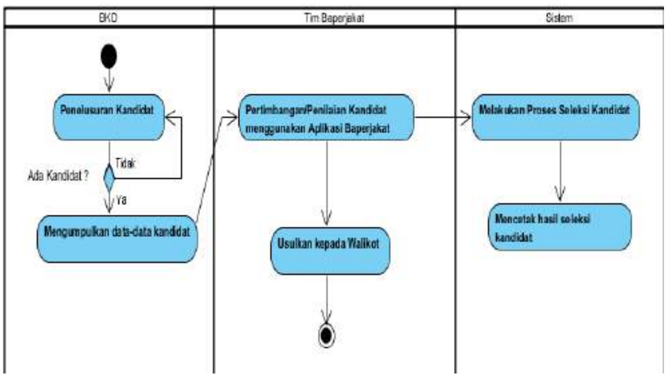


Fig 2. Application business process

B. Construction

The next stage is to identify the system architecture to be implemented as well as the model system to be designed. This phase will produced several documents such as software requirement specifications (SRS) and software architecture document (SAD).

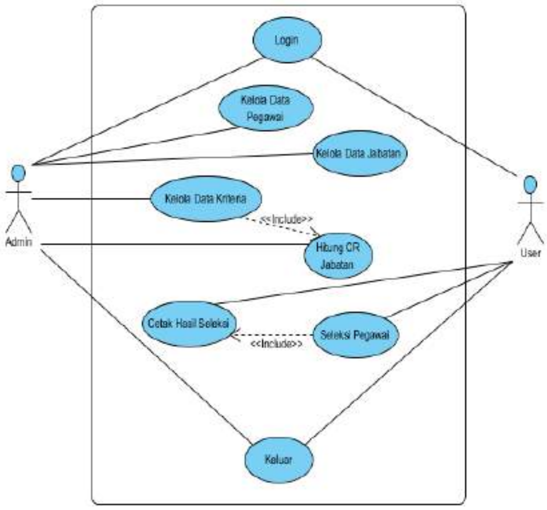


Fig 3. Use Case Diagram

Figure 3 shown 2 actors of the system that are admin and user. Admin are employees of BKD that have the privilege to login and manage employee data, position and criteria. Meanwhile, user is Baperjakat team member that will be able to perform calculation process of consistency ratio based on the AHP model and selecting employee who will occupy the structural position.

C. Transition

Figure 4 show the main page of the Baperjakat application. There are several options to run the application according to its functions. On the right is the homepage that there is an option to manage employee data and data positions. Then, on the left there is a choice such as the criteria menu, assessment menu, the user settings menu and exit menu. There is also time function that can show when user open the application and calendar to indicate the date when user uses this application.

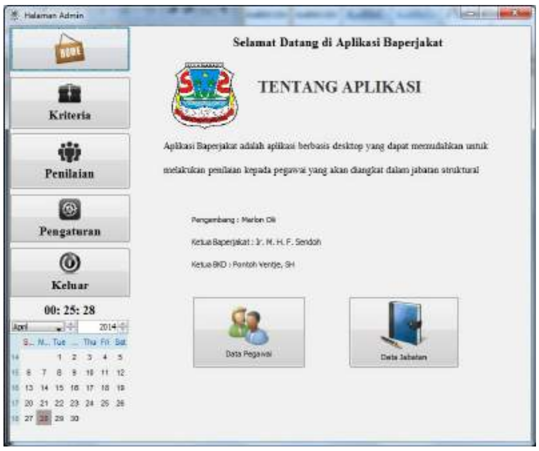


Fig 4. Application homepage

IV. Result and Discussion

The AHP models is used to assess consistency with the assessment procedure following the guidelines from the Kusrini book[2]. Where consistency will be accepted if the result of the calculation is worth ≤ 0.1 and not accepted if the consistency is > 0.1 accordance with the terms specified in the calculation method of AHP. Weights used are the value of AHP process. Next are the criteria of the post of head of Personnel and Training Manado city. (1) Group which has the value of 7; (2) list of sequence rank (DUK) which has the value of 5; (3) position competency which has the value of 3; (4) diklat pim which has the value of 2; and (5) level of education which has the value of 2. Moreover, steps that are used to calculate the consistency ratio are:

- Creating a pairwise comparison matrix as shown in table 3. At this stage, the comparison between the assessment criteria with the other criteria

TABLE III. Pairwise Comparison Matrix

Kriteria	Golongan	DUK	Kompetensi	Diklatpim	Pendidikan
Golongan	1	1.4	2.33333333	3.5	3.5
DUK	0.71428573	1	1.66666666	2.5	2.5
Kompetensi	0.42857143	0.6	1	1.5	1.5
Diklatpim	0.2857143	0.4	0.66666667	1	1
Pendidikan	0.2857143	0.4	0.66666667	1	1
Jumlah	**2.71428576**	**3.8**	**6.33333333**	**9.5**	**9.5**

- Create matrix of criteria values. The matrix is obtained using the formula column row new value = old value of the column row / column number each time. The result can be seen in table 4.

TABLE IV. Matrix of Criteria Values

Kriteria	Golongan	DUK	Kompetensi	Diklatpim	Pendidikan	Jumlah	Prioritas
Golongan	0.36842105	0.36842105	0.36842105	0.36842105	0.36842105	1.84210525	**0.36842105**
DUK	0.2631579	0.2631579	0.2631579	0.2631579	0.2631579	1.3157895	**0.2631579**
Kompetensi	0.15789473	0.15789473	0.15789473	0.15789473	0.15789473	0.78947365	**0.15789473**
Diklatpim	0.10526316	0.10526316	0.10526316	0.10526316	0.10526316	0.5263158	**0.10526316**
Pendidikan	0.10526316	0.10526316	0.10526316	0.10526316	0.10526316	0.5263158	**0.10526316**

- Create matrix of the sum of each row. The matrix is made by multiplying the value of priority in table 3 with table 4. The calculation result can be seen in table 5.

TABLE V. Summation Matrix of Each Row

Kriteria	Golongan	DUK	Kompetensi	Diklatpim	Pendidikan	Jumlah
Golongan	0.36842105	0.51578947	0.85964911	1.28947367	1.28947367	**4.32280697**
DUK	0.18796993	0.2631579	0.4385965	0.65789475	0.65789475	**2.20551383**
Kompetensi	0.06766917	0.09473684	0.15789473	0.23684209	0.23684209	**0.79398492**
Diklatpim	0.03007519	0.04210526	0.07017544	0.10526316	0.10526316	**0.35288221**
Pendidikan	0.03007519	0.04210526	0.07017544	0.10526316	0.10526316	**0.35288221**

- Consistency ratio calculation. To calculate the consistency ratio made into a table as shown in table 6.

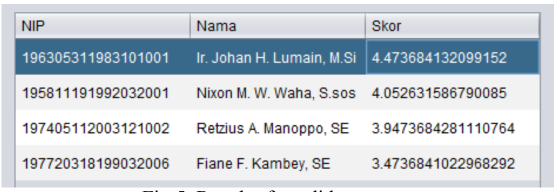
TABLE VI. Consistency Ratio Calculation

	Jumlah per Baris	Prioritas	Hasil
Golongan	**4.32280697**	**0.36842105**	**4.69122802**
DUK	**2.20551383**	**0.2631579**	**2.46867173**
Kompetensi Jabatan	**0.79398492**	**0.15789473**	**0.95187965**
Diklatpim	**0.35288221**	**0.10526316**	**0.45814537**
Tingkat Pendidikan	**0.35288221**	**0.10526316**	**0.45814537**
Jumlah			**9.02807014**

From table 6 obtained values as follows:

- Sum of all values = 9.02807014
- Number of criteria (N) = 5
- λ max (sum of all values/N) = 1.805614028
- CI ((λ Max-N)/N) = - 0.6388771944
- CR (CI/IR) = -0.6388771944 / 1.12 = - 0.570426

Because of CR < 0.1, then the consistency ratio for the position of Head of Training and Section is acceptable.



NIP	Nama	Skor
196305311983101001	Ir. Johan H. Lumain, M.Si	4.473684132099152
195811191992032001	Nixon M. W. Waha, S.sos	4.052631586790085
197405112003121002	Retzius A. Manoppo, SE	3.9473684281110764
197720318199032006	Fiane F. Kambey, SE	3.4736841022968292

Fig 5. Result of candidate assessment

Figure 5 show the final result of candidate assessment who had been calculated by the AHP process.

V. Conclusion

Using the software engineering methodology Disciplined Agile Delivery (DAD) and decision support system model of Analytical Hierarchy Process (AHP), a Baperjakat application has been developed that can be used to facilitate or assist decision-making process of selecting suitable candidate to occupy structural position in Manado city government offices. The application generates employee assessment in accordance with the order of values obtained after implementing the AHP model raging from highest to lowest.

REFERENCES

[1] Ambler, Lines, 2012, Disciplined Agile Delivery: A Practitioner's Guide to Agile Software Delivery in the Enterprise, IBM Press.

[2] Kusrini, 2007, Konsep dan Aplikasi Sistem Pendukung Keputusan. Penerbit Andi, Yogyakarta.

[3] L. Saaty Thomas, 1993, The 1993 Thomas L. Saaty and Jacob Wolfowitz Prizes, American Journal of Mathematical and Management Sciences Volume 14, Issue 1-2, 1994.

[4] Sparague, R. H. and Watson H. J., 1993, Decision Support Systems: Putting Theory Into Practice, Englewood Clifts, N. J., Prentice Hall.

[5] Turban, Efraim & Aronson, Jay E., 2001, Decision Support Systems and Intelligent Systems, 6th Edition, Prentice Hall: Upper Saddle River, NJ.

[6] Tim Fokus Media, 2012, Pokok-pokok Kepegawaian – Pegawai Negeri Sipil (PNS), Fokus Media, Jakarta.

A Study of Square Loop Resonator Filter at 2350 MHz for Nanosatellite Application

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***Abstract*—Tel-USAT payload communication block consists of S-Band transmitter, narrow band pass filter, and low dimension antenna in simplex mode. Filter selects the signal at certain frequencies to minimize the interference from adjacent frequencies. For nanosatellite application, filter has some requirements: small dimension, space proven material, and narrow bandwidth. One of methods that meets the requirements is a loop resonator. In designation of the loop resonator, there are some modifications such as chamfer in the corner of resonator, gap to avoid the loosely coupled and perturbation. The filter in the dimension of 40mm × 40mm × 1.67mm with FR4 dielectric material has obtained 44dB return loss or 1% reflected power back to input port, 6dB insertion loss, and phase shift 180 degree in linear characteristic.**

***Index Terms*—Nanosatellite; Filter; Resonator**

I. Introduction

Telkom University is conducting the research and development of nano-sized satellite (nanosatellite) named Tel-USAT. In the planned orbit, Tel-USAT receive command from ground station by using Telemetry, Tracking, and Command (TTC) module at frequency 437.43 MHz for sending satellite sensor data in half duplex mode and camera data which the data has been encrypted into stream bit at frequency 2350MHz. Filter designed in S-Band frequency to minimize the interference from adjascent frequencies, and reduce the bandwidth generated by S-Band transmitter and high power amplifier. The 1200 bps and FSK modulation is used for the data transmission process. Microwave filter theory and its applications grew in the years preceding World War II, by pioneers such as Mason, Sykes, Darlington, Fano, Lawson, and Richards for low-frequency filters in radio and telephony [1]. The microstrip ring resonator was first proposed by P. Troughton in 1969 for the measurements of the phase velocity and dispersive characteristics of a microstrip line [2]. Since 2011, loop resonator introduce for GSM and WCDMA dual mode filter application in center frequency 1800 MHz and 1900 MHz. using Rogers R03006 dielectric material with relative permittivity 6.15 [3]. Another loop resonator research is WIMAX in frequency 2.3 GHz. It is using RT/Duroid 6010 with relative permittivity 10.2 produces fractional bandwidth 5.1% [4]. This paper propose the loop resonator filter design for nanosatellite camera data transmission by using FR4 dielectric material with relative permittivity 4.3 and loss tangent 0.025. The filter obtained 2% fractional bandwidth with single narrowband characteristic related to the operating bandwidth and low dimension constraint. Section I explains the filter implementation in nanosatellite communication block, the state of art and project explanation. In section II, a brief description of loop resonator in resonance frequencies and filter geometry. In section III, we analyze the design and measured results of loop resonator filter in S-Parameter and S-Parameter phase. Finally, our conclusions are commenced in section IV.

II. THEORY OVERVIEW

The selection of frequency due to low rain attenuation and atmosphere attenuation in the range of S-Band, as well as Tel-USAT cooperate with the Indonesian Amateur Radio Organization (ORARI) in terms of amateur frequencies ranges 2300 MHz - 2450 MHz [5]. Material selection with a loosy dielectric substrate occur in nanosatellite implementation, when nanosatellite has a constant pressure of 1 Atmosphere. In 1 Atmosphere pressure liquid or lumped circuit will explode, so the solid materials such as FR4 dielectric fit to the condition. Following the design of the narrowband elliptic filter at center frequency 2350MHz.

A. Stripline

Microstrip filter consist of 3 layers, stripline using cooper as conductor, FR4 lossy dielectric material that separates stripline and other cooper plate as the ground plane. For loop resonator model, the conductor width W_eff (f) is the function of center frequency and the space between stripline and groundplane seperated by dielectric material which has effective permittivity value _eff.The characteristic impedance is 50 Ohm, equation 1 describes the effective stripline dimension [2].

$$W = L = \frac{h\eta_0}{z_0\sqrt{\epsilon_{eff}0}} \tag{1}$$

The dimension of microstrip and stripline is inversely proportional to the meterial relative permittivity. Microstrip dimension for loop resonator design with 2% fractional bandwidth is 40mm × 40mm × 1.67mm. Figure 2 shows one possible circuit arrangement. The resonator length is equal to the resonant frequency of filter and material relative permittivity [6].

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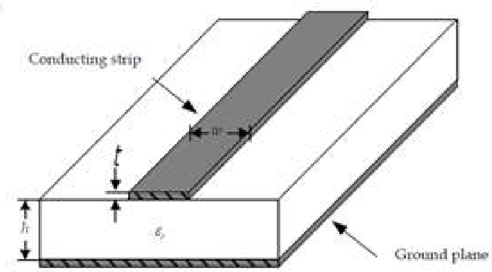

Fig. 1. Microstrip [6]

$$\lambda_g = \frac{c}{f\sqrt{\epsilon r}} \quad (2)$$

$$\frac{w}{d} = \mu = \frac{8e^{A}}{e^{2A} - 2} \quad (3)$$

$$l = \frac{\left(\frac{\pi}{180^{\circ}}\right)}{\sqrt{\epsilon e k}} w \quad (4)$$

The input port of filter connected to S-Band transmitter and output port to antenna. Both of port impedance is 50 Ohm. From equation (3) and (4) feedline length is 6 mm and feedline width is 1.2 mm.

B. Bending

A curved microstrip line can be modeled as a cascade of sections of microstrip lines with chamfered bends. Illustrated in Figure 3a is a typical bend in a microstrip line for an arbitrary bend angle also shown in the same gure are the reference planes that dene the edges of the bend. The equivalent circuit of the bend, in the region restricted to the connes of the reference planes, is shown in Figure 3b [2]. Figure 3 shows the equivalent-circuit representation of the bend in the corner of resonator, the inductance L represent the association of discontinuity. For optimum chamfer, the ration of the width of the chamfered region to the width of the microstrip line is approximately 0.5 [2]. Figure 4 explain the effect of the various chamfer side from 0 mm to 8mm toward reflected power.

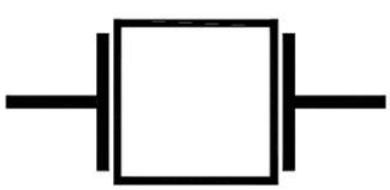

Fig. 2. Loop resonator [2]

The 1.3 mm chamfer side and 1 mm resonator width generates the optimum magnitude response with return loss ¡ 27dB. Inductance affected by resonator length and characteristic impedance [7].

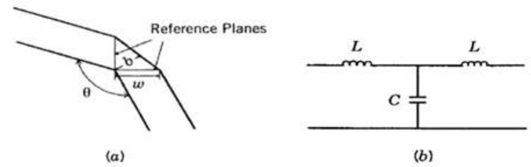

Fig. 3. Bending curve (a) Bending angle (b) Equivalent circuit

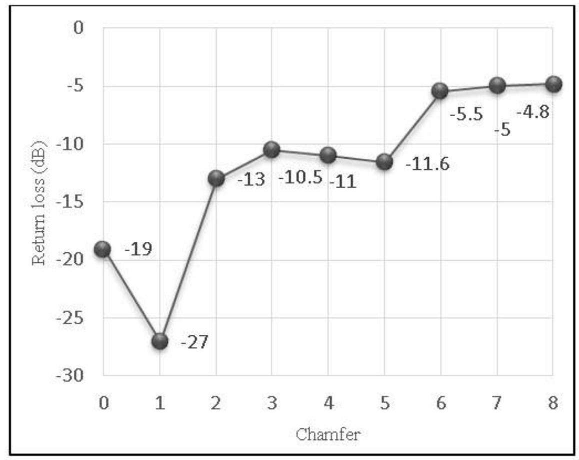

Fig. 4. Chamfer and Return Loss

$$L = 0.0847 Z_0 l \, nH \quad (5)$$

The chamfer side inversely proportional to the inductance effect that generates higher reflected power.

C. Perturbation

Perturbing stubs commonly used to excite two splitting resonant frequencies or more. Perturbation can build up a local resonance and maintain the continuity of the standing-wave pattern inside the perturbed region [2]. Figure 5 shows the various perturbation dimension from 0 mm to 8 mm affect the resonant frequency generated by loop resonator.

The perturbation in resonator region gives the advantage of line width flexibility. In the filter design, perturbation is the sensitive modification that can excite the natural characteristic of loop resonator which commonly used to dual band mode. The optimum perturbation dimension in increasing the resonant frequency without frequency splitting is 2mm. When the perturbating stub located in zero voltage, which is a short circuit and the linear position to the annular angle, the resonant modes will not split until the dimension of perturbation is equal to the line width. The larger splitting range can be obtained by increasing the perturbation width.

D. Resonator Gap

Coupling gaps separate the feed line and the resonator. Coupling gaps represent the capacitance effect which affected by the relative permittivity of dielectric material (ϵ_r), the surface area (A), and the distance between the resonator and

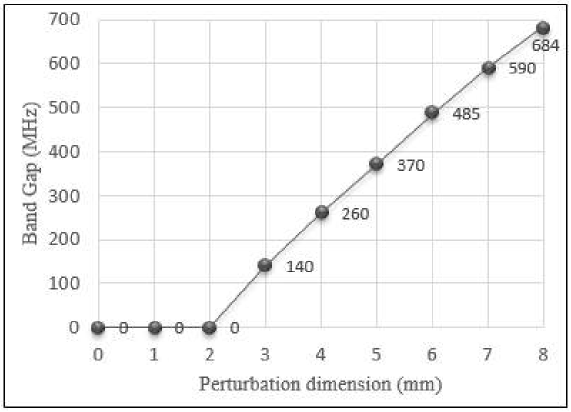

Fig. 5. Perturbation Dimension and Dual Band Gap

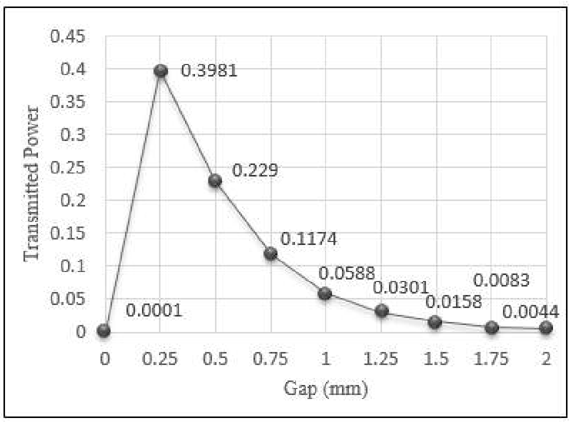

Fig. 6. Feedline Gap and Transmitted Power

feedline (d). The value of capacitance (farads/meter) between parallel plates obtained according to the equation 7 [7].

$$C = \frac{0.2249\epsilon_r A}{d\epsilon_0}\, pf \tag{6}$$

If the distance between the feedlines and the resonator is large, the the coupling gaps do not affect the resonant frequencies of the ring. This type of coupling is referred to the literature as loose coupling. Loose coupling is a manifestation of the negligibly small capacitance of the coupling gap as shown is the equation 7. [2] Figure 6 shows the various coupling gaps distance that affect the power transfer from the feedline. The larger distance generates loose coupling effect, which the optimum distance for loop resonator with FR4 dielectric material is 0.25mm.

III. RESULT AND ANALYSIS

As well as [3,4,6,7] the modification of conventional square loop resonator model shows in Figure 2 is the additional of chamfer, perturbation, and gap. Chamfer is typical bend in a microstrip line for an arbitrary bend angle θ approximately 45 degree in the edge of resonator. Perturbation as shows in Figure 3 is the additional resonannce at center frequency, coupling gap optimization to avoid the loosely coupled because of

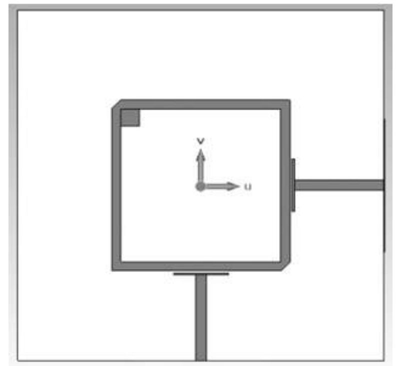

Fig. 7. Filter geometry

TABLE I
FILTER DIMENSION

Parameter	Dimension
Patch Width	40 mm
Substrate Thickness	1.6 mm
Patch Thickness	0.035 mm
Perturbation	2 mm
Feed Gap	0.25 mm
Feedline Length	6mm
Length of Square Resonator	17.47 mm
Feedline Width 1	1 mm
Feedline Width 2	1.2 mm
Resonator Width	1 mm

the distance between resonator and feedline is large [4]. The final design is shown in Figure 7, while the fabricated filter is shown in Figure 8. Based on the geometry showed in Tabel 1, filter generates magnitude response shows in figure 9. Return loss value in simulation result is 45 dB. It means the reflected signal back to port can be neglected, as numerical reflection signal is 0.01.

From input power 1, it will occur the optimum power transfer. The insertion loss in the simulation is -4.3 dB, as the numerical transmitted signal is 0.4 from 1 input power. The input power is not totally transmitted by filter because of the dielectric loss and conductor attenuation. The higher tangential loss value, the dielectric attenuation will increase, so it affects the transmitted power. In the realization process, filter generates the return loss value approximately -44 dB, as well as the simulation result. The insertion loss value in the realized filter is -6 dB, it is equal to 25% transmitted power in the realization result. Dielectric material affects the attenuation constant that generated by filter. Another important component of propagated waves in dielectric material is the magnitude phase response. Figure 10 explains the return loss phase and insertion loss phase. Based on the simulation result, the insertion loss phase is equal to -30 degree. This value is acceptable along the center frequency and bandwidth occupied phase is linear. Return loss phase in frequency 2350 MHz is 180 degree. It means there is phase reversal in input port

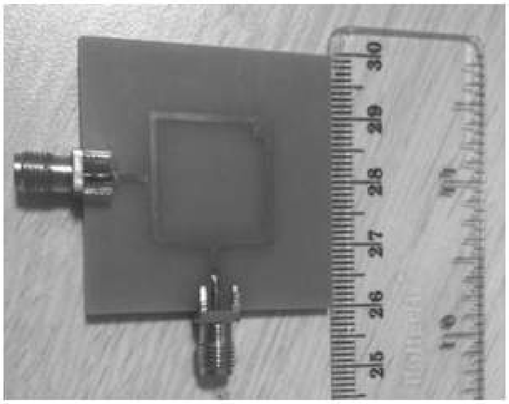

Fig. 8. Filter realization

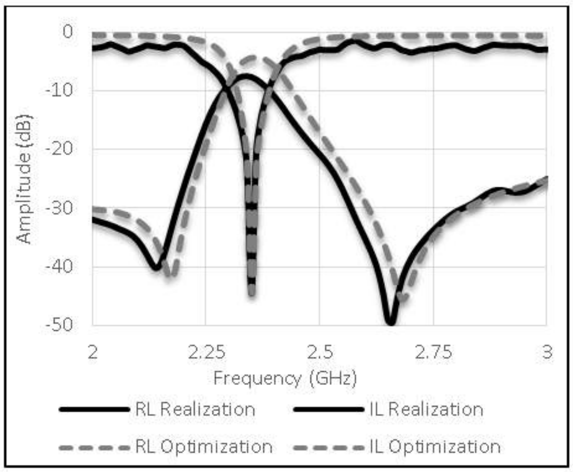

Fig. 9. S-Parameter

toward signal source. This phase can be neglected if the return loss magnitude is small. Figure 10 shows the insertion loss phase of the realized filter is 130 degree. The realization insertion loss phase is linear along the center frequency. In realization result, there is group delay response in loop resonator. The equal ripple from elliptic response generate high group delay response. The realization return loss phase is -180 degree. It is reflect the signal to the source.

TABLE II
RESULTS COMPARATIVE

Parameter	Simulation	Realization
Center Frequency	2350 MHz	2350 MHz
Bandwidth	41.7 MHz	50 MHz
Return loss	-45 dB	-44 dB
Insertion loss	-4.3 dB	-6
Return loss phase	180 degree	180 degree
Insertion loss phase	-30 degree	130 degree
Impedance	48.4 Ohm	49 Ohm
VSWR 1	1.01	1.01

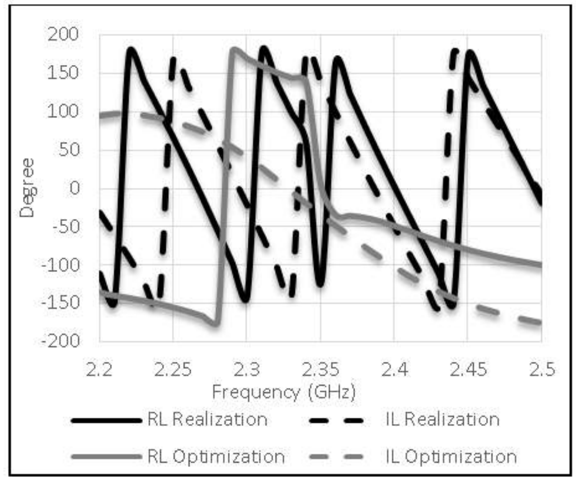

Fig. 10. S-Parameter Phase

IV. CONCLUSION

The designed filter with loop resonator works in frequency 2350MHz, and it obtained 44dB return loss and 6dB insertion loss. Some of the input power transmitted by loop resonator while other input power transform into heat and absrorbed by loosy dielectric material and reflected to input port. The insertion loss phase is linear in center frequency to avoid the changing of information, there is phase group delay in realization in which the realization phase is equal to 130 degree, while there is phase shift in reflected signal.

REFERENCES

[1] Pozar, David M. Microwave Engineering. John Wiley & Sons. Inc. Third Edition.
[2] Chang, Kai. Lung-hwa hsieh. Microwave Ring Circuits and Related Structure. A John Wiley & Sons. Inc. Second edition
[3] Chen, Chang. Chen Weidong. Zhang Zhongxiang. A Novel Dual-mode Dual-band Bandpass Filter with DGS. PIERS Proceedings. Marrakesh. Morocco. March 20-23. 2011
[4] Esfeh, Babak Kazemi. Ismail Alyani. Abdullah Raja Syamsul Azmir Raja. Narrowband Elliptic Bandpass Filter Using Dual-Mode Microstrip Square Loop resonator for WiMax Application. 2009
[5] Organisasi Amatir Radio Indonesia. Pembagian Dan Penggunaan Segmen Band Frekuensi Amatir Radio (Bandplan). 2009
[6] Konpang, Jessada. A Dual-Mode Wide-Band Bandpass Filter Using the Microstrip Loop resonator with Tuning Stubs.
[7] Maththael, George L. Leo Young, E. M. T Jones. Microwave Filters, Impedance-Matching Network, and Coupling Structure. Artech House

Competitive Intelligence through e-Tourism Development in Manado City

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Internet had widely influence in this time being. The world that had not connected with internet will be consider as isolated world from "international community". Internet had already make the world become "global village". The growth of Manado City recently had become one of the most developed cities in Indonesia. The Manado City had the objective to become one of the known city in Tourism. It need a global publication to help introducing Manado over the world. Here Internet play the main role as internet can promote Manado City to global world, become the world destination in tourism and this is the concept of e-Tourism. With e-Tourism, SWOT analysis can be obtained to help Manado City developing the competitiveness specially in tourism. With the Strength, Weakness, Opportunity, and Threats can fulfill the recommendation to develop the Manado City e-Tourism globally.

Keywords: Competitive Intelligence, e-Tourism, Manado City Tourism

I. INTRODUCTION

Global changes that occur in this information age, present unavoidable competition. The Use of Competitive Intelligence plays an important role in this competition, because the existing competition, especially in the field of tourism, can help the city of Manado to the level of the World Tourism City. Media promotion of tourism through the Internet can be a gateway of Manado City to be widely known as the City of the World Tourism. The ultimate goal is to develop tourism in the city of Manado via e-tourism by using Competitive Intelligence, e-tourism is an effective marketing tool for the world class promotion. Competitive Intelligence through existing strategies can be further refined for the development of e-tourism at a more advanced stage. The hope is to encourage decision-makers to pay more attention to the aspect of tourism promotion and build the city of Manado as a world tourism brand through the use of Competitive Intelligence as a strategy in e-tourism. Development of an e-strategy was also influenced by two factors, internal and external; and focus in this paper is the internal factors. The use of e-strategy will only be seen from the view of Competitive Intelligence.

II. COMPETITIVE INTELLIGENCE AND E-TOURISM

A. Competitive Intelligence

Competitive Intelligence is a continuous process of systematically collecting and analyzing information about the activities of competitors and trends in the business (the trend of political, economic, technological) to achieve company goals. In particular: continuous process from planning and direction, collection of information about the activities of competitors and the general business trends, evaluation and analysis (the conversion of information into intelligence), and the presentation of the analysis results. Basis of Competitive Intelligence is knowing the difference between information and intelligence. Information is based on the fact that the form of numbers, statistics, data about people and companies. Intelligence is a collection of information that has been filtered, evaluated and analyzed, also called knowledge.

B. e-Tourism

The Internet has provided a new economic environment for trading. e-commerce is one of the sectors that are growing and many tourism businesses were involved in the development of Internet services, such as travel agents, tour operators, airlines, hotels. Internet usage in the tourism business is known as e-tourism or e-business tourism. The difference with the traditional markets is the speed at which information can be communicated, access to global coverage, and a minimal cost to make an on-line business.

III. CYCLE OF COMPETITIVE INTELLIGENCE

The basic unit of Competitive Intelligence is the intelligence cycle, which is a process to transform information into intelligence. There are four steps in the cycle of Competitive Intelligence, it is a continuous (uninterrupted) process :

- Planning and direction
- Collection
- Evaluation and analysis
- Deployment / Presentation

a. Collection Strategy

This is an initial step in the Competitive Intelligence activities that include: a clear understanding of user needs to determine the purpose of using Competitive Intelligence, why Competitive Intelligence is needed, and who need it. This is very important because it served as the base for the next step. Reviewing activities of the company to look for information that already exists within the company. Establish a plan of collection and analysis, including: (1). Time planning, the schedule of Competitive Intelligence implementation in order that obtained the results on time and not late or outdated. (2) Determine the type of competition. (3). Determine the type of resources used. (4). Determine the type of analysis that will be used.

b. Collecting Information

This step is a very important activity of collecting the information to be processed into intelligence that can be used by management. Information may include formal information (written information) and informal information (human information). Information was obtained from various sources of information that can be used by the company. The information gathered is organized, which means that the information must be accessible to everyone in the company to be used as a tactical planning, making simple decisions, or simply to learn the information. However, analyzing the information is still to be done by the Competitive Intelligence unit.

c. Evaluation and Analysis

After the data/information collected through various resources, the next step is to do an evaluation of the information. It means doing the validity and reliability of the information. Validity is the accuracy and completeness of the information. Reliability is the trust of information sources on the basis of the performance. The analysis used depends upon the problem to be solved and the ability of the company. From the analysis results will obtained a finding, knowledge (intelligence) which is a strategic decision of the company to face the competitor activity.

d. Dissemination/Presentation of the analysis results

The final step is the presentation of intelligence, which is reporting the answers to the questions/needs of the executive.

IV. RESULTS

A. The e-tourism strategy (5C)

Through observation of the strategies undertaken by the tourism practitioner who already have a market in addition to the tourism products they had developed, strategy marketing is done in e-tourism focuses on 5 C.

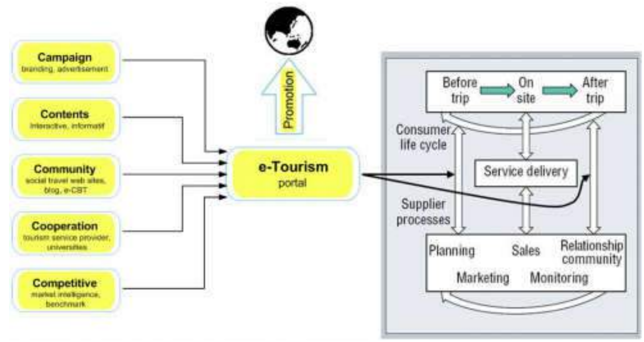

Figure 1. Implementing e-tourism strategy on the tourist life cycle and suppliers' processes

1. Campaign

Campaign, plays an important role in the promotion of tourism, this involves branding a destination and destination in order to strengthen the image of tourists may be interested in visiting the area. This can be done through variety of media to build brand awareness with other targets in the local community, to help improve the image of the destination. Advertising on the internet on the well known websites and have external links to other web sites, then search engine optimization can be improved. Ranking obtained in the search engines will be improved because of search engines searching method will ranked the site most widely referenced and most relevant keywords that are desired by the user.

2. Content

Content concerning the information contained, interactive and informative site. With informative contents, the tourists who want to know a tourist destination can easily know any place he wanted to visit. It could also achieved by updating information on online encyclopedia site, so users who want to find information about these keywords can be helped with the explanation there. With periodic updates on the website further enhance the content of the existing content. e-CBT (e-Community Base Tourism) can also be informed here, the products are owned by local people, handicraft which revolves around the small and medium enterprises owned by the community.

3. Community

Communities can be built through blogs, social media websites, e-CBT, a forum, or a social travel site. Maintaining these sites can be done by comment with additional information or other things that could build up the image of a destination area. Also by publishing an e-newsletter in order to provide additional information about a tourist spot. Word of mouth information with the help of the internet has gone global with the various social sites and social travel site will provide an additional positive information which would increase the attractiveness of the tourist areas.

As for developing e-CBT can develop local cultural events, festivals that rooted in local communities.

4. Cooperation

Build cooperation links with external providers of tourism services, such as travel agents, hotels, resorts, diver operators, transportation, local government and regional tourism destination, the telecommunications operator (cellular or Internet Service Provider). The University can also provide various resources or activities that can support tourism such as seminars or activities related to tourism. Local and regional governments can collaborate to build a destinations package that have the same theme or a theme that is different but the geographical location adjacent to each other.

5. Competitiveness

Know and understand the market by implementing various strategies obtained by benchmarking extensively with other tourism website, or to the annual reports of various destinations in other countries. The state of the tourism market, especially on e-tourism, also can be known by regular reports from the site UNWTO (United Nations Worlds Tourism Organization) or from the official websites of other organizations, through research on an area of tourism in order to improve the quality of tourism products in the region. Working together with the university to build or develop a tourism product through research that can be promoted through the internet. Results of this analysis will be an input for decision-makers mainly to market a tourism product through the Internet in order to become more competitive.

Figure 2. e-Tourism (5C) Strategy Implementation Mind Mapping

Table 1. SWOT analysis of Manado City e-Tourism Development

Internal Factors **Eksternal Factors**	**Strengths (S)**	**Weaknesses (W)**
	- Early promotion - Government Support - Human Resources - Information and communication technology - Law support - Potential nature and culture - Supporting infrastructure	- Management of e-Tourism - The cost of using ICT - Level of professionalism especially e-Tourism - Coordination, integration, and synchronization between and within institution in promoting tourism especially e-Tourism. - e-Readiness - Financial Support - Local Tourism Culture
Opportunities (O) - Tourism Development Trends - The development of Information and Communication Technology - Development of Internet - Internet users amount - e-Tourism in Indonesia - The presence of courses support tourism particularly e-tourism	**SO – Strategy** - Build an alluring branded, to make it easy to associate the brand with manado city tourism on any promotion media. - Build a strong image of a destination so that the branding and image will establish a product with a high level of attractiveness. - Build e-Tourism network with a network of links between tourism website, or with internet network community.	**WO – Strategy** - Encourage and strengthen local community-based tourism (Community tourism base, CBT) so that people can attract investors and use Internet to do promotion (e-CBT). - Cooperation with academic practitioner to increase the e-Readiness and how to increase local tourism culture, with a wide range of strategic planning.
Threats (T) - Global e-Tourism competition - Site Security - Internet network outage - Inflation and the global economic recession - Security Stability	**ST – Strategy** - Continue to strengthen the globalized promotion while developing destinations to other area or to establish synergies with destinations in the cities around. - The Government encourages the community to build a destination together that supports tourism activities in Manado. - Applying the law enforcement for law violator that infiltrate on Internet sites. - Benchmarking with other tourism areas, especially e-tourism in order to make Manado tourism become more competitive.	**WT – Strategy** - Empowering communities with the development of CBT, so there is a competitive local products that can be promoted via the Internet. - Achieving a common vision, especially at the Culture and Tourism Department that works as a bridge to the other institution, in order to improve the competitiveness of tourism, especially e-Tourism in the city of Manado so that it can become more professional.

REFERENCES

[1] Badan Pusat Statistik Kota Manado. 2004. *Manado dalam Angka*. Manado: BPS.

[2] Dou Henry, Manullang Sri. 2004. *Competitive Intelligence, Technology Watch and Regional Development*. Jakarta : MUC Publishing.

[3] Dou Henri Jean-Marie, Manullang Sri Damayanti, Dou Jean-Marie Jr. 2006. *Competitive Intelligence, Technology Watch for Industries*. France-Indonesia: Badan Penelitian dan Pengembangan Industri

[4] Rangkuti Freddy. 2006. *Analisis SWOT Teknik Membedah Kasus Bisnis*. Jakarta: PT Gramedia Pustaka Utama.

[5] Underwood Jim. 2003. *Intelijensi Kompetitif*. Jakarta: Prestasi Pustaka Publisher

[6] Kahaner Larry. 1997. *Competitive Intelligence : how to gather, analyze, and use information to move your business to the top*. New York: Touchstone.

[7] Staab Steffen, Werthner Hannes. *Intelligent Systems for Tourism. 2002.* Computer.org/intelligent.

A Preliminary Design and Testing of The On Board Data Handling (OBDH) for Nano-Satellite Using an Atmospheric Balloon

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***Abstract*—There are must-to-do steps in the nano-satellite development process, including the engineering model and flight model test. The atmospheric balloon test can be used to analyze the On Board Data Handling (OBDH) performance. The obtained results from the test are 13 images data captured from the 2900 MASL maximum altitude, also the sensor data i.e. temperature, pressure, humidity, position, and the direction of the balloon payload.**

Keywords—The On Board Data Handling (OBDH); payload; telemetry; atmospheric balloon.

I. Introduction

The Atmospheric Balloon Competition (KOMBAT) 2015 has been held by LAPAN (an Indonesian aviation and space agency) in collaboration with Ministry of Research, Technology, and Higher Education in Indonesia. At the same time, Telkom University was in the middle of nano-satellite developing process, specifically about the On Board Data Handling (OBDH) performance test. So, the Telkom University joined the competition in other way to test and analyze the performance of the OBDH system. There are some research about the OBDH had been done, i.e. [1] using Samsung S3C2440A processor and CAM130 camera with the OV9650 sensor; [2] and [3] using FPGA with single camera and array cameras to enhance the image; and [4] using the array cameras equipped by MMED LPC1768 processor as OBC. In contrast to previous research, the processor used in this research is ATMega2560 followed by reasons: it has varied pins and has a large memory [5].

In the competition, it has been arranged various binding rules related to critical issues, such as the size of the balloon payload, the weight of the balloon payload, the frequency used for communication between the payload with the ground station for both of data sensor or delivery of image data, and sensor data table format [6]. To anticipate this, the system is designed to be lightweight and also has a reliable communications for sending sensor data and image data.

Balloon payload can be used to determine the condition of the atmosphere in a way to record and transmit sensor data related to temperature, relative humidity, pressure, horizontal wind speed, and the image data in real time [6]. So, the qualify sensors must be chosen in order to gather those information. ZMCamera as the camera [7], the GPS to determine the position of the payload [8], DHT 21 as the humidity and temperature sensor [9], as well as BMP 180 to determine the altitude and pressure of the payload [10].

This paper is organized into 4 sections with the first section contains a preliminary study. The second part is about the system design. The third part is the data analysis from the measurement results. And, the fourth part contains the conclusion.

II. System Design

The system designed in this experiment includes balloon payload system which consists of several sensors that required to measure the conditions around the payload, as well as the antenna pointing system at the ground station that controlled remotely. The design of the system follows the race regulations, related to the functions of the sensor should be provided such as height, position, temperature, pressure, humidity, and image data.

Ground station specifications that must be fulfilled, are: the antenna alignment is not done manually, but it must be controlled remotely by wireless. At the balloon payload itself is designed to weigh no more than 200 grams and must accommodate all sensors, structure, the battery, and communication devices of the balloon payload to the ground station [6]. Fig. 1 is a general system of communication between the antenna pointing system at the ground station and the balloon payload system containing various sensors and communication devices.

A. *Antenna Pointing System Design*

The controller of the antenna direction is made by using a Bluetooth module HC-05. It follows the requirements of competition which had to use a wireless communication with a minimum distance of 2 meters [6]. In Fig. 2, it is illustrated the system of antenna controller with the input from a PC. Then,

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that is forwarded by HC-05 on the PC and received by the HC-05 on the antenna pointing system. The battery of antenna control block is 7 volts to supply Arduino, 3.3 volts to be distributed to HC-05, and 5 volts to the azimuth and elevation servo. The antenna movement system is designed to work by giving input by pressing the "R", "L", "U", "D" and "N" on a PC which is sent via Bluetooth HC-05. Then, this input will be translated by Arduino Uno to move azimuth direction CW, CCW, or stop; and also elevation CW or CCW.

Servo that works on the azimuth axis has the continuous movement properties. By pressing the "R" on the PC, it will make the clockwise movement, while the "L" for the counterclockwise movement. Added the "N" key in order to dismiss the azimuth movement. Servo works continuously by inserting value into the servo from 0^o to 180^o, if the value is less than 90^o then the servo will move counterclockwise and value over 90^o will move contrary. While the value equal to 90^o to stop the movement of the servo.

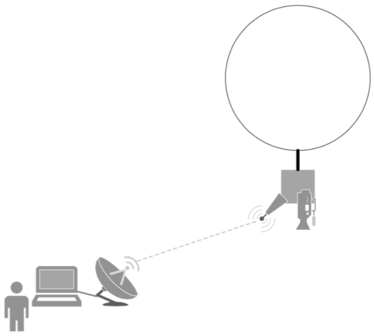


Fig. 1. Balloon payload and antenna pointing system.

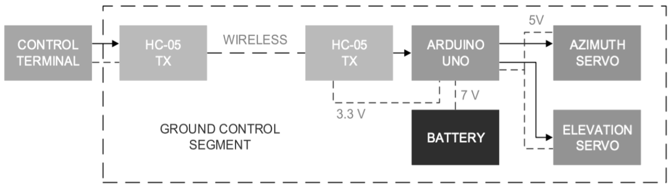


Fig. 2. The schematic diagram of the antenna pointing system at the ground station.

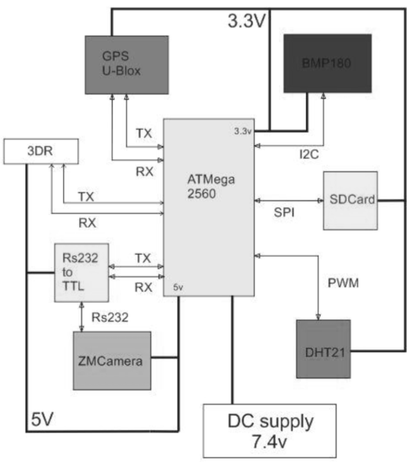


Fig. 3. The schematic diagram of the balloon payload.

For the elevation movement, it is using 2 pieces of the servo with the 0^o to 180^o movement ability. This elevation servo moves based on the amount of the angle, not continuously. The driving system of the antenna to the elevation direction is designed to be capable to make the 0^o to 90^o movements by pressing the "U" (to increase 1^o) and pressing the "D" (to decrease 1^o). If there is no command, the servo will maintain the end position.

B. Payload of The Weather Balloon System Design

The purpose of KOMBAT is to measure the conditions in the atmosphere at a maximum altitude of 3000 MASL [6]. The integrated sensors are GPS, temperature and humidity sensors (DHT 21), altitude and pressure sensor (BMP 180), and a camera to take pictures.

Fig. 3 describes a schematic system of the weather balloon payload. The On Board Data Handling uses ATMega2560 that connected to the sensor DHT21, BMP180, GPSU-Blox, and ZMCamera via RS232 to TTL interface. In addition, ATMega2560 has I2C interface, PWM, and SPI [5]. Voltage requirements of each sensor and also OBDH supplied by a 3.7 volts battery arranged in series in order to produce 7.4 volts. All the sensor data will be transmitted through 3DR modules corresponding to the 3DR module channel of the ground station.

For the beginning of the communication between the ground station and the payload (Fig. 4), the 3DRs channel must be paired and locked. Then the GPS locks the location so the location data is forwarded to ATMega2560 but not yet written to the SD card. When the payload started to fly, all the sensors perform their respective functions, but the sensor data is still stored in the register in the sensor. When there is a request from ATMega2560, the sensor sends the data to ATMega to be stored on the SD card in the string form that will be sent within an interval of 1 second. When the payload altitude reaches 500 MASL, the camera takes a picture then ATMega sent it to the SD card. The data then will be sent to the 3DR. On the process of sending pictures, all the sensors keep records of sensor data, but ATMega will not ask sensor data that still stored in the sensor registers. So, only the last data will be stored in the sensor registers if ATMega2560 does not request it. This process is repeated in every multiple of 200 meters above sea level. The SD card inserted into the payload is 2 GB to accommodate the image data (that is no more than 20 images with a size of 320×240 pixels) and all of the sensor data during the flight.

1) ATMega2560

ATMega series microcontroller is used because it has a capacity of 256KB flash memory and 4KB SRAM, so as to process large programs it will not interfere the system stability. Additionally, the number of I/O pin is available as many as 86 pins and serial port is 3 pieces. That makes it possible to use many modules simultaneously by using this microcontroller [5].

2) ZMCamera 0.3 Megapixels

ZMCamera 0.3 MP is used as image acquisition sensor by using a RS232 communication protocol and a CMOS sensor, so it takes a RS232 to TTL converter in order to communicate with ATMega2560 [7]. ZMCamera requires a supply of 5 volts with the resulting image has a JPEG format, so that the size of the image has been compressed. Results image of ZMCamera is not directly transmitted to the ground station, but it is stored on the SD Card. This is done so that the sensor data does not accumulate and overload in ATMega2560.

Fig. 4. The balloon payload preparation before flight.

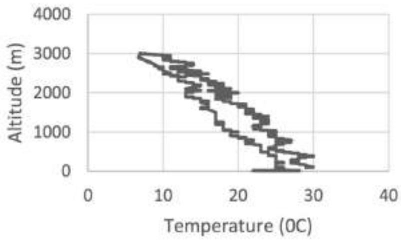


Fig. 5. The result of temperature (°C) vs. altitude measurement.

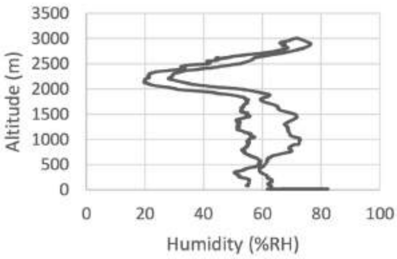


Fig. 6. The result of humidity (% RH) vs. altitude (MASL) measurement.

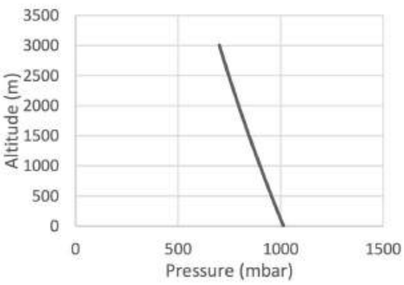


Fig. 7. The result of pressure (mbar) vs. altitude (MASL) measurement.

3) GPS U-Blox Neo-6M

The selection of U-blox GPS modules NEO-6M version because it uses NMEA protocol and also has available antenna so as to accelerate the process of data acquisition from satellites [8]. The data obtained from this module is longitude, latitude, and direction data. The voltage needed was 3.3 volts to operate this GPS module.

4) DHT21

To obtain the data of temperature and humidity, DHT21 sensor which has the capability of -40°C up to 80°C is used [9]. This sensor uses a digital communications using the PWM pin in the ATMega2560. Advantages of this sensor, it can be used for a long time without disturbing the system stability. Just as the GPS module, the applied voltage is 3.3 volts.

5) BMP180

To measure the altitude and pressure, BMP180 sensor that uses I2C communication protocol with ATMega2560 is used [10]. Because of using that protocol, it takes a pull up resistors on the SDA and SCL pins. The altitude values can be obtained from the pressure data, so it takes the calibration beforehand by putting pressure on the sea water surface. This module can work with voltage requirement of 3.3 volts.

III. Data Analysis

The system testing is done by laying the payload on the atmospheric balloon with the average vertical speed of 5 m/s. The first image capture is done when it reached a height of 500 MASL; then, the next image capture will be taken periodically for the next 200 MASL until the altitude reaches 3000 MASL. After that, the data transmission of the temperature sensor, humidity, and pressure are sent in real time every second to be processed in ground station.

A. ZMCamera Image

The camera module has worked well with capturing first image at an altitude of 500 MASL, then a second image at an altitude of 700 MASL and so on up to a maximum altitude of 2900 meters above sea level. The size of the pictures taken at 320×240 pixels with an average of 9 seconds capturing time and delivery time to the ground station by 9 seconds. So, the total time to take and send pictures is 18 seconds. The images result of the ZMCamera module shown in Appendix 1.

B. Temperature and Humidity

DHT21 sensor is used to measure the temperature and humidity. It is performed until a maximum altitude of 3000 MASL. Fig. 5 and 6 shows the obtained data from the temperature and humidity measurement. The minimum temperature there at an altitude of 3000 MASL that is equal to 7°C and the temperature slowly rises when the balloon down to the altitude of 500 MASL. For humidity value; at an altitude of 2000 up to 2500 MASL, the humidity is at its lowest point that is equal to 20% up to 40% RH.

C. Pressure

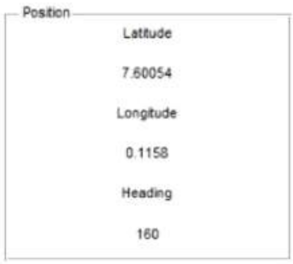
The pressure measurements obtained from BMP180 sensor that measures the height of the balloon payload. To get the pressure value, a calibration should be performed prior to the pressure value at sea level. Fig. 7 is a graph obtained from BMP180. The graph states that when the altitude increases, the pressure will be reduced.

D. Payload Tracking

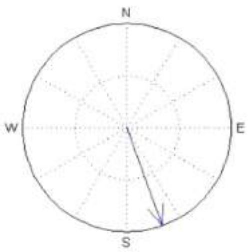
Which is used to perform the tracking of the balloon payload is the data from the GPS sensor in the form of data longitude, latitude, and compass. To get the GPS data from the satellites, the GPS must remain capable of receiving the location data from the NMEA protocol of the satellite. The adjustment takes about 1 minute. From the value of the coordinates of longitude and latitude, the location of payload can be obtained by mapping that value on the graphic user interface (GUI). Fig. 8 shows the results of the initiation of GPS, compass directions and the movement of the payload. The tracking of the balloon payload is predicted to fall in the mainland.

IV. CONCLUSION

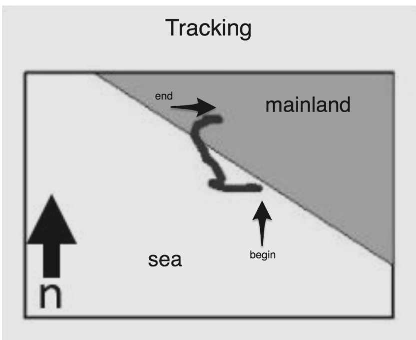
From the research and testing, it has been made a telemetry system that serves to measure the atmospheric conditions related to temperature, relative humidity, pressure, position, altitude, compass directions, and also image data. It is obtained 13 image data captured from 500 MASL until the maximum altitude at 2900 MASL. The temperature lowest level measured is $7^{o}C$ at 3000 MASL and the lowest humidity is 20% RH at an altitude of 2000 – 2500 MASL.



(a) The latitude and longitude tracking data.



(b) The compass direction of the payload movement.



(c) The movement of the payload.

Fig. 8. The results of payload movement tracking during the balloon flight test.

ACKNOWLEDGEMENT

This study is an early step to test the performance of On Board Data Handling (OBDH) as a small part of the nano-satellite research at the Telkom University funded by DIKTI. The participation in the atmospheric balloons competition (KOMBAT) is funded by the LAPAN through a long selection process. Ararkula team from Telkom University became the winner for the atmospheric balloon category at KOMBAT 2015.

REFERENCES

[1] Edwar and M. A. Murti, "Implementation and Analysis of Remote Sensing Payload Nanosatellite for Deforestation Monitoring in Indonesian Forest," in 6th International Conference on Recent Advances in Space Technologies (RAST), Istanbul, 2013.

[2] D. Cahyono and A. Nugroho, "Design and Realization Camera Controller for a Remote Sensing Payload of Nanosatellite FPGA (Field Programable Gate Array) System Based," in 6th International Conference on Recent Advances in Space Technologies (RAST), Istanbul, 2013.

[3] W. Pakartipangi, B. Syihabuddin and D. Darlis, "Design of Camera Array Interface Using FPGA for Nanosatellite Remote Sensing Payload," in International Conference on RADAR, Antenna, Microwave, Electronics and Telecommunications (ICRAMET), Bandung, 2015.

[4] H. W. E. A. A. F. Alfian Subiyakto, "Design of Synthetic Aperture Optical System (SAOS) for Nano-Satellite Remote Sensing," in Regional Conference on Computer and Information Engineering (RC-CIE), Yogyakarta, 2014.

[5] Atmel Corporation, "Atmel ATMega640/V-1280/V-1281/V-2560/V-2561/V Datasheet," Atmel Corporation, 2014.

[6] LAPAN, "The Guidance Book of Atmospheric Balloon Payload Competition (KOMBAT) 2015", *In Bahasa*, LAPAN, 2015.

[7] Shenzhen Quick Zoom Technology Co., Ltd., "Serial Camera Module Data Sheet," Quick Zoom Technology Co, Ltd.

[8] U-Blox, "NEO-6 Series: Versatile u-blox 6 GPS Modules Data Sheet," u-blox AG, 2011.

[9] Aosong (Guang Zhou) Electronics Co., Ltd., "Temperature and Humidity Module AM2301 Product Manual," Aosong (Guang Zhou) Electronics Co., Ltd..

[10] Bosch Sensortec, "BMP180 Digital Pressure Sensor Data Sheet", Bosch Sensortec, 2013.

APPENDIX 1

Fig. 9. ZMCamera image taken from 500 MASL.


Fig. 10. ZMCamera image taken from 700 MASL.


Fig. 11. ZMCamera image taken from 900 MASL.


Fig. 12. ZMCamera image taken from 1100 MASL.


Fig. 13. ZMCamera image taken from 1300 MASL.


Fig. 14. ZMCamera image taken from 1500 MASL.


Fig. 15. ZMCamera image taken from 1700 MASL.

Fig. 16. ZMCamera image taken from 1900 MASL.

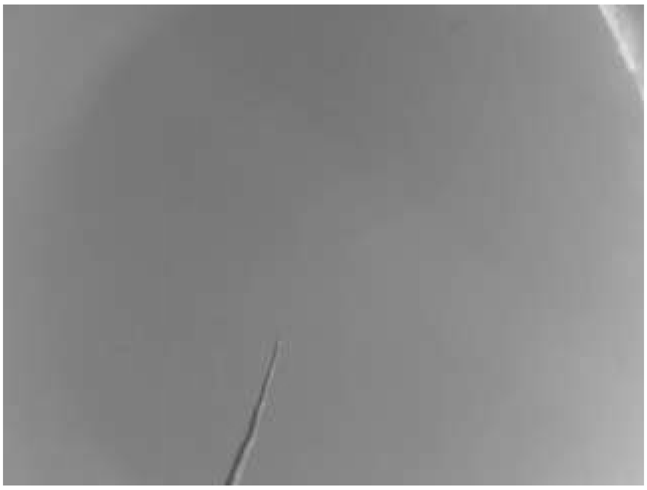


Fig. 17. ZMCamera image taken from 2100 MASL.

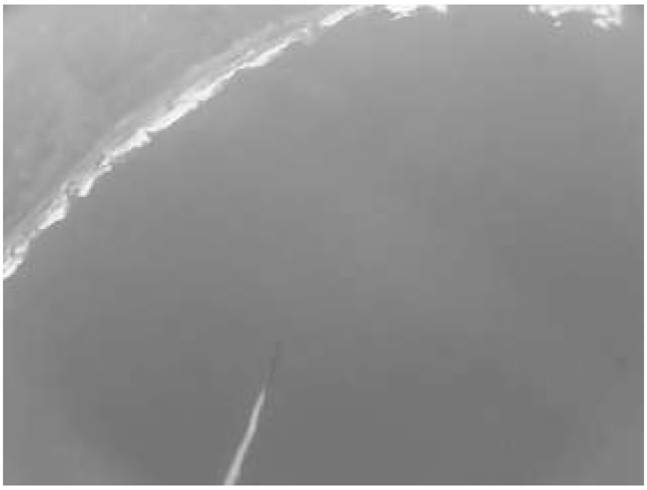


Fig. 18. ZMCamera image taken from 2300 MASL.

Fig. 19. ZMCamera image taken from 2500 MASL.

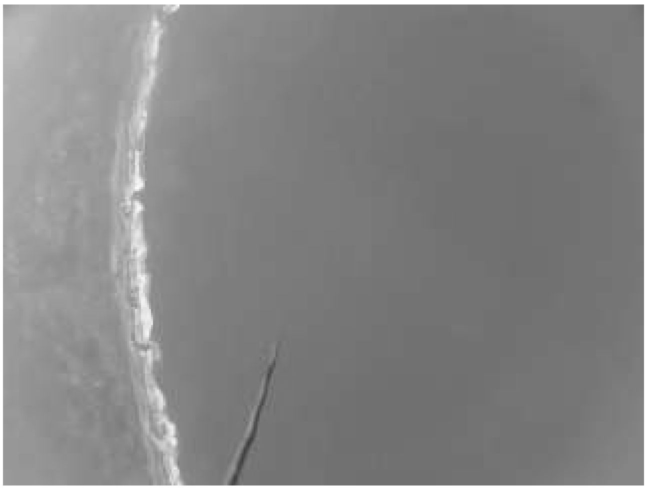


Fig. 20. ZMCamera image taken from 2700 MASL.

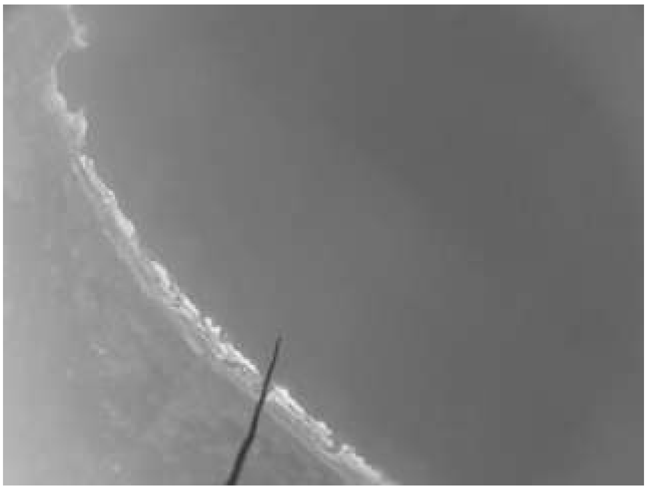


Fig. 21. ZMCamera image taken from 2900 MASL.

Application For Parking Area With Android Smartphone

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Abstract— The development of technology has an impact for the transportation sector, one of them is about parking the vehicle. Parking is a state where a vehicle is not moving for a short time because the driver is going for somewhere else.

To find an empty parking spot, a driver must look up for a parking lot that has been allowed for parking. it is quite difficult to find a parking spot because the space available for parking is very limited and the amount of vehicles that need parking is very high. It needs a relatively long time for a driver to get a parking spot and can cause a traffic jam in the parking lot because many drivers are trying to find an empty spot for their vehicles.

This is the reason why the author develops an application for parking lot using Android Smartphone. This paper uses Rational Unified Process Methodology and the tool that used for this paper is Photoshop CS5, Adobe Dreamweaver, MySQL, and using Unified Modeling Language tools. The programming language that used for this paper is PHP (Hypertext Preprocessor).
Key words: Parking, Android

I. INTRODUCTION

The electronic world has good relevance with Technology Development. The increasing development of technology will bring convenience to people's lives. Smartphone is very widely used and has become an important part of everyday life. For example, an android based mobile phone is one of the advanced electronic media, popular among the public and lot of Android applications began to appear.

In the transportation sector, one of the technology development can be found in the parking sector. Today many parking lots have improved the quality of parking services, either by adding a camera in the parking lot or providing parking attendant to assist driver who wants to find an empty parking spot. To get the empty spot, a driver must surround the parking lot several times, because the availability of space is limited and the number of cars that want to park is huge. It needs a relatively long time for the drivers to get a parking spot and can cause a traffic jam in a parking lot because many drivers is trying to find an empty spot for their vehicles.

Based on the description above, there is an opportunity to develop an application for parking lot using Android Smartphone. This application can help driver (either cars or motorcycles) to find an empty spot in a short time easily.in short time easily.

II. RESEARCH QUESTION

The research question is how to develop an application for parking lot using Android Smartphone that can help driver(either cars or motorcycles) to get an empty spot for parking quickly?

III. STATE OF ARTS

A. Parking

Parking can be interpreted as a place to stop or to put the vehicles (cars and motorcycles) for a few moments at a designated place. Parking is a state of the vehicle stops or does not move for a while and left by driver [1].

B. Vehicle

Vehicles are thing that driven by technical equipment located on the vehicle.

Vehicle are classified into several types, namely [2]:

a. Motorcycle
Motorcycles are two-wheeled vehicles without a sidecar.
b. Minibus
Minibuses are vehicles that equipped with more than eight seats excluding the driver's seat, with or without equipment hauling luggage.
c. Bus
Buses are vehicles that equipped with more than eight seats excluding the driver's seat, with or without equipment hauling luggage.
d. Truck
Trucks are motor vehicles other than motorcycles, minibus and bus.

e. Special vehicles

Special vehicles are motor vehicles that used for special purposes or transporting special items.

C. Android

Android is a mobile operating system that adopts Linux operating system, but has been modified. Android was taken over by Google in 2005 of Android, Inc. as part of a strategy to fill the mobile operating system market. The main advantage of Android is an application that has integrated approach [3]. Android as operating system based on Linux is used for cellular phones such as smartphones and tablet computers (PDA). Android provides an open platform for developers to create their own applications that are used by a variety of mobile devices [4].

D. Methodology

The methodology used in the development of a parking lot simulations are Rational Unified Process (RUP). RUP is a software engineering methodology developed by collecting a variety of best practices. The main feature of this method is to use use-case driven and iterative approach [5].

IV. ANALYSIS AND DISCUSSION

A. Definition of Developed Application

Application for parking lot using Android Smartphone is developed to help driver (cars and motorcycles) to get an empty spot using smartphone with Android based that can be access anywhere with internet connections.

B. Analysis of Target Users

Table 1. Target Users

User	**Role**	**Responsibility**
Adminstrator	Adminstrator	- To add officers - Change username and password - View incoming and outgoing reports of vehicles
Officer	Officer	To add the identity number of parking
User	Society	To look up for an empty spot in parking lot

C. Use Case Model

This stage will explain and illustrate the description of the running system.

Table 2. Use Case Descriptions

Use Case	**Description**
Taking cars / motorcycles receipt	Contains barcode that has vehicle’s licence plate
Look up for an empty parking spot	Driver look up for an empty parking spot
Restoring receipt	Driver must return receipt before get out of the place where the driver is located.
Receive a receipt	Officer scans barcodes on the receipt number.

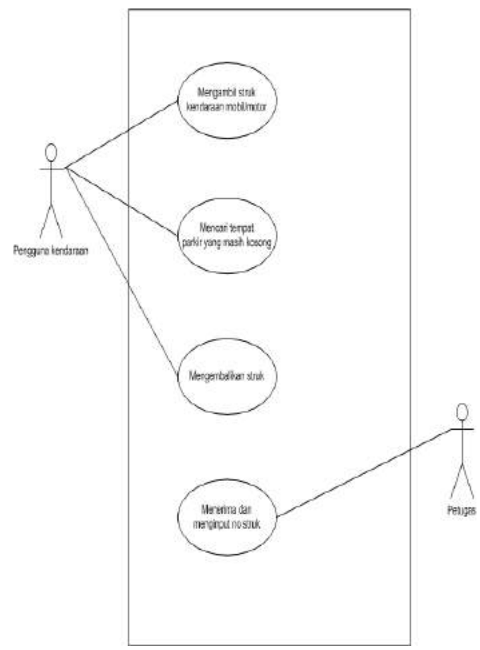

Figure 1. Use Case of Running System

Before entering the parking lot, cars or motorcycle riders must take a parking receipt. The rider can choose the button A, button B or Button C. To know that the parking lot is full or not, the rider must install the application on smartphone. Status of Parking lot have been filled became blank when officer enter the license plate number and the riders will be out of the parking lot

V. DESIGN

In this stage, we are going to discuss about the development of parking lot application using Android Smartphone.

A. Use Case Diagram

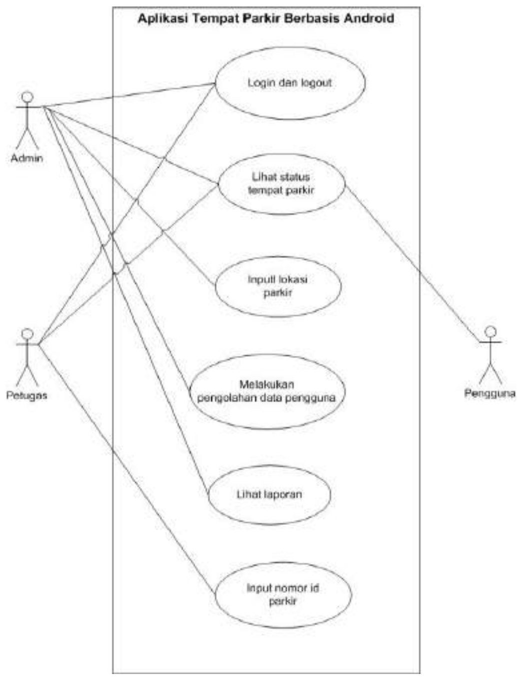

Figure 2. Use case diagram of current system

B. Implementation Interface

In Figure 3, the main menu storyboard shows the button of cars and motorcycle

Figure 3. Main Menu Storyboard (User)

In Figure 4, car menu storyboard displays filled parking lot number based on existing location.

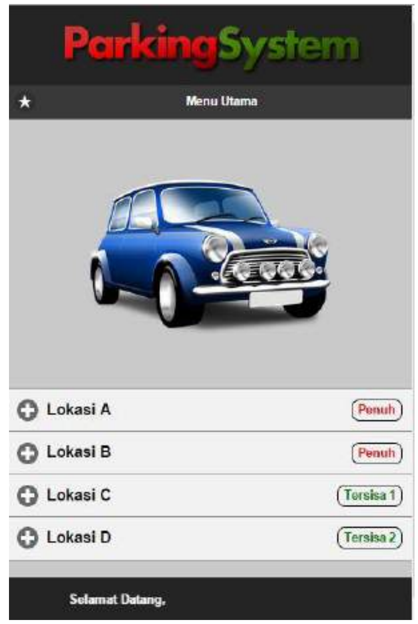

Figure 4. Car Menu Storyboard (User)

In Figure 5, parking lot status storyboard displays both the total of empty parking lot status and filled parking lot status.

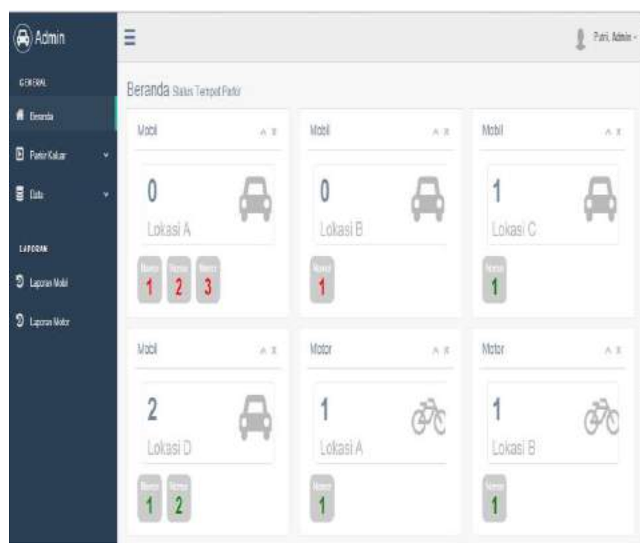

Figure 5. Parking Lot Status Storyboard (Officer)

In figure 6, outgoing vehicles storyboard (officer) displays number form, parking identity (parking receipt) to be filled by officer and to displays price that must be pay by outgoing vehicles.

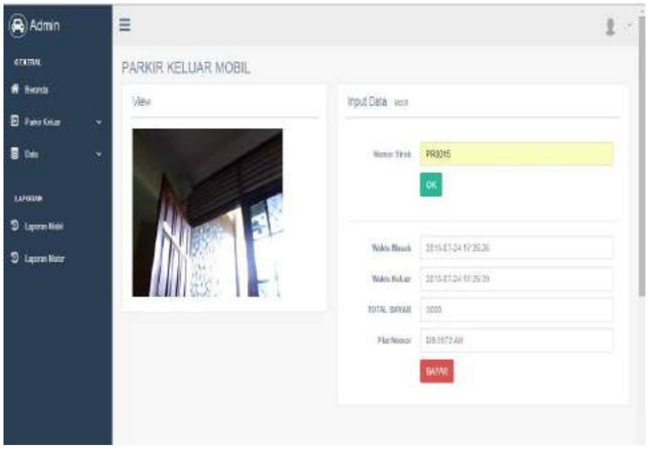

Figure 6. Outgoing Vehicles Storyboard (Officer)

In figure 7, location's data storyboard (administrator) used to input new location, change location and delete location not being used.

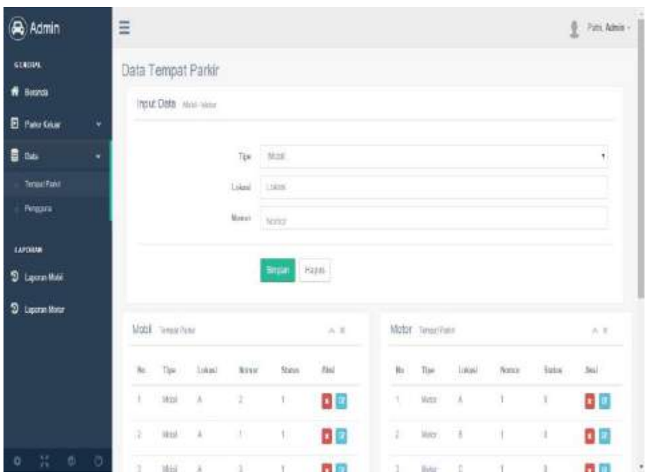

Figure 7. Location's Data (Admin)

In figure 8, Graph Reports Storyboard (Administrator) to displays incoming vehicles report by using graph.

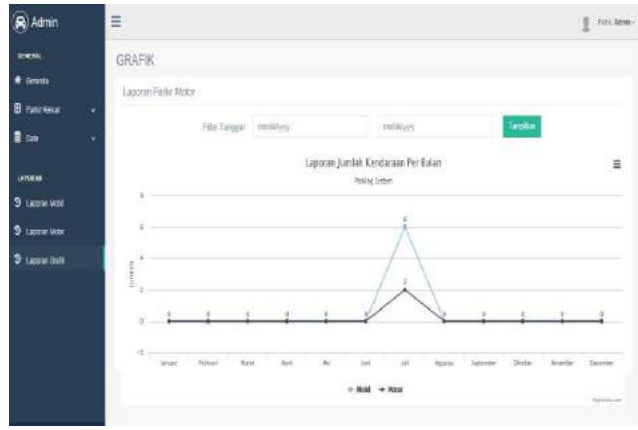

Figure 8. Graph Reports Storyboard (Admin)

VI. CONCLUSION AND ADVICE

A. Conclusion

Based on the results of the application parking lot using Android Smartphone Development, it can be concluded that:

1. This Application can displays a total of empty parking spot.
2. This application can help drivers to find an empty parking spot In the building.
3. This application can reduce traffic jam when the driver is looking for an empty parking spot.

B. Advice

Therefore, there is an advice to improve this application in the further to make it even better such to add parking lot booking feature.

REFERENCE

[1] Undang-Undang No.22 (2009). Undang-Undang Lalu Lintas & Angkutan Jalan 2009. Jakarta: Transmedia Pustaka.
[2] Undang-Undang Republik Indonesia No.14 Tahun 1992 tentang Lalu-Lintas dan Angkutan Jalan. Jakarta: Transmedia Pustaka.
[3] Suprianto, D. Agustina, R. (2012). Pemrograman Aplikasi Android. Yogyakarta : Mediakom.
[4] Murya, Y. (2014). Andoid Black Box. Jakarta : Jasakom.
[5] Rosa, A dan Shalahuddin, M. (2011). Modul Pembelajaran Rekayasa Perangkat Lunak (Terstruktur dan Berorientasi Objek). Bandung: Modula.

Fan Temperature Detection Using Microcontroller

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Abstract — Current fan is not able to display room temperature and humidity. It also cannot run automatically on a specified time or room temperature. Therefore it should consider a system that can overcome these weaknesses. This research will make a room temperature control system that can detect temperature and humidity, and displays the results to the application. Systems made useful to control the fan so that the room temperature is still cool. The system will interact with the user through the application, both for switching on and off the fan directly or automatically based on temperature and time. This research uses the C# programming language in the creation of applications and C for coding of microcontroller. The methodology used is prototyping. Testing conducted concluded that the application can display the values of temperature and humidity, a function to turn on and turn off the fan directly or based on temperature and time can run well.

Keywords: ***fan, detection, microcontroller, temperature.***

I. BACKGROUND

Humans need cool air in order to move comfortably. Many electronic devices are made to maintain the room temperature stays cool. The fan is one of the air conditioners that were encountered but the current fan is not able to display the room temperature and humidity. The fan also cannot adjust automatically with the room temperature or time. In addition, the fan is executed by pressing the on-off, by hand. To overcome the above drawbacks and to support the performance and automation on the fan requires a microcontroller and a program to control it.

The microcontroller is an integrated chip that typically becomes part of an embedded system designed to perform one or more specific functions in real time. Microcontroller shape is very small and simple and includes all the necessary functions on a single chip.

Based on the description above, this research will create an application that can control the fan *(on-off)* based on the room temperature and time automatically using microcontroller so that the room temperature is still cool.

II. LITERATURE REVIEW

A. Detection

The detection is a process of identifying a problem if there is a failure in the prevention and notify the executor [1]. The detection tool is a tool used by user to help identify a problem.

The process of identifying the problem starts from the existence of a problem that appears (where there is a state which is not in accordance with the state in general) was realized and find a way out, either by step overall settlement or quite simply reached the point of prevention.

B. Microcontroller

Immersa [3] stated that the microcontroller consists of CPU, Memory, I / O ports and timers like a standard computer, but is designed only to carry out a specific function in regulating the system.

There are many microcontrollers in stores however microcontroller used in this research is microcontroller DHT22 as a temperature sensor. The microcontroller set on Arduino board. The main component in the board Arduino used is a microcontroller 8 bit with brands ATmega made by the company Atmel *Corporation.* A simplified block diagram exist in ATmega328 microcontroller used in Arduino Uno (*unoDFRduino*) can be seen in Figure 1.

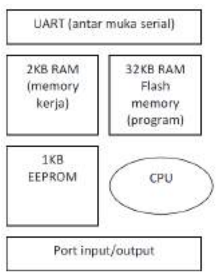

Figure 1. Block diagram of *microcontroller ATmega328* [2]

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This research is used a microcontroller DHT22 [6] because it has several advantages over another temperature sensor.

C. Temperature Sensor

The temperature sensor is a component that is normally used to convert heat into electricity to ease the analysis magnitude [4]. The temperature sensor is made using a metal or a semiconductor material. This method is used because the metal or semiconductor material can change its resistance to electrical currents depending on temperature.

D. Data Communication

Data communication is necessary for exchanging data between sender and receiver in this study. This data communication is needed to give commands from the application to the Arduino to turn on and turn off the fan through relay, and to send the value of the temperature and humidity of the sensor to the application via the Arduino.

E. Related Research

There are several studies related to this research, which are the Temperature Gauge System Design Using Arduino and C # .Net by Lucky Yuditia Putra and the Design and Implementation of Control Room Temperature Using Microcontroller-Based Sensor LM35 Arduino Uno by Fadilla Zennif.

Putra [5] used LM35 sensor to determine the temperature of the room easily and use .Net C # as the programming language. The system made can measure the temperature of the room with a tolerance value of data is not stored in one minute. When the temperature exceeds the set temperature, the fan will rotate automatically as an air conditioner. The system works every second and displays the result on desktop application and saves it to the database as a repository.

While Zennifa [7] made the room temperature controller based on input from the keypad. This tool uses the temperature sensor with the code LM35 and type LM35DZ, which has its advantages and conveniences that range of temperatures measurability quite wide, has a high accuracy, the economic cost and also use Arduino. The system also use LCD as a display and *keypad* as one of the factors that made the temperature can be controlled. The magnitude of the temperature is read will be displayed on the LCD and then the output will be shown on rotation of the fan.

III. ANALYSIS

A. Preleminary Analysis

Based on analyzing the previous researches and the observation made, this research generating some specifications of the requirements needed. This research will build a system that can control a fan using DHT22 as microcontroller and create an application to control the system.

As for the specifications of the application made is:

1. Application can detect the room temperature and humidity.
2. Application can transmit temperature values from sensor DHT22 to the microcontroller.
3. Application can display temperature values that exist on the microcontroller in the application.
4. Application can turn off and turn on the fan based on room temperature and time.
5. Application can turn off and turn on the fan without the value of the room temperature or time.

B. User Analysis

Table 1. User Analysis

No.	User	Task
1.	Human	Use and control the application and system.

C. Data and Communication Analysis

This section will discuss the communication between users, systems and applications created as follows:

Users used the system access the existing applications on the computer but before that, the user must choose the *ports* used by the microcontroller. If the *ports* are correct, the application will display the room temperature data were taken from the temperature sensor DHT22 through the microcontroller.

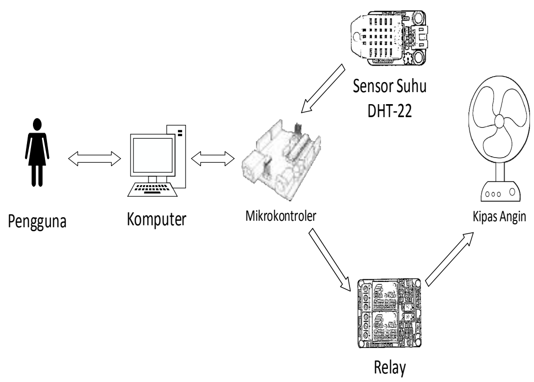


Figure 2. Overview of System Performance

D. Flowchart

Figure 3 described the process of application and detection system made as follows:

To get started, a user is required to access an application then the user selects the ports that are connected to the microcontroller. If the user choose the right ports, then the application will display the temperature and humidity in the room, as well as time.

For setting, if a user fills temperature limit, then the application will store it into a variable InputSuhu later then do timer settings. If the user set the timer, then the application will save to a variable timer then activate the system. When the user activates the system by pressing an on button on the application, the system will run in accordance with the temperature and timer settings that have been done in the previous stage. If the temperature is not filled and the timer is not set by the user, then the system will automatically run the fan. By the time the user wants to turn off the system, then the user simply presses the off button available on the application, if not then the system will run continuously in accordance with the arrangements that have been made.

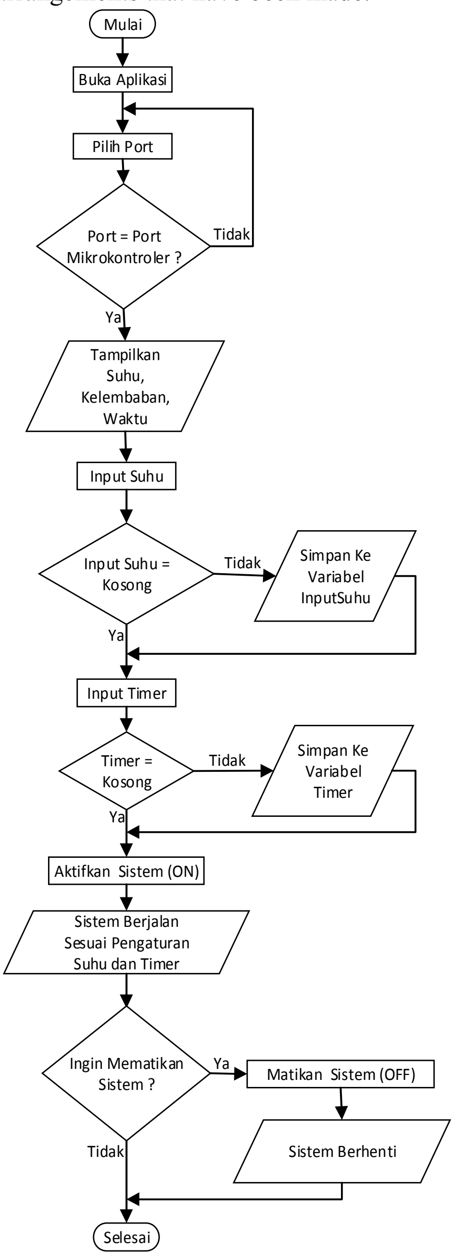


Figure 3. Application and Detection System Flowchart

IV. DESIGN AND IMPLEMENTATION

After conducting the analysis, the design of detection systems and applications is:

A. *Interface Design*

After doing analysis, the creation of application is made. Figure 4 shows the application interface. The design shows when a user has selected ports a microcontroller connected and has hit the refresh button, the user can fill in the normal temperature textbox as the room temperature limit. It aims to be switched on when the fan is above the normal temperature and will shut down if it is at normal temperature or below. Users can choose the time to turn on and turn off the fan. Besides that, the application will display the indoor temperature and humidity changed every second.

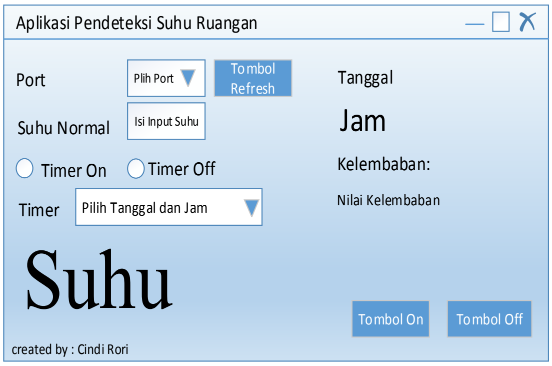


Figure 4. Interface Design

B. *Database Design*

To keep the temperature and humidity values required a design database that described in this following *Class Diagram*:

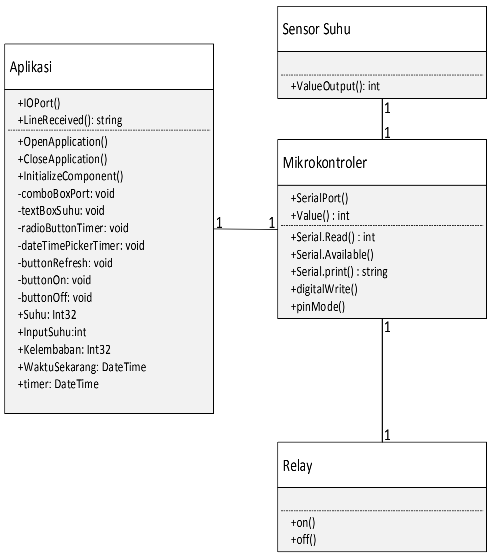


Figure 5. *Class Diagram*

C. *Hardware Design*

Applications will be connected to the microcontroller via USB so that the user can interact. As seen in Figure 6, pin 5v and gnd useful as a power of the microcontroller to the breadboard, henceforth connected to temperature sensors and relays. Pin 2 microcontroller connected to the relay will give the command on or off, while pin 3 microcontroller connected to the temperature sensor will accept the values of temperature and humidity. Fan cable consists of two small wires. 1 cable is cut to go first to relay to then be connected to a power source. It is intended that the process of turning on and off the fan can be controlled by the relay.

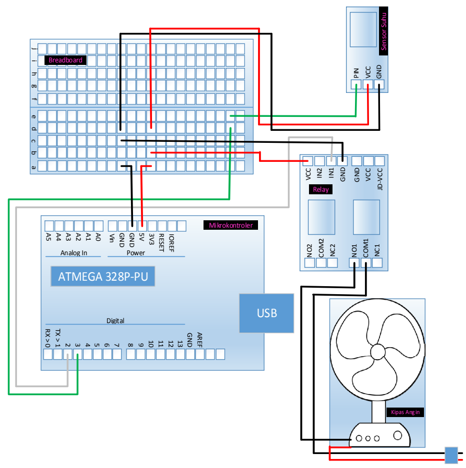


Figure 6. Hardware

The existing hardware circuit further implemented using various tools as follows:
In the figure 7, all the components are inside the box. Sensor facing out of the box in order to test the air temperature easily. Fans used as a boaster tool is a laptop cooling fans. Connections between components using jumper cables through breadboard. Using breadboard facilitate cables jumper can be connected without the need for soldered and easily moved. USB cable from the microcontroller is made out of the same box with a USB fan cable connected to the relay so it is easy to connect to the computer.


Figure 7. Control System Room Temperature

V. TESTING AND RESULTS

A. *Application Testing*

After the application and hardware are connected, the system will be test if it is running well or not.
The purpose of the application testing is done as follows:

1. To check whether the system is made to detect the temperature of the room.
2. To test whether the system can send the temperature value from the sensor to the microcontroller DHT22
3. To determine whether the system can display temperature values that exist on the microcontroller in application.
4. To ascertain whether the fan can function based on the temperature of the room.
5. To ascertain whether the fan can function based on the specified time.
6. To ascertain whether the fan can function without the value of the room temperature or time.
7. To test whether the function of the system can only be accessed through an application.

B. *Criteria of Testing*

The criteria of testing for this application is:

1. Application can be run when a right port is chosen.
2. All the functions of the application is running properly.
3. Controlling the fan runs fine on AC or DC current.

C. *Testing Case*

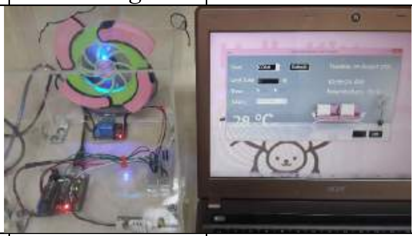
Here are some testing cases and their result:

Table 2. Communication and Relay Test

No.	*Activity*	*Result*
1.	*Choose the ports*	*All ports are on*
2.	*Press the refresh button*	*Value of temperature and humidity will be displayed*
		
3.	*Command turn on*	*Fan on*
		
4.	*Command turn off*	*Fan off*

Table 3. Time and Temperature Test

No.	*Activity*	*Result*
1.	*Command turn on after temperature*	*Fan on and off based of temperature setting*

No.	*Activity*	*Result*
	setting	
		
3.	*Command turn on after on time setting*	*Fan on based of time setting*
4.	*Command turn on after off time setting*	*Fan off based of time setting*

VI. CONCLUSIONS AND RECOMMENDATIONS

A. Conclusions

1. The system can switch on and off the fan automatically based on the room temperature and time setting.
2. The system can display the temperature and humidity directly.
3. The detection system and application are run well.

B. Recommendations

As for suggestions for the development of the system is:

1. The use of other tools that can control the fan without using applications on the computer.
2. Adding a tool to control the rotary speed of the fan.
3. Making the system design more efficiently and ergonomic

REFERENCES

1. Disky (2015). Pengertian Deteksi. [ONLINE] Available at : http://news.palcomtech.com/wp-content/uploads/2012/01/Jurnal_Disky_ImplementasiHonepot.pdf. [Diakses 5 Agustus 2015]
2. Djuandi, F. (2011). Pengenalan Arduino. [ONLINE] Available ats : www.tobuku.com/docs/Arduino-Pengenalan.pdf. [Diakses 28 Februari 2015].
3. Immersa. (2014). Pengenalan Mikrokontroler. [ONLINE] Available at : http://www.immersa-lab.com/pengenalan-mikrokontroler.htm. [Diakses 29 Februari 2015]
4. Komponenelektronika. (2014). Sensor Suhu. [ONLINE] Available at : http://komponenelektronika.biz/sensor-suhu.html. [Diakses 5 Maret 2015].
5. Putra, L. Y. (2013). Perancangan Sistem Pengukur Suhu Menggunakan Arduino dan C#.Net. Jakarta: Universitas Mercu Buana.
6. Vcc2gnd. (2014). DHT22. [ONLINE] Available at : http://vcc2gnd.com/sku/DHT22. [Diakses 5 Maret 2015].
7. Zennifa, F. (2012). Perancangan dan Implementasi Pengontrol Suhu Ruangan dengan Menggunakan Sensor LM35 Berbasis Mikrokontroler Arduino Uno. Univesitas Andalas. Padang.

Monitoring and Controlling *Green House* Application
Case Study: Tomato Plant

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Abstract **- Green house is a building where the cultivation of the plant by setting several categories that are inside to fit the needs of growth and development of cultivated plants. Green house made of glass or plastic. Green house become heat because the electromagnetic radiation coming from the sun warms plants, soil and other items in it. In the cultivation of crops, farmers must control the temperature level of the planting area by regulating the lighting in the room. Farmers also have to control the moisture level of the soil to support plant growth processes in order to grow properly. This causes farmers to be located directly on the spot and can not leave the area of cultivation in a long time, for example when traveling out of town. Short Message Service (SMS) is a short message service which is one of the features or services from a mobile phone, which is used for exchanging messages with others via text message. One of the SMS technology developed at this time is the SMS Gateway. After going through the stage of implementation, this application has been through the testing phase and it is known that the application is built to facilitate farmers for controlling the room temperature and soil moisture levels even though farmers are not on the planting area.**

Keywords - Green house, Short Message Service (SMS), SMS Gateway

I. BACKGROUND

Green house is a building where the cultivation of the plant by setting several categories that are inside to fit the needs of growing and flower plants. *Green house* made of glass or plastic. *Green house* into heat because the electromagnetic radiation coming from the sun warms plants, soil and goods Other therein.

Tomato plants are one of the plants that can be grow in a *green house.* To get a good fruit of the tomato plant, farmers must keep the temperature level of the planting area between 24º C until 28º C. Room temperature control can be done by adjust the lighting in the planting area. Besides keeping the temperature level, farmers must control the moisture level of the soil to support the growth of tomato plants in order to grow properly. This causes farmers to be located directly on the spot and can not leave the area of cultivation in a long time, for example when traveling out of town.

Many farmers are using SMS services for supporting the activities of daily life. *Short Message Service* (SMS) is a short message service which is one of the features or services from a mobile phone, which is used for exchanging messages with others via text message. One of the SMS technology developed at this time is the SMS *Gateway.* According to Purnamasari (2010), SMS *Gateway* is an application that transforms the SMS from *Mobile Equipment* to a computer or laptop. SMS like features of cellular phone call, but there are different terms of features and functions that can be made based on business needs.

Based on the description above, there is an opportunity to build an application for controlling *green house* with tomato plants based on SMS *gateway* using a microcontroller that can facilitate farmers in controlling the rate at room temperature by adjusting the lighting in the room and control soil moisture levels by adjusting the watering of plants although farmers are not on the planting area.

II. LITERATURE REVIEW

A. Green house

In making this application I use *green house* as a case *study. Green house* (green house) is a building where plants are cultivated. (Freeman, 1998).

B. Temperature

Surya (2006) explains that the temperature is a measure of heat coldness of an object. Temperature is proportional to the average kinetic energy of particles. The greater the average kinetic energy of the particle, the higher the temperature.

C. Humidity

Wijaya (2008) explains that humidity is a symptom or circumstances indicate the amount of water vapor. Humidity is strongly influenced by moisture and sunlight intensity. As the symptoms of other abiotic, the humidity is also very important for keterlangsungan living things.

D. Tomato

The word 'tomato' is a loan word that comes from the Indonesian language Astek tribe (one of the Indian tribes). 'Tomato' is derived from the word 'xitomate' or 'xitotomate'. The information showed that tomato plants are not native to Indonesia (Pitojo 2005).

E. SMS Gateway

SMS *Gateway* is an application that transforms the SMS from the *Mobile Equipment* to a computer or laptop, SMS like features of cellular phone call, but there are differences in terms of features and functions that can be made based on business needs (Purnamasari 2010).

F. Electronic Cicuits

Electronic circuits are electrical circuits containing electronic components. For example, the detection system passing through the door, or even automatic plant watering system that works on the basis of the level of dryness growing media containing electronic circuit (kadir 2013)

G. Microcontroller

In making this application I use a microcontroller. According to Tooley (2008), the microcontroller is a small control devices *(micro)* that is designed specifically for applications of control and not for versatile applications

H. Arduino Uno

Kadir (2013) explains that, *arduino uno* is one of the products that are labeled arduino actually is an electronic board containing a microcontroller ATmega328 (a keeping that functionally acts like a computer

I. Ethernet Shield

Ethernet Shield is an NIC *(Network Interphase Card)* for *arduino* so that data can be sent or received from a network computer. This *shield* provides a *library* for developers that making it easier for programmers to create applications *of real-time monitoring* (Mandrani 2014).

J. Hypertext Preprocessor (PHP)

In doing programming on the *web,* the author uses the *Hypertext Preprocessor* or PHP programming language. PHP is a *script* that is used to create a dynamic *web* page and the page to be displayed will be made at the time the page is requested (Anhar 2010

K. Development methodology

Prototype model begins by gathering user requirements for the software to be created. Then it made a *prototype* so that the user has a view on the software to be built.

III. ANALYSIS

A. Preliminary Analysis

At this stage, conducted interviews and based on interviews with farmers and a lecturer in the faculty of agriculture, it is known that by using *green house* we can minimize the damage to the plant, because not all plants need direct light from the sun. Such as nurseries, some plants have to be on guard so as not exposed to direct sunlight. As for some specific standards in making *green house,* such as the size of the building should be large because in the *green house* are many tools or materials used in supporting the needs of the plant. They are: security (paranet) to avoid pests. Supply plenty of water and lighting in *green house,* the existence of a plant nursery, storage, processing a nursery, and planting soil after seeding.

From interviews can be concluded that by growing tomato plants in the *green house,* the farmer must maintain the existing level diruangan temperature by regulating the lighting in the room. Farmers also have to keep the soil moisture level by regulating the watering process of planting area.

Requirement Specification

1. Applications can control the room temperature by regulating the lighting in the room.
2. Applications can control the soil moisture level by regulating the process of watering the tomato plants.
3. Applications can automatically provide warnings about the values of temperature and soil moisture which has passed through the prescribed limit.
4. Users get information on how to use the application.
5. Applications can provide information about the state of *green house* such as how many lights are on or how much the value of temperature and humidity.
6. Applications can display the *history* data from the sensor.

.

B. User Analysis

Table 1. user analysis

No.	User	Task
1.	Farmer	Use and provide input to the application used.

C. Data and Communication Analysis

At this stage it will conduct an analysis of the data entered, output and data communications that occur in it. As for the output of data entered and received by applications and users as well as existing data communications can be viewed through the following process image.

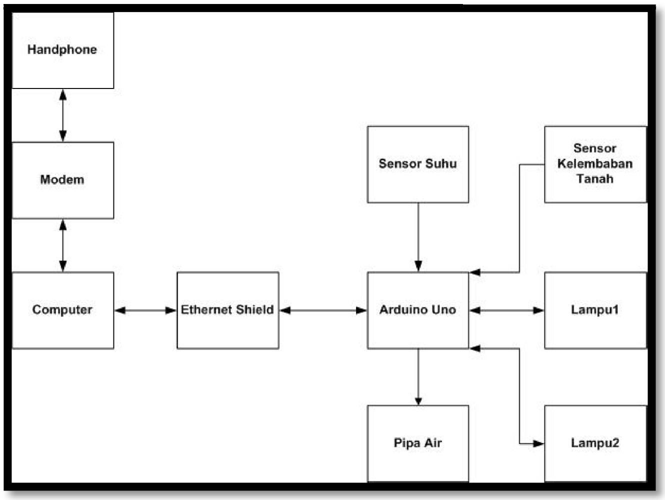

Figure 1. System work flow diagram

D. *Flowchart*

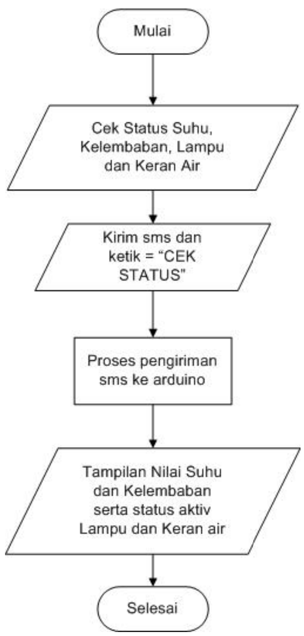

Figure 2. *Flowchart* check the status of temperature, soil moisture, and light

In the picture above *flowchart* diagram explaining the process to determine the state of temperature, humidity and light in the *green house.*

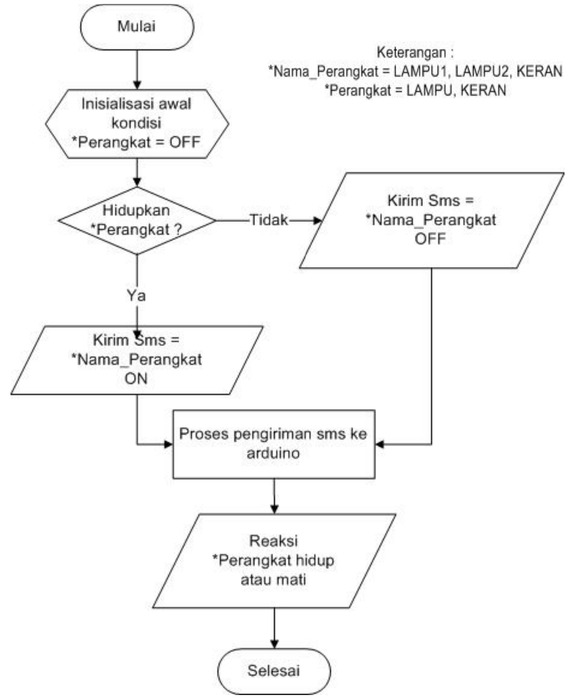

Figure 3. Lamp and Pipe *Flowchart*

IV. Design and Implementation

At the design stage hardware on electronic devices and software for applications to be built, an explanation can be seen in the points there.

The following is a block diagram of controlling *green house* application design based SMS *gateway.*

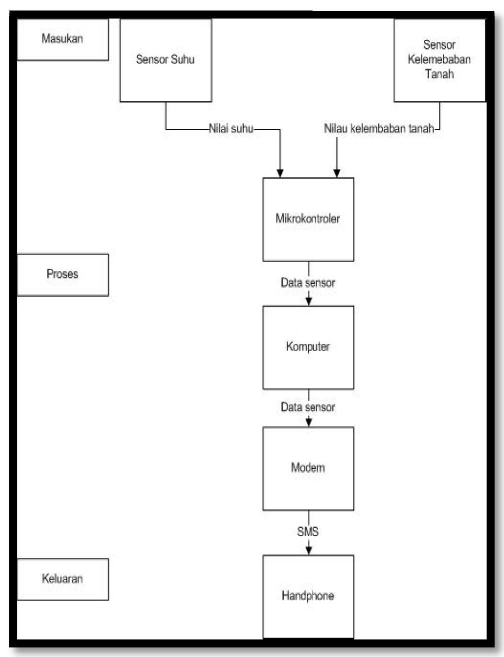

Figure 4. Notification Process

A. *Software implementation*

Software design objectives is to control temperature and humidity levels in the soil *green house* via *mobile phone* using SMS service. Software design is done for a microcontroller, display settings and *history* on*the web,* and SMS *gateway* using Gammu. In controlling *green house,* the instructions given via SMS, users will be entered into the *database* by gammu as a *gateway.* The incoming data will be sent to *the Ethernet shield* which is connected by *the router* and will be implemented in *Arduino.*

Similarly, in *Arduino* that has been mounted sensor. The sensor will take data from *the green house* and sent to the *database.* And if the data obtained have exceeded the specified user, then automatically the application will send the data to the specified mobile phone number. And to see the sensor data records in the *green house,* the user can see through *the web* interface, which is used for programming the SMS *gateway* and display on the *web* is *Notepad ++* programming language PHP. As for the microcontroller is *Arduino* IDE with C language programming language

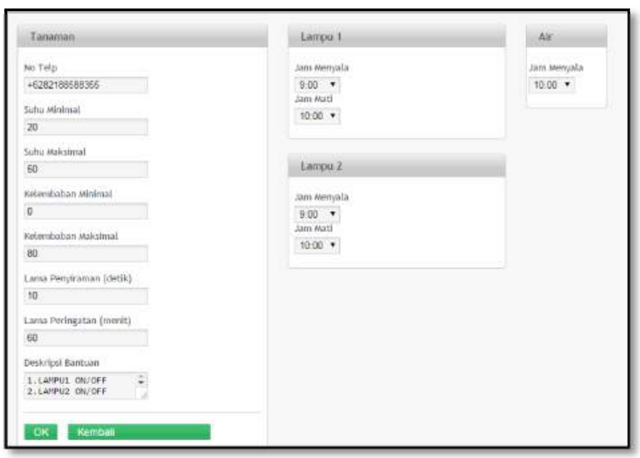

Figure 5. Application dashboard

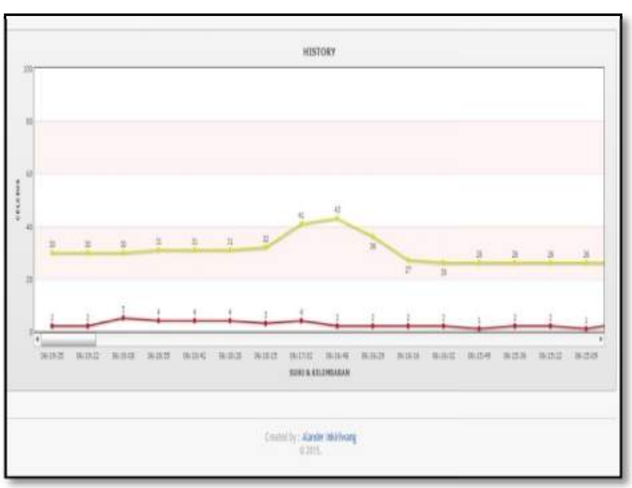

Figure 6. Data Monitoring

Figure 4.3 displays the room temperature recording implementation and control of soil moisture in *green house* application-based SMS *gateway.*Users can view a recording of temperature and soil moisture sensor in the *green house.* The yellow color shows a graph of temperature and red for soil moisture.

V. TESTING AND RESULTS

A. *Testing*

The purposes of testing in general are:

1. Ensure all the needs that have been registered in the analysis phase have been implemented.
2. Ensuring that data communication that exists between applications with microcontroller has been running as designed.
3. Can test whether an electronic device that wants to be controlled to function in accordance with what is expected.
4. Users can define the maximum and minimum value of temperature and soil moisture.
5. Destination phone number to receive alerts can be determined by the user.
6. Users can find out how the use of the application.
7. Applications can provide warnings about the value of temperature and soil moisture which has passed through the prescribed limit.
8. Users can determine the numbers of lamps are lit along with the value of temperature and soil moisture in the *green house.*
9. Applications can execute commands automatically.
10. Users can see the changes of temperature and soil moisture values in the *web interface.*

B. *Testing Criteria*

Here are the criteria in the test:

1. Applications can be executed when a *mobile phone* connected to the network.
2. All functions of the application can run as expected.
3. Powered by computer, *mobile phone,* microcontroller, *green ho use,* installation of water and electrical installations.

C. Testing case and its result

For the test case consists of a few cases, the result displayed on right column:

Table 2. Data Communication test and result

No.	Action	Result
1.	Command turn on the lights	Lights on
2.	Command turned off the lights	Lights off
3.	Command opens a water pipe	water pipes open
4.	The microcontroller's response to the SMS users	Users receive SMS newest state

Table 3. Notification test and result

No.	Action	Result
1.	Sensor over the limit values specified	system sends notification to thespecified *mobile phone* number
2.	Sensor berada pada nilai normal	The system does not send notifications

Table 4. Application respon (web based)

No.	Action	Result
1.	Displays progress graphs of temperature sensors.	Development graph temperature values displayed reasonably well
2.	Displays progress graphs soil moisture sensor.	The development chart humidity values displayed reasonably well
3.	Featuring a standard value application usage	The development chart humidity values displayed reasonably well
4.	Provide a phone number and the arrangement features a standard temperature and humidity	User managed to change the destination number and standard temperature and humidity
5.	Providing automation configuration system works	Users can change the default workflow automation system

VI. Conclusions and Suggestion

A. Conclusion

1. This study has built a system of monitoring and control on tomato plants that can run well.
2. The data communication between sensor, web applications and commands via sms can integrate well.
3. The historical data of web applications can help farmers in their tomato plants to monitor the situation from distance.

B. Suggestion

Here are suggestions for the future development of the system:
1. Undertake research to a wider area of the plant and determine the exact position of the sensor.
2. Adding a prediction system for every season as the recommendation for farmers.

References

Ainun, M. (2012), Utilization of organic liquid fertilizer on the growth and yield of some varieties of tomato *(Lycopersicum esculentum* L.), Thesis, UNSYIAH Banda Aceh, the Faculty of Agriculture.

Anhar. (2010). *Guide Mastering PHP and My S ql autodidact.*

Kadir, A. (2013). *A Practical Guide Learn the applications using the Arduino Microcontroller and programming.*

Canisius (2007). *The basics of farming.* Available di: https://books.google.co.id/books?id=te9kIRK5c6wC&pg=PA96&dq=kelembaban+adalah&hl=id&sa=X&ved=0CCMQ6AEwAmoVChMI74iDvr37xgIVBAuOCh0GxgiV#v=onepage&q=kelembaban%20adalah&f=false .[Diakses July 20, 2015]

Mark Freeman. (1998). *Gardening in you green house.* Available di: https://books.google.co.id/books?id=gwxtcwimz14C&printsec=frontcover&dq=greenhouse+mark+freeman&hl=en&sa=X&ved=0CCUQ6wEwAGoVChMI0sqG4YH1xgIV0giOCh1kmAYj#v=onepage&q=greenhouse%20mark%20freeman&f=false. [Accessed July 20, 2015]

Purnamasari, C., Simamora, S., Susanti, F (2010). *Implementation of SMS Gateway* In Providing Solutions Reports Students Course.

Rosa, the US and Saladin, M. (2013). *Software Engineering Structured and Object Oriented.*

Setijo, P. (2005). *Captive tomato seeds.* Available at: https://books.google.co.id/books?id=BO86wO2_BEEC&pg=PA9&dq=tanaman+tomat+adalah&hl=en&sa=X&ved=0CC0Q6AEwA2oVChMItsPSiff9xgIV0MSOCh3CkAQV#v=onepage&q=tanaman%20tomat%20adalah&f=false . [Accessed July 23, 2015]

Silfianti W. (2013). *Design of Information Systems.* Available at: http://wsilfi.staff.gunadarma.ac.id/. [Accessed July 19, 2015]

Sutarman. (2012). *Introductioto Information Technology.* Jakarta: Earth Literacy.

MIMO System Data Modeling Based on Adaptive Linear Neuron

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***Abstract* - Data modeling is a process of finding a model from a set of data in a process of the system , in this study will be carried out modeling of system data and dynamic multivariable known as Multi Input Multi Output (MIMO). MIMO data can be retrieved from a process in the system Waste heat boiler. Black box method requires a series of data that illustrates the process that occurs in the waste heat boiler system. Data needed in the form of input-output pairs of processes that occur. System modeling process consists of several stages, namely the collection of experimental data, the determination of the structure of the model, the selection method used and perform the validation of models that have been produced. One method used is the Adaptive Linear Neuron algorithm working with delta-rules algorithms. With the above method produced four pieces of model approaches of waste heat boiler process, the outcomes are the form of the models with the best performance besed on the AARE, NRMSE, NMBE respectively: 0.3434, 0.0042 and -0.2232.**

Keywords— Data Modeling, Waste Heat Boiler, Adaptive Linear Neuron

I. INTRODUCTION

Data modeling is a process of finding a model from a set of data in a process of the system, in this study will be carried out modeling of system data and dynamic multivariable known as Multi Input Multi Output (MIMO). MIMO types of data like this can be taken from a series of data on the processes in industry or large-scale plant. In this study will be used data from Waste Heat Boiler system. *Waste heat boiler* system is an important part of an industry, and relates also to the efficient performance of a mill. Utilization *boiler* functions effectively and efficiently needs to be done to save or optimize production costs. The process of calculating the performance of the boiler is very risky if done in a state of operation *(online)* to facilitate the calculation is usually done in the form of *offline* simulation, of course, by lowering or find *a model boiler* beforehand.

Waste heat boiler (WHB) is a complex and dynamic system, one approach to find a *model* of a complex system is to identify in a black box. Application of this method requires a series of black box data that reflects or gives an overview of how the performance or processes that occur on the system's *waste heat boiler,* usually require the data in the form of input-output pairs of processes that occur.

Process Modeling system has several steps that must be done to achieve the desired model of the system of data collection experiments, the determination of the structure of the model, the selection method used for the structure and validate the final part of the models that have been produced [4].

Adaline is one form of artificial neural network algorithm that uses *delta-rules* for renewing the weight [1] [2] [10] [13]. *Adaline* form is simple, making it easy and quick to apply [12] [16],*Adaline* work utilizing computing machines.

With the methods above, the process to obtain a model of *waste heat boiler* systems that are dynamic and nonlinear can be done with a shorter time.

Research on *waste heat boiler,* already been done by some previous research [5] [8] [11].

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II. ADAPTIVE LINEAR NEURON

Adaline was first developed by professor Bernard Widrow and Ted Hoff, his colleagues at Stanford University in 1960. Adaline a single layer network with multiple input neurons are connected directly to an output neuron and a bias. As shown in Fig. 1.

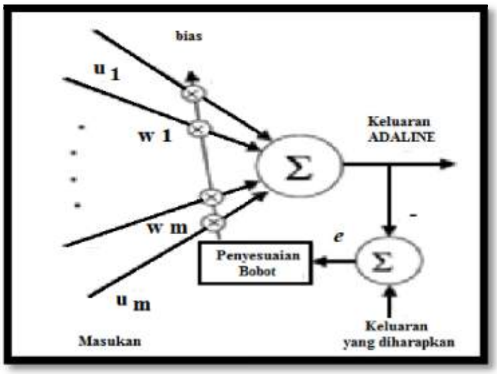

Fig. 1. Structure of one unit adaline model
(Zhang, 2006)

Weights modified with delta rule (often called the *least mean square).* During the training, the activation function used is a function of identity, (f (x) = x).

$$net = \sum_i x_i w_i + b \qquad (1)$$

$$y = f(net) = net = \sum_i x_i w_i + b \qquad (2)$$

Errors obtained from the squared difference between the target (t) and network output (f (net)). In the delta rule, the weight is modified such that the minimum mistakes. Function criterion that is often used is the sum of squared errors smallest average *(Least Mean Square* or LMS) that is expressed by:

$$E = \frac{1}{2}\sum_i (target - y)^2 \tag{3}$$

III. DESIGN AND ANALYSIS

This research will be carried out modeling of the data of a system that has multiple input and multiple output (MIMO), in order to obtain a model of the system. As for the system to be modeled is the *Waste Heat Boiler (WHB)*

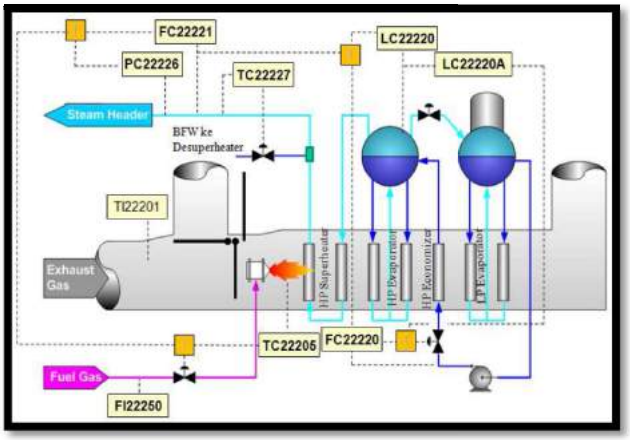

Fig. 2. Preocess diagram of *waste heat boiler*

With the state of the system above was taken measurement points with the input point as much as four points, namely: *fuel gas* (the rate of *fuel), exhaust gas* (exhaust gas), *boil feed water* (rate BFW to *HP drum), desuperheater* (rate BFW desuperheater) *and* as the output is taken five points, namely: the vapor pressure, the rate of steam, steam temperature, level *HP drum* and combustion chamber temperatures. The data is taken from historical data of plant operation as many as 50,000 records, with the time period of each of data a second, a total of nine points measuring point.

A. Model Selection

Modeling the many variables that affect the *multivariable* ARX WHB then selected, in this case the model equations are formulated with four inputs and five outputs with the general equation as follows:

$$y(k) + A_1 y(k-1) = B_1 u(k-1) + e(k) \tag{4}$$

Formulation four inputs and five outputs:

$$y\,(k) + a1\mathbf{y1}(k-1) + a2\mathbf{y2}(k-1) + a3\mathbf{y3}(k-1) + a4\mathbf{y4}(k-1) + a5\mathbf{y5}(k-1) = b1\mathbf{u1}(k-1) + b2\mathbf{u2}(k-1) + b3\mathbf{u3}(k-1) + b4\mathbf{u4}(k-1) \tag{5}$$

Thus,

$$y\,(k) \;- a1\,\mathbf{y1}(k-1) + -a2\,\mathbf{y2}(k-1) - a3\,\mathbf{y3}(k-1) - a4\,\mathbf{y4}(k-1) - a5\,\mathbf{y5}(k-1) + b1\,\mathbf{u1}(k-1) + b2\,\mathbf{u2}(k-1) + b3\,\mathbf{u3}(k-1) + b4\,\mathbf{u4}(k-1) \tag{6}$$

where A_I and B_I is a matrix that contains the parameters to be estimated, A_I with size [5 x 5] and B_I with size [5 x 4], so that the total estimated parameters size [5 x 9].

Then *multivariable* ARX structure is brought into forms that make up *Adaptive Linear Neuron* five output neurons states, the number of inputs to the neurons respectively are nine pieces can be seen in Fig. 2.

B. Model Estimation

The estimation process is done by using algorithms *Adaptive Linear Neuron*. The system is assumed to order one, with neuron model as shown in Fig. 3.

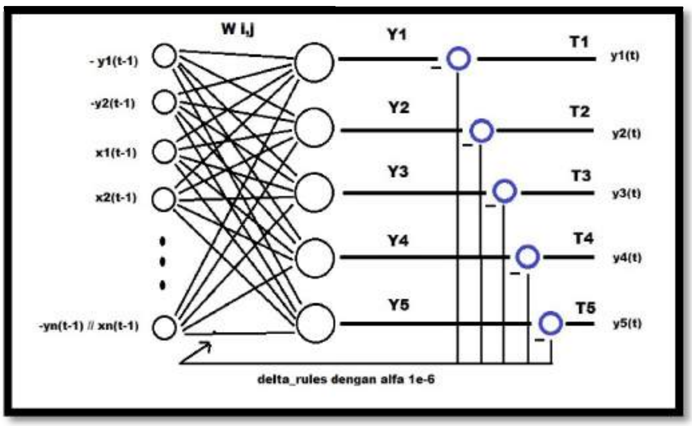

Fig. 3. *Adaptive linear neuron* WHB

With a common form *Adaline* neuron equation:

$$\mathbf{y} \;=\; \sum_{i=1}^{9} w_i\, x_i \;=\; \boldsymbol{x}^{\boldsymbol{T}}\,\boldsymbol{w} \tag{7}$$

Where, x is input *adaline* input and w is weight, in this case a pair of data input and output *(u, y).*

$$x = [y_1\ y_2\ y_3\ y_4\ y_5\ u_1\ u_2\ u_3\ u_4]$$

$$w = [w_1\ w_2\ w_3\ w_4\ w_5\ w_6\ w_7\ w_8\ w_9]^T$$

so that the equation becomes:

$$y = w1\,y1 + w2\,y2 + w3\,y3 + w4\,y4 + w5\,y5 w6\,u1 + w7\,u2 + w8\,u3 + w9\,u4 \quad (8)$$

Then with regard (8) and (6), then the equation for the fifth output *Adaline*, can be simplified into:

$$yn(k) = -wn,1\,y1(k-1) + -wn,2\,y2(k-1) + -wn,3\,y3(k-1) + -wn,4\,y4(k-1) + -wn,5\,y5(k-1) + wn,6\,u1(k-1) + wn,7\,u2(k-1) + wn,8\,u3(k-1) + wn,9\,u4(k-1) \quad (9)$$

Then, there is the renewal of all weights are performed using the renewal mechanism *Widrow–hoff weight* with a new equation becomes:

$$w_i\,baru_{\,n} = w_i\,lama_n + \eta(target - \hat{y})_n\,x_{\,i} \quad (10)$$

Where : *n* symbolizes *n-th* output neurons.

Thus the output of *Adaline* neurons in the present study, has five outputs, where the form was developed from Adaline standard form that has only one output. Thus related to the learning process using *Widrow-hoff,* mechanisms, then use the value of η for all output neurons. So as to determine the value of MSE learning to use the equation:

$$MSE = \frac{1}{n}\sum_{i=1}^{n} y_{\,i} \quad (11)$$

This situation reflects that the WHB system has output interrelated with each other, according to circumstances on the real plan.

IV. SIMLATION RESULTS AND ANALYSIS

In this study, the learning process is performed on a data set consisting of 50,000 input-output data, which is done in two stages. In the first study conducted on the 1 s / d 40,000, with 350,000 iterations *epoch.* The *epoch* of 350,000, have been four sets of weights at a certain *epoch* to *epoch-produce* four models estimated. In the second stage, the four models that has been obtained in the first stage is validated with 10,000 data that is not included in the learning process at first stage*(fresh data).* Then the model will be seen the capabilities and characteristics of each, the next of the four models will be selected Where the result is set to be the best for the final model *of waste heat boiler system.* Here are the results and analysis of research conducted

A. The Result of the Model

The following are the result of the models base on Adaline algorithm process :

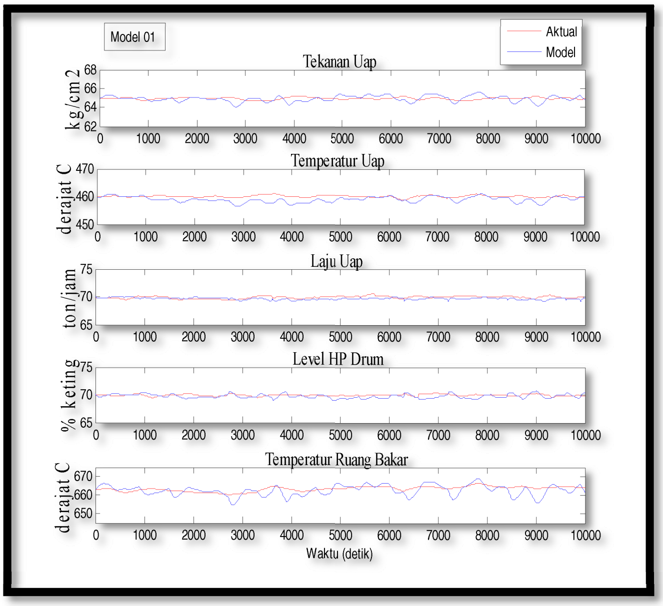

Fig. 4. Model 01 testing

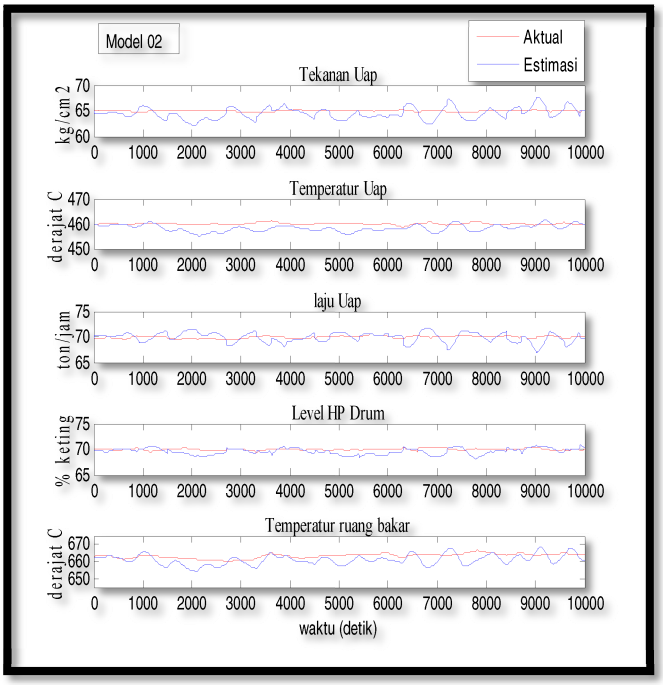

Fig. 5. Model 02 testing

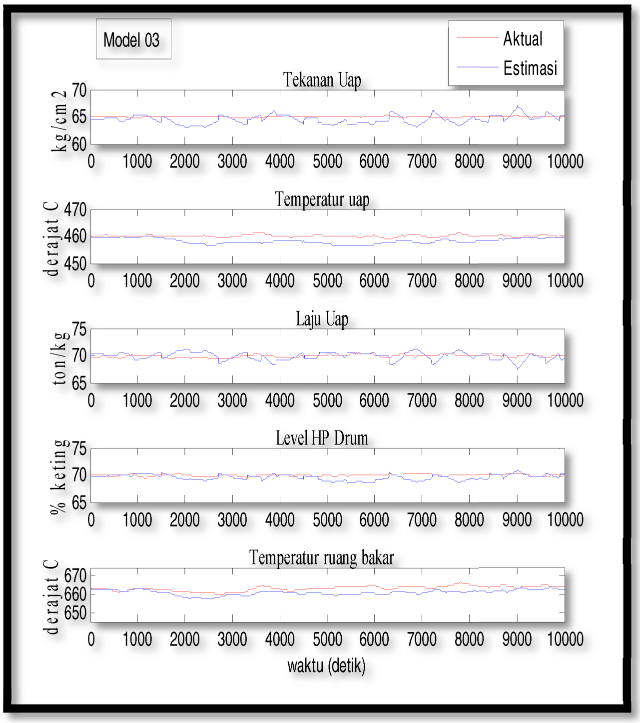

Fig. 6. Model 03 testing

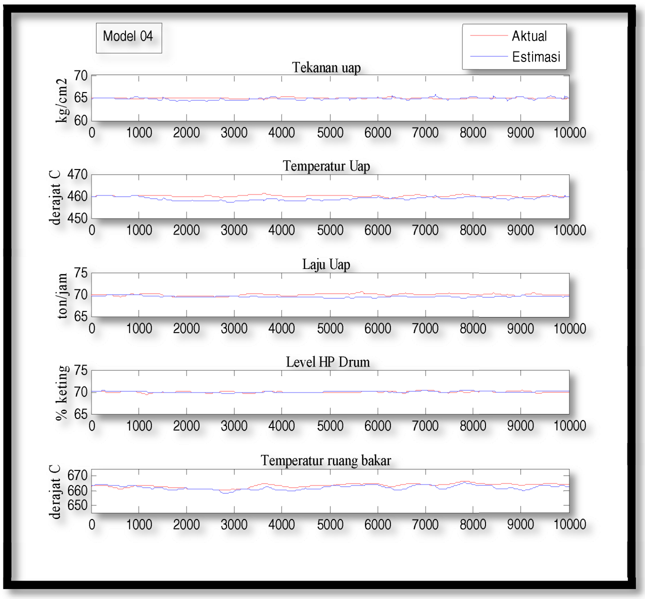

Fig. 7. Model 04 testing

B. *Model Results and Comparisons*

The following will be displayed the comparisons of the four models based on : AARE, NRMSE, and NMBE [9]. Then from the four models have their merits, will choose one model that has the smallest error, and better according to the state of the real system WHB in the field.

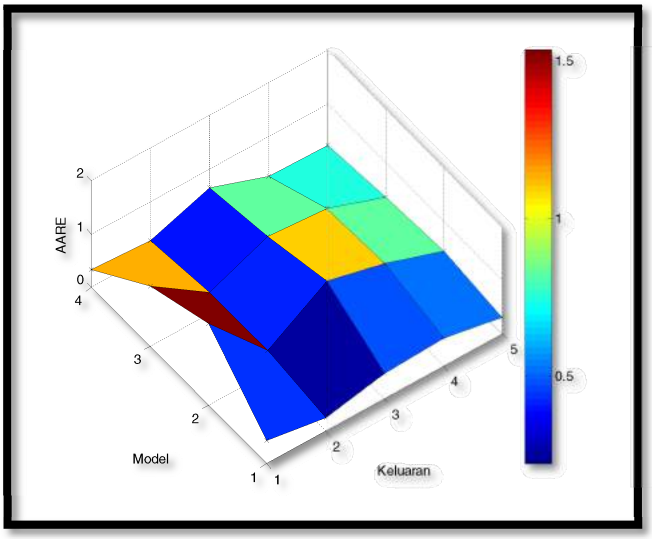

Fig. 8. AARE Comparison

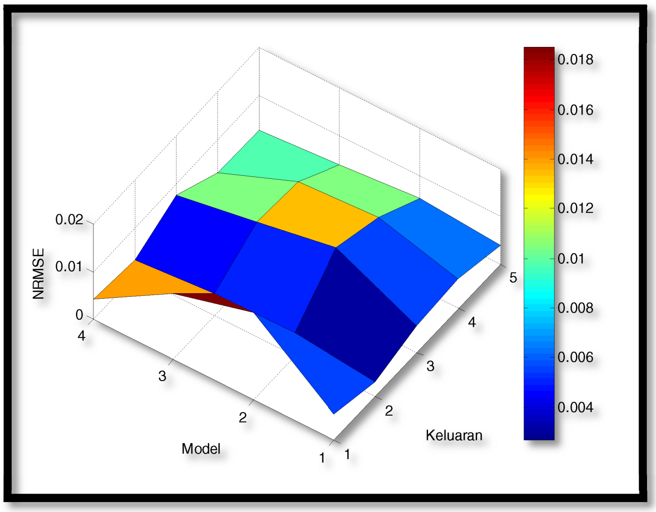

Fig. 9. NRMSE Comparison

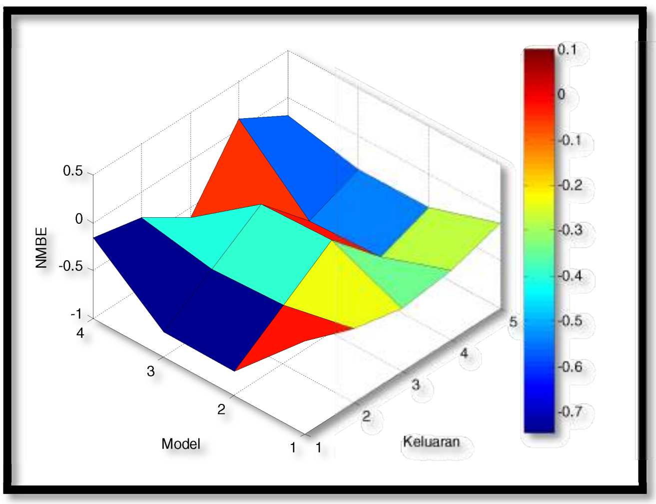

Fig. 10. NMBE Comparison

From three types of comparison above shows the model 01 and the model 04 has better performance than both of the model 02 and the model 03. If the average is taken of each model to be compared, it can be seen in Table IV.1, that the model 04 have a lower value then the other models.

TABLE I. MODEL 01 – 04 PERFORMANCES

MODEL	AARE	NRMSE	NMBE
Model 01	0,3988	0,004874	-0,19702
Model 02	0,8641	0,010603	-0,40841
Model 03	0,6946	0,008535	-0,42328
Model 04	0,3435	0,004161	-0,22322

With the error calculation shown above, all models have a good performance, but in this section we will choose which one will be represent from four models, so the model 04 is considered the most representative to reflect the performance of the system WHB in the field based on the small AARE and NRMSEl, with NMBE value is -0.2232 which is also smaller than other models.

V. CONCLUTION

The research that has been done can be drawn some of the conclusions:

The learning process can be done by *Adaline* and can recognize the pattern of input-output data are given, so that the weight of learning outcomes can be used to get 4 pieces model of the system, with the results of the four models based on AARE NRMSE and NMBE which has a validation value as shown in Table I ,

The model 04 is a model that has the smallest error value compared with other models.

VI. FUTURE WORKS

From this research, it can be suggested some future works as follows:

Build a model with WHB transient data sets, which can be seen in the response time of the system start up.

The network structure can be developed by using Generalized Adaline to accelerate convergence when identifying and modeling in online modeling process.

VII. REFERANCES

[1] Widrow B. dan Hoff, ME Jr., (1960). "*Adaptive Switching Circuits*," IRE WESCON Conv. Rec., Part 4, pp. 96–104.

[2] Widrow, B.(1966). "*Adaptive Filters I: Fundamentals*," Rep. SEL-66-126 (TR-6764-6), Stanford Electronic Laboratories, Stanford, CA

[3] Paulraj Ponniah. (2007): '*Data Modelling Fundamental*", John wiley & sons New Jersey, wiley - interscience International, Inc.

[4] Soderstrom, T., dan Stoica, P.,(1989) , '*Sistem Identification*", Prentice-Hall international series in sistem and control engineering.

[5] Rismayasari D., Joelianto E., dan Chaerani D.,(2009),*" The Implementation of Robust-Optimization-Based Model Predictive Control to Waste Heat Boiler",* International Conference on Instrumentation, Control & Automation, ICA2009, Bandung Institute of Technology, Indonesia

[6] Joelianto, E., Widiyantoro, S., Ichsan, M., (2009): *Time Series Estimation on Earthquake Events using ANFIS with Mapping Function. International Journal of Artificial Intelligence*, Vol. 3, Number A09

[7] Joelianto, E., Rahmat, B., (2008): *Adaptive Neuro Fuzzy Inference Sistem (ANFIS) with Error Backpropagation Algorithm using Mapping Function. International Journal of Artificial Intelligence*, Vol. 1, Number A08

[8] Hamidah, NL. dkk (2010),." Analisa Kinerja W*aste Heat Boiler* dengan Metode Kesetimbangan Panas dan Massa*", ITS undergraduate paper* , Teknik Fsika FTI – ITS.

[9] Jothiprakash, H., Magar, R., Sunil, K., (2009), **"***Rainfall–Runoff Models Using Adaptive Neuro-Fuzzy Inference System (ANFIS) For an Intermittent River", International Journal Artificial Intelligence* Vol. 3,1-33.

[10] Zhang, W.,(2006)," *Sistem Identification Based on Generalized Adaline Neural Network",* International Journal of Intelligence Control and Systems, vol. 11, no. 1.

[11] Akbar, W.A., dkk,(2010)," Perancangan sistem pengendalian pembakaran pada *duck burner* WHB berbasis *logic solver",* ITS undergraduate paper , Teknik fisika FTI – ITS.

[12] Francoise, B., dan Widrow, B. (1993)," *Simple Algoriths for Fast Adative Filtering",* journal WNN'93, 231-236.

Neural Networks, vol. 16, no. 5–6, pp. 735–744, June–July 2003
[13] Haykin, S. dan Widrow, B. (2003), '*Least-Mean-Square Adaptive Filters*', John Wiley & Sons Inc.
[14] Padang, E. (2010),"*Model Prediksi Gempa Bumi Zona Tumbukan Maluku Berbasis Anfis Termodifikasi*",Tesis,Instrumentasi dan Kontrol, Institut Teknologi Bandung.
[15] Kecman, V.,(2001), *'Learning and soft computing: support vector machine,neural network,and fuzzy logic model*", Massachusetts Instutute of Technology, A Bradford book,.
[16] Widrow, B. dan Rumelhart, D., (1994)." *Neural network Application in industry, Busines, and Science",* communication of the ACM, vol. 37, no3., pp. 93–105, march 1994
[17] Walsh, P., NCE (2000), "*Power Spectral Density Measurement Techniques*" Paradyne, Broadband Access Solutions,.
[18] Widrow, B., dan Kamenetsky, M, , "Statistical efficiency of adaptive algorithms,"

Platform Displacement Impact on Mobile WiMAX Handover Performance over HAPS Cannel

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***Abstract*—High Altitude Platform Station (HAPS) is one of a new infrastructure in telecommunication, which is like airship or aircraft that is placed on the stratosphere. The distance of platform is about 17-22 km from the earth's surface. However, in its application HAPS that placed on the stratosphere layer of air is affected by the wind gusting. Position HAPS can change from the normal state that is called Platform Displacement. In this research discuss about platform displacement impact that experienced by the mobile station (MS) during the handover (HO) process. User is mobile WiMAX user who is using VoIP communications services. Performance of the system is calculated by some parameters value, there are throughput, data dropped, and HO (handover) delay. System will be simulated using simulator OPNET Modeler 14.5. In the simulation results can be seen the influence of platform displacement and speed of the user. When platform moves to the higher altitude will increase the area of cells coverage so that each cell can interfere its neighbor cells, especially in the transition region where MS do HO process. This condition can disturb HO process and increase data dropped during HO process.**

***Keywords*—HAPS displacement, MS velocity, throughput, data dropped, delay HO.**

I. Introduction

High Altitude Platform Station (HAPS) is a novel wireless infrastructure. It could be the solution for the problem that is faced by an existing wireless infrastructure [1]. Expensive operating cost, capacity, coverage, power consumption, and link budget, are the most common problems in terrestrial tower-based and satellite system. HAPS is proposed to overcome those common problem and at the same time is expected to be the complimentary wireless system. HAPS is placed in the stratospheric layer about 17 – 22 km above the ground [2]. By this position, HAPS will remain stay at their original position. Wind is expected to be calm and therefore HAPS displacement is not significant. In general, the movement of HAPs can be divided into three kinds. Those are vertical movement (up and down), horizontal (sideways) and tilt (inclination). HAPS displacement will cause adverse effects to the system performance [3].

On the other hand, we are now witnessing the progress of broadband technologies development. WiMAX is one of the kinds. WiMAX is able to provide many kinds of services such as VoIP, data, and multimedia services. However, WiMAX is initially developed for terrestrial tower-based wireless system. The system specification in WiMAX was designed to comply with the channel characteristic for such a system. In case of HAPS, WiMAX system specification needs to be re-evaluated again given that HAPS channel and geometry is not the same with terrestrial tower-based system.

One of many emerging problems for WiMAX to be deployed in HAPS system is become our concern. Handover performance of WiMAX's user inside HAPS system is evaluated in this contribution. There are some studies related to HO of the user in WiMAX system [4]-[6]. Additionally, we investigated effect of platform displacement to the performance of HO [3]. Due to HAPS affected by natural conditions, one of which is the movement of the wind, it can change the position of HAPS. The position's change will affect the performance of the system. In the simulation we use OPNET Modeler 14.5. This simulator gives a more full featured and easy to set the parameters have to be simulated, such as the parameters of HAPS, WiMAX technology and HO parameters as well as the MS movement trajectory. In the simulation we observe and analyze the platform displacement impact for the performance of HO WiMAX systems. System used is a WiMAX mobile communication system are conditioned in a state where the MS moves with a certain speed and pass some cells resulting in handover.

The rest of the paper is as follows. In Section II, we explain system model and architecture of HAPS and WiMAX. Section III presents simulation result and discussion. Finally, we conclude our result in Section IV.

II. System Model

HAPS can serve a large coverage and large capacity, capable to deliver services to both areas with dense populations (urban), medium (suburban), or rarely (rural) and even to remote areas (remote). Another advantage of the opportunities for HAPS is increased performance telecommunications networks that quality telecommunications services will go up, by combining terrestrial advantages in terms of capacity as well as advantages in terms of the ability of satellite coverage.

The most common problem of transmitting high-speed data such as in WiMAX system is a multipath

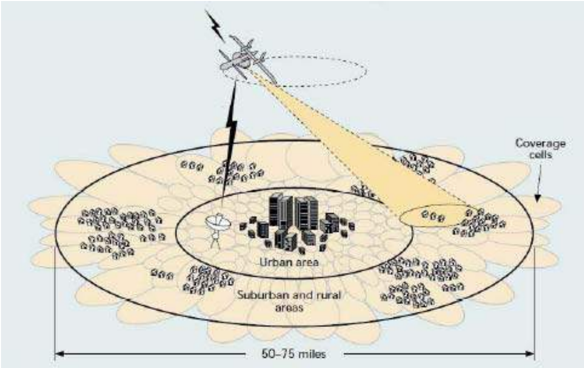

Fig. 1 HAPS system model for WiMAX application.

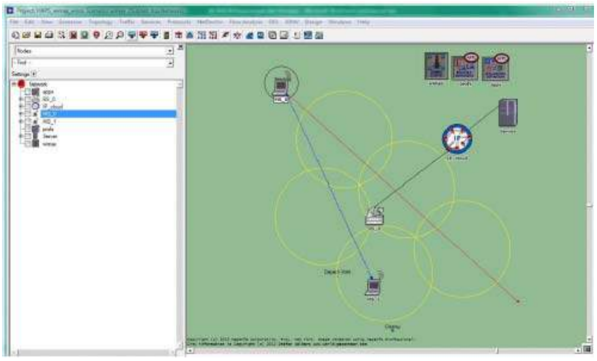

Fig. 2 Topology of the ground segment developed using OPNET Modeler 14.5.

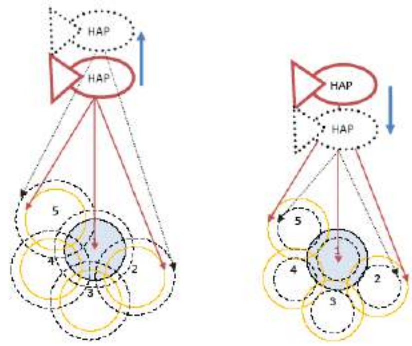

Fig. 3 Vertical movent of HAPS.

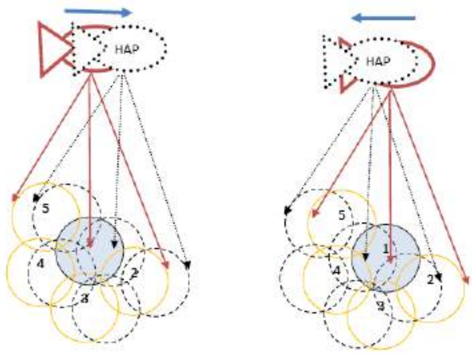

Fig. 4 Horizontal movent of HAPS.

propagation. There are many works have been done to characterize multipath behaviour of the satellite or terrestrial wireless channel. When WiMAX is going to be delivered through HAPs channel, the characterization of its channel should be firstly evaluated. In the case of HAPs, mobile users which are located within the coverage will have their own elevation angle to look at the HAPs. Therefore, they may have a LOS or NLOS situation relative to HAPs position depending on their elevation angle. Consequently, the contribution of multipath problem will vary from case to case with respect to mobile user position relative to the HAPs. In this contribution multipath behaviour experienced by HAPs communication channel is described using an experimental data.

Fig. 1 shows the system model of WiMAX's user communicating through HAPS system. HAPS is assumed to have many small beams inside the coverage. The size of each beam is not the same. In the nadir, size of beam is samaller than edge beam. This model is designed to approach the realistic model of HAPS antenna beam. In general, HAPS network modeling in this paper is divided into several parts. Those are MS segment (WiMAX MS device), ground segment (server, internet), and the segment sky (HAPS network).

In the case of HAPS encounters displacement due to the gravitational effect and wind, in Fig. 2 we show the model that is developed in the OPNET simulation environment. In the figure we show the topology of the ground segment to be simulated. The simulation will be given an observation area consisting of 5 cells, with each cell diameter is 7 km. The cell is placed right below the platform so that the cells formed are cell nadir and tier-1 cells. As seen in the figure, cell number 1 is a temporary nadir while cell number 2, cell number 3, cell number 4 and cell number 5 is a tier-1 cells.

Handover process scenario is designed to put the MS observations outside the cell, while the mobility of the serving cell and the target cell based mobility trajectory or path specified. MS will pass 2 target cells for the purpose of the creation process of handover. The simulation scenario is described in two parts, first MS movement modeling and platform movement modeling.

MS movement across 3 cells so MS will have 2 times HO process. MS moves to follow the trajectory. The movement of MS at the time of the platform has not experienced a change in position (displacement). In this condition, the MS trajectory crosses right in the middle of the cell. For the study, will be scripted MS moving at 50 Km/h, 80 km/h, 120 km/h and 200 Km/h. Movement modeling of the platform in this research is the vertical movement that consists of up and down, next to the horizontal movement of the platform shift to the west and to the east.

Vertical movement as described in Fig. 3 is the movement in which the position of HAPs move up or down. Scenario is simulated by changing the height of HAPS. Based on ITU, vertical movement of HAPS allowed is ± 700 meters far. Movement of the platform is simulated for a 700 meter vertical movement up or down. In this simulation, the height of HAPS before displacement is 20 km at the time of HAPS move further upwards, a position HAPs HAPs 20.7 km while moving downward, the position of HAPs to 19.3 km.

Horizontal movement is a movement in which the position of HAPS shifted to west and east. Scenario is simulated by changing the position of HAPS to the west

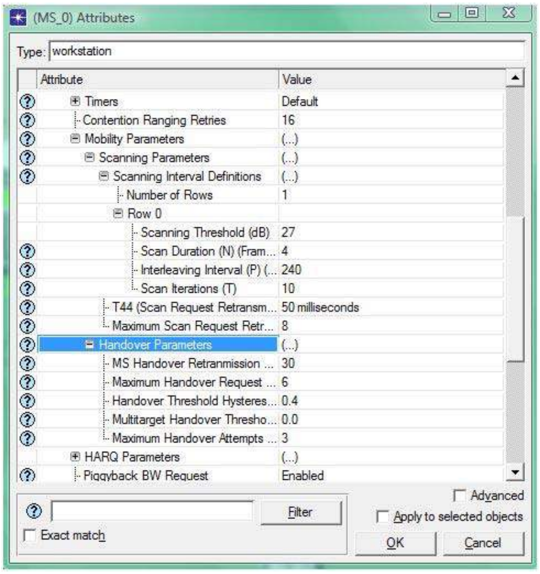

Fig. 3 WiMAX Attributes.

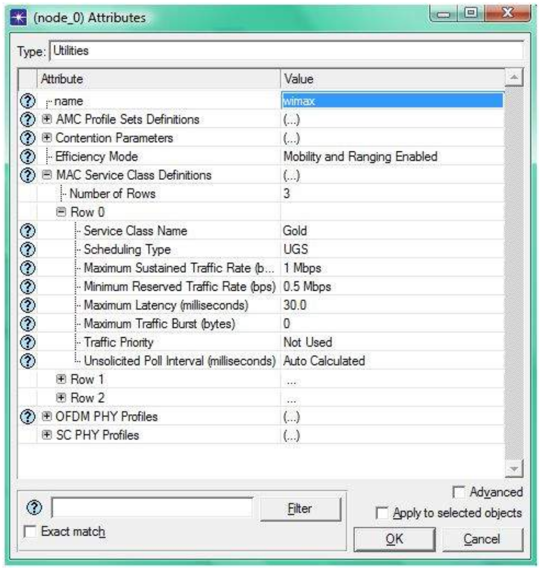

Fig. 4 Profiles Attributes.

or east manually by changing the position coordinates of HAPs. Based on ITU, horizontal movement of HAPS allowed is ± 400 meters.

Movement of the platform is simulated for the horizontal movement of 400 meters to the west or to the east. In this simulation, a height of 20 km HAPS is assumed to be fixed, but the position of HAPs only shifted course to the west or to the east. This is called as a HAPS horizontal displacement. It is depicted in Fig. 4. The beam is shifted west and east but the size and shape are constant.

As was explained earlier that the HAPS platform replacement for BS is placed in the stratosphere. In this simulation, the platform has a multi-antenna, each antenna generating the cell with a diameter of 7 km. On condition that the platform shifted to the west or to the east simulated by changing the position of the antenna 400 meters to the west or to the east. In addition, there are several parameters that need to be adjusted about frequency band, transmitter power and antenna gain. Frequencies used for communication HAPS, 3.5 GHz, that frequency is allocated for terrestrial communications, as big as 47 dBi antenna gain and transmitter power of 5 Watts.

IEEE 802.16e system parameters are set via the existing components or nodes used. Not all of the parameters available in the simulation so that additional parameters need to be made in the network to covering the services to be provided by the IEEE 802.16e system, types of service classes, and traffic services. The main components of general network setting are WiMAX attributes and profiles attributes, which in this paper the value of attributes are described in Fig. 3 and 4 respectively.

Some parameters for HO process is shown in Figure 4. Scanning threshold value is 27 dB, which means that MS will start the process of scanning and sending a message MOB-SCN-REQ when it detects signal strength serving sel 27 dB. Next up is the HO threshold hysteresis value is 0.4 dB, which means that MS will make a decision when the value HO hysteresis (difference in signal strength serving sel and target sel) 0.4 dB.

III. Simulation Result

For mobile users who use the service, HO is an important part of sustainability communications, such as HO procedure that runs too slow to interfere with the transmission of data services that are being served. As has been explained previously that MS moves across 3 cells and do 2 times of HO. Thus, there are 2 times for measurement the value of throughput, data dropped, and delay HO for 2 speed scenario, for medium-speed vehicles which is 50 km/h and 80 km/h and to high vehicular speed are 120 km/h and 200 km/h.

A. Throughput Simulation Result

In this paper, throughput are all data traffic (bits / second) were successfully received and forwarded to the above by the WiMAX MAC layer. Throughput simulation results can be seen in Table I. If we see, the impact of platform displacement changes the value of throughput randomly. This means that HAPS displacement does not significantly affect for throughput rate changes. This means that platform displacement does not give any impact to the signal strength and the error rate.

If we analyze from speed changes, when MS moves medium vehicular speed in 50 km/h and 80 km/h, the connection still in stability for MS to access data, so the resulting throughput is still higher than the MS that moves with high vehicular speed. Throughput on high vehicular speed down significantly, because MS decreases of gain (signal distortion).

B. Data Dropped Simulation Result

In this research, the value of data dropped is calculated from how much data traffic in the high layer drop by the WiMAX MAC MS that caused the buffer is overflow.

TABLE I THROUGHPUT SIMULATION RESULT.

Platform Movement	MS Velocity (Km/jam)	Throughput HO1 (bits/second)	Throughput HO_2 (bits/second)
West	50	49.400	24.600
	80	16.400	22.000
	120	100	100
	200	74	72
East	50	43.000	49.100
	80	18.400	8.750
	120	98	100
	200	74	74
Up	50	36.000	35.640
	80	18.287	9.200
	120	100	120
	200	74	73
Dowm	50	41.500	34.000
	80	13.900	18.700
	120	100	100
	200	71	73

TABLE II DATA DROPPED SIMULATION RESULT.

Platform Movement	MS Velocity (Km/jam)	Data Dropped HO_1 (bps)	Data Dropped HO_2 (bps)
West	50	150	150
	80	260	260
	120	360	360
	200	600	400
East	50	160	160
	80	240	240
	120	400	300
	200	600	600
Up	50	150	150
	80	17.500	240
	120	500	400
	200	600	600
Down	50	150	150
	80	240	160
	120	450	340
	200	600	600

The amount of the supplied buffer is 64 KB. In the Table II, data dropped value is the highest among all the existing value is when the platform moves up and MS moves with a speed of 80 Km/h. Based on the table, when the position HAPS moves up and user moves with 80 km/h has the highest value of the data dropped. This is due to the possibility of interference from other cells during the process of HO.

When HAPS moves up, each coverage of each cell will be projected larger than normal cells. In this case the value of SNR has possibility to interfere other cells. in this case, MS detects neighbor cell has same level of SNR as criteria HO so MS detect yhe other cell as target cell and establish a connection with the exchange of information during HO process. This causes MS will receive data sent from two cells at the same time, from other cell and target cells. Therefore, MS received much data and there are possibility errors of received data then MS will drop it.

Based on Table II it can be seen that when HAPS moved up and user moved on 80 km/h, the value of the data dropped is the highest one. This is due to the possibility of interference from other cells during the process of HO. At the time of HAPS move up, then the coverage of each cell will be projected larger than normal cells. This not only happens when the MS moves at a speed of 80 Km/h and during the process of HO-1, but can also occur in the HO-2 or even both of them. During retrieval of simulation results, when MS moves with a speed of 120 Km/h, MS also has the possibility has same condition.

C. HO Delay Simulation Result

HO delay is calculated from the time when MS sends MOB_MSHO-REQ message that aims to begin the process of intial ranging stages of HO until the MS is ready to connect to the destination cell. During HO process, time is an important process that is also taken into account. To support real-time services, eg VoIP applications, HO delay allowed is less than 50 ms.

HO delay calculation has been summarized into Table III. It can be seen that, in 4 different positions HAPS, almost HO delay is measured still allowed for VoIP service's.

TABLE III HO DELAY SIMULATION RESULT.

Platform Movement	MS Velocity (Km/jam)	Delay HO_1 (second)	Delay HO_2 (second)
West	50	0.035	0.035
	80	0.035	0.03
	120	0.03	0.03
	200	0.03	0.03
East	50	0.03	0.03
	80	0.035	0.03
	120	0.03	0.25
	200	0.03	0.03
Up	50	0.03	0.03
	80	-	0.025
	120	0.035	0.035
	200	0.03	0.03
Down	50	0.03	0.03
	80	0.025	0.03
	120	0.025	0.025
	200	0.03	0.03

HO delay. Except when the position of platform moved up and MS moved on speed of 80 km/hour. It can be seen that, the value of the HO delay is not detected. This is due to an error that occurred during the HO process, the interference from other cells.

This error resulted in the calculation of the time required to perform the HO MS is not detected by the system. This not only happens when the MS moves at a speed of 80 km/h and during the process of HO-1 only, but can also occur in the HO-2 or even both of them and also not only on the value of the velocity alone. During retrieval of simulation results, when MS moves with a speed of 120 km/h, MS also has the possibility to get HO process failure.

IV. CONCLUSION

We have evaluated and simulated HAPS system to determine the effect of HAPS displacement for WiMAX's user during HO process. User moves with several speed such as 50 km/h, 80 km/h, 120 km/h and 200 km/h and cross 2 target cells so that there will be 2 times of HO process. The results presented are the results of some simulation process and we take the average value to show there.

Of the research found that the platform displacement have an impact on the performance of HO, especially in data dropped and handover delay. When MS moves with 80 km/h and platform position changes with altitude 20.7 km, MS detect any interference during HO process, so that increasing the value of data dropped and the HO process is also interrupted.

REFERENCES

[1] Iskandar and S. Shimamoto, "On the downlink performance of stratospheric platform mobile communications channel," in Proceedings IEEE Global Telecommunication Conference (GLOBECOM), ISSN: 1930-529X, DOI: 10.1109/GLOCOM.2006.922, pp. 1-5, 2006.

[2] Iskandar and S. Shimamoto, "Analysis of CDMA capacity for multiple stratospheric platform mobile communications under imperfect power control and fading," in Proceedings IEEE Wireless Communications and Networking Conference (WCNC), ISSN: 1525-3511, DOI: 10.1109/WCNC.2007.468, pp. 2512 - 2516 , 2007.

[3] Iskandar and K.Z. Arief, "Effect of HAPS Movement on the Performance of Downlink Power Control CDMA System," in Proceedings of IEEE 10th International Conference on Information Science, Signal Processing, and Their Applications (ISSPA 2010), DOI 10.1109/ISSPA.2010.5605580, pp. 682-685, May 2010.

[4] Al-Saedi, F.A. Thweny, and W.A. Maddallah. *Evaluation of Handover Process in WiMAX Networks*, 2012.

[5] A. Bhaskar, D. Eyers, and Z. Huang, "Handover Delay in Mobile WiMAX: a Simulation Study," in Proceeding of International Conference on Parallel and Distributed Computing, 2011.

[6] Janevski and Toni. Mobility Sensitive Algorithm for Vertical Handovers from WiMAX to WLAN, 20th Telecommunications forum Telfor. 2010.

Polygon WebGIS of Distric Level for Development and Monitoring of PUSKESMAS in Health Care Services

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***Abstract*—In the future, Community Health Center (PUSKESMAS) spread throughout Indonesia is expected to play an important role in improving public health. Currently, the government still has obstacles in monitoring the readiness level of PUSKESMAS service and conditions. This research is aimed at creating a prototype, web-based geographic information system (WEBGIS) with the polygon method. This application is used as PUSKESMAS monitoring apps at the district level. Monitoring conditions include the availability of health workers, medical equipment standards, the PUSKESMAS kind and type, PUSKESMAS building conditions, as well as the spread of disease. This application is named SIGAPNAS (Sistem Informasi Geografis PUSKESMAS Nasional); a prototype of this application is an earlier prototype development that is expected to assist the public in seeking PUSKESMAS information. This application can also help the government in monitoring the development and levels of PUSKESMAS and the spread of a disease in an area. SIGAPNAS is expected to help the government succeed in national health programs through information technology.**

Keywords—PUSKESMAS; Polygon; SIGAPNAS; WEBGIS

I. INTRODUCTION

The government is currently developing a public health insurance program, providing health services for Indonesian citizens. Through the social security agency called Badan Penyelenggara Jaminan Sosial (BPJS) Kesehatan [1] authorities are trying to improve citizens' lives by providing health insurance. However, the implementation process encountered several obstacles, including hospital capacity and capability in providing health services, especially for hospitalization. Those problems can actually be solved by optimizing PUSKESMAS' role and facilities to support of health care. It became the main motivation for developing SIGAPNAS application, to optimize the role of PUSKESMAS with the aid of a Web-based Geographic Information System (WebGIS). Information technology could facilitate citizen to obtain information and help the government to monitor the level of readiness and state of health services in PUSKESMAS.

The use of web-based GIS has been developed in many applications, such as web services for geographic information systems [2], web GIS server solutions using open-source software [3], web GIS solution and 3D visualization towards sustainability of Georgetown as world heritage site [4], and development of mHealth for public health information collection, with GIS, using private cloud [5].

PUSKESMAS is one of the government-owned health facilities [6][7] spread out in every district throughout Indonesia. But However, today people still have difficulties in finding information about PUSKESMAS and communities still choose the hospital because they think PUSKESMAS are not adequately prepared to provide health service standards, such as in the availability of medical personnel, equipment, facilities for inpatient and emergency facility. On the other hand the government also encountered obstacles in promoting PUSKESMAS as one of the spearheads of health services. The government either has not been able to develop and monitor relevant health centers optimally, as well as the readiness of health services in PUSKESMAS. Those problems have inspired SIGAPNAS application development.

This application aims to provide information for community, such as the location of PUSKESMAS, health personnel availability, completeness of medical devices, and health programs enrolled by PUSKESMAS. In terms of local government, this application is expected to monitor condition, completeness, and the preparedness level of PUSKESMAS in providing health services. The Government could implement PUSKESMAS sustainable development by monitoring type and category of PUSKESMAS that shows readiness to provide inpatient services and emergency care. Governments can also monitor number of health personnel availability and completeness of medical devices.

SIGAPNAS prototypes were made in the scope of the districts with the purpose of facilitating the monitoring by using mapping feature. The method used in this study is a polygon WebGIS. This system uses water fall methodology [8]. This paper is organized as follows. Section II describes the

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features and the function of SIGAPNAS. Section III describes the design of polygon WEBGIS . Section IV is about the result and the discussion. Finally, section VI concludes the paper.

II. Features and The Function of SIGAPNAS

SIGAPNAS WebGIS application has functions for providing information and monitoring the availability of health workers, the type of clinic, disease, and the condition of the PUSKESMAS's building. Figure 1 shows the diagram of SIGAPNAS functions.

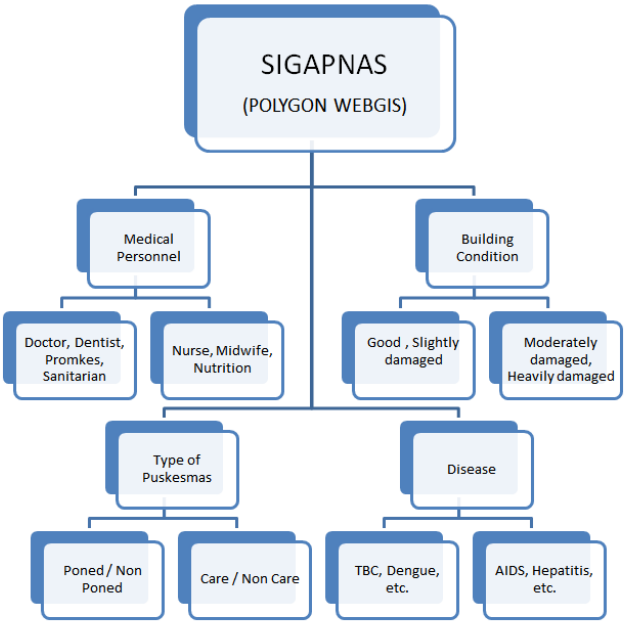


Figure 1. SIGAPNAS Functions Diagram

WebGIS that has been developed was based on map polygon to define the area at the district level. In the previous stage dot based mark on each PUSKESMAS on WebGIS was in use [9]. Polygon method is used to mark area districts in WebGIS. It is easier for people and governments to get the picture about the level of preparedness and PUSKESMAS services located in a certain district area. Usually one district consists of several PUSKESMAS.

In addition, this application is also equipped with a PUSKESMAS searching feature and route guidance towards PUSKESMAS from the center of the crowd. This application could be updated by each PUSKESMAS and could be monitored by the authorities.

III. Design of Polygon WEBGIS

A. Functions

In the first stage development of polygon-based WebGIS was done by searching and selecting sub district area on google maps. Figure 2. Shows districts area selection. For example, the selected district is Batununggal district.

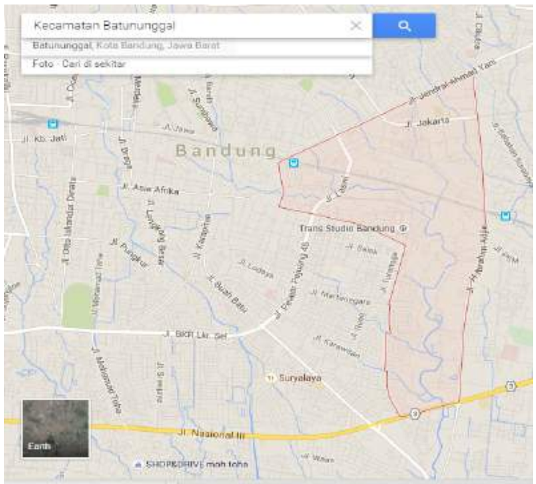


Figure 2. Batununggal District

Next step, determine district boundary points of the shaded area and retrieve the coordinates by capturing the coordinates to form a polygon area in accordance with district area. The more coordinates samples taken, the better forms of polygons formed. Figure 3 shows the method of determining and taking the coordinates to form a polygon of Batunnunggal district.

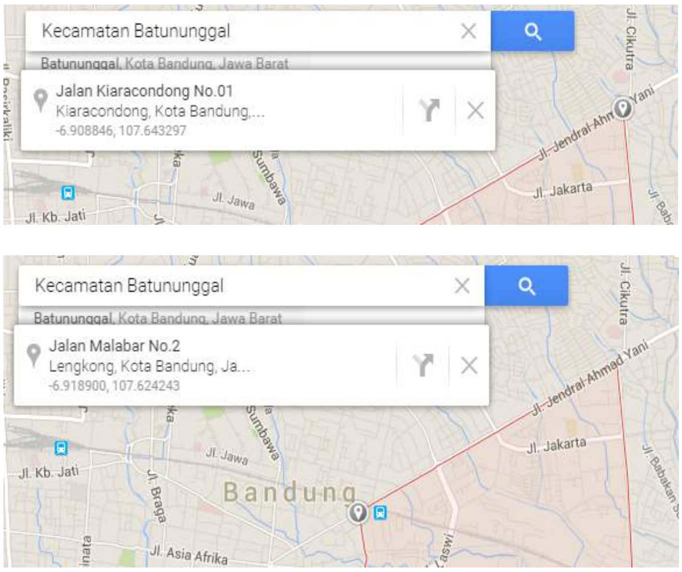


Figure 3. Batunungal district coordinates sampling

The results of the determination of Batunungal districts polygon coordinates can be seen in Table 1. There are 20 coordinates points taken to form a polygon of Batununggal district for webGIS. SIGAPNAS application uses the city of Bandung which has 73 PUSKESMAS spread across 30 districts, as prototype. Information data of PUSKESMAS taken from survey results and the experimental data did not depict the actual data. One district can consist of several PUSKESMAS. In the application of this SIGAPNAS webGIS information can be seen based on PUSKESMAS or district levels.

Table 1. Polygon Coordinate

No.	Coordinate	No.	Coordinate
1	-6.918815, 107.624200	11	-6.947103, 107.636280
2	-6.921968, 107.624800	12	-6.945430, 107.641967
3	-6.923757, 107.624200	13	-6.937107, 107.642449
4	-6.925376, 107.631581	14	-6.931792, 107.643179
5	-6.927932, 107.636559	15	-6.930877, 107.643940
6	-6.934493, 107.635014	16	-6.929109, 107.644101
7	-6.937688, 107.635229	17	-6.928661, 107.644370
8	-6.937645, 107.634843	18	-6.927394, 107.644413
9	-6.941884, 107.634692	19	-6.926978, 107.644541
10	-6.941990, 107.634950	20	-6.908731, 107.643251

After getting all of Batununggal district coordinates point, to form a polygon area on google maps, the entire coordinates are combined and incorporated into the google map code. Figure 4 shows the code for google map polygons of the set coordinates.

```
var kecbatununggal;
var batununggalCoord = [
new google.maps.LatLng(-6.918815, 107.624200),
new google.maps.LatLng(-6.921968, 107.624800),
new google.maps.LatLng(-6.923757, 107.624200),
new google.maps.LatLng(-6.925376, 107.631581),
new google.maps.LatLng(-6.927932, 107.636559),
new google.maps.LatLng(-6.934493, 107.635014),
new google.maps.LatLng(-6.937688, 107.635229),
new google.maps.LatLng(-6.937645, 107.634843),
new google.maps.LatLng(-6.941884, 107.634692),
new google.maps.LatLng(-6.941990, 107.634950),
new google.maps.LatLng(-6.947103, 107.636280),
new google.maps.LatLng(-6.945430, 107.641967),
new google.maps.LatLng(-6.937107, 107.642449),
new google.maps.LatLng(-6.931792, 107.643179),
new google.maps.LatLng(-6.930877, 107.643940),
new google.maps.LatLng(-6.929109, 107.644101),
new google.maps.LatLng(-6.928661, 107.644370),
new google.maps.LatLng(-6.927394, 107.644413),
new google.maps.LatLng(-6.926978, 107.644541),
new google.maps.LatLng(-6.908731, 107.643251)
];
kecbatununggal = new google.maps.Polygon({
  paths: batununggalCoord,
  strokeColor: '#FF0000',
  strokeOpacity: 0.8,
  strokeWeight: 3,
  fillColor: '#FF0000',
  fillOpacity: 0.35
 });
 kecbatununggal.setMap(map);
  var objbatununggal = {
     'id':'batununggal',
     'area':kecbatununggal
   };
  kecbatununggal.objInfo = objbatununggal;
```

Figure 4. Google map Polygon Code.

The results of the polygon code on google map are shaded areas of Batununggal district as shown in Figure 5. The area of the polygon can be set and linked to the database on the server to display the information as needed.

Figure 5. Polygon map kecematan batununggal

One of the polygon features on the webGIS SIGAPNAS map shows the fulfillment of medical personnel and equipment availability standard. Polygon colors will change according to the fulfillment of standards in every PUSKESMAS that is located in a district. Figure 6 addressing code is used to change the color of the polygon area in accordance with the set of standards limit.

The green color indicates that each PUSKESMAS in the district already meets meet the set of standards limit. Yellow of the polygon shows that PUSKESMAS have met 50% of minimum standard, while the red color indicates that one or more PUSKESMAS do not meet standards or less than 50%. The following is a standard formula applied to the polygon area:

Calculation of Standard Limits for districts
x = standard limits
n = number of variables
Green → each n / x at PUSKESMAS in districts = 100%
Yellow → 100% > each n / x at PUSKESMAS in the districts of>50%
Red → each n / x at PUSKESMAS in districts <50%

```
foreach ($kecamatan ->result() as $camat) {

    foreach ($statistik->result() as $s) {
        if ($s->kode_kec == $camat->kode_kec) {
            $jumlah=$s->jumlah;
            if ($jumlah==null) {
                $jumlah=0;
            }

            $detail['kode_kec'][$i]=$s->kode_kec;
            if (($jumlah/$standar)<1) {
                if (($jumlah/$standar)<0.5) {
                    $detail['warna'][$i]="red";
                } else {
                    $detail['warna'][$i]="yellow";
                }
            } else {
                $detail['warna'][$i]="green";
            }
        }
```

Figure 6. Polygon Coloring Code

In the SIGAPNAS prototype, webGIS polygon can display data in the form of a dialog that appears when district polygon is accessed. To access the dialogue, showArrays and event listener functions should be used.

1. Creating an event listener 'click' on each district area polygon.

   Event listener 'click' can be added to the function area / polygon. The word 'click' is taken as to enable event listener to access the polygon, so that each cursor action 'click' could be responded. Figure 7 shows the code for the event listener function. After providing the function 'click', the next function is to call additional functions for displaying data, namely the show Arrays function.

Figure 7. Event Listener Code Function

2. Creating show Arrays function

Show Arrays function (event) has a function to retrieve JavaScriptObject Notation (JSON) data from the server to be displayed in the form of a dialogue in every area of the polygons. JSON datais already available on the server. Consequently, show Arrays function is aimed for displays only. Figure 8 shows the syntax of the function call show Arrays in the JSON data from the server.

Figure 8. Show Array function and JSON Code

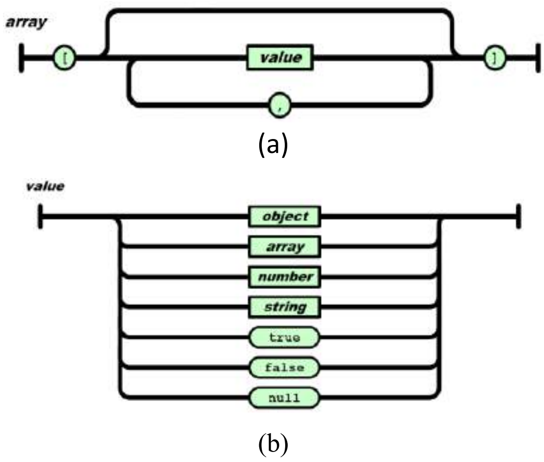


Figure 9. (a) Array and (b) value in JSON [9]

In JSON an object is an unordered set of name or value pairs. An object begins with { (left brace) and ends with } (right brace), each name is followed by: (colon) and the name/value pairs are separated by , (comma) [10]. An array is an ordered collection of values and a value can be a string in double quotes, or a number, or true or false or null, or an object or an array. All of the structures, can be nested[10].

The sample results from the data of the event listener and show Array function using JSON on polygon web GIS can be seen in Figure10. The two functions can be called from the server display disease data for each PUSKESMAS in Sukasari district.

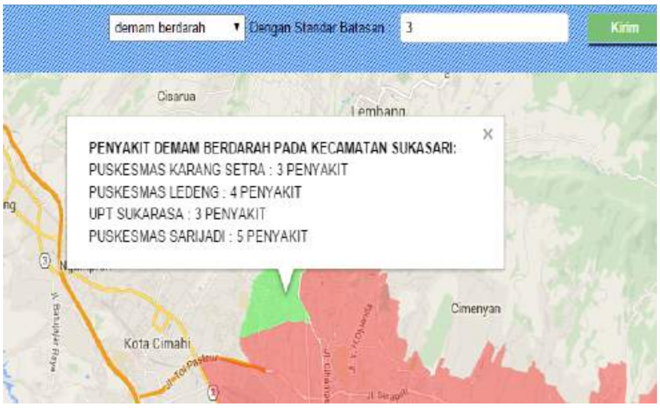


Figure 10. Data Dialog from Event Listener and ShowArrays.

IV. RESULT AND DISCUSSION

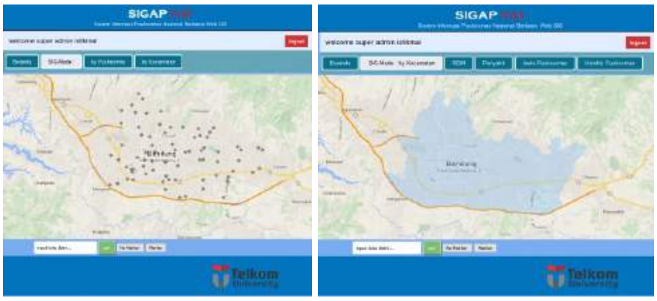
The front page of SIGAPNAS applications as it is accessed via the web[11] is shown in Figure11.When a user enters the SIGAPNAS application, he may find variety of menus including, the search function for PUSKESMAS, and tracking as shownin Figure12. The user can directly search for PUSKESMAS on the map and get the necessary information.

Figure 11. Front Page Menu Display

Tracking function is used to help people find different routes to the PUSKESMAS, such as from Bandung town square. PUSKESMAS search function can be accessed by entering the name of PUSKESMAS. If the user does not know the name of PUSKESMAS, it be done by directly through the map by activating the function of PUSKESMAS marking in the map. On the map, the marking for PUSKESMAS can be hidden or shown by clicking the marker function.

Figure 12. SIGAPNAS Menu

Information about health services in SIGAPNAS application can be displayed based on PUSKESMAS or districts area, as can be seen in Figure13. In the display based on PUSKESMAS, the data to be displayed is the detailed data for PUSKESMAS, while when based on districts the data are total data for the districts if there is more than one PUSKESMAS.



(a) (b)

Figure 13. SIGAPNAS WEBGIS in accordance with

(a) PUSKESMAS (b)District

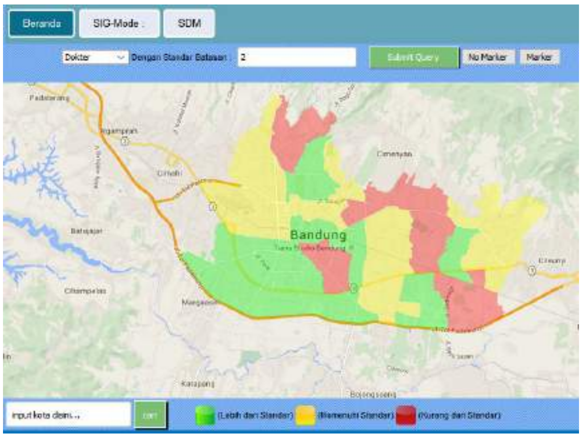


Figure 14. Polygon-based WEBGIS for Doctors in Distric Level

SIGAPNAS polygon-based application can display PUSKESMAS data based on the district. Figure 14 shows webGIS polygon of the number of the availabe doctors, where 2 is the standard number for doctors in charge for each health center. The red indicates there is a health center in the district that not have a doctor. The green color code indicates that all health centers in the districts already have at least two doctors, while the yellow one indicates that each health center has at least one doctor.

At SIGAPNAS application, to get more of polygon district information could be viewed directly to PUSKESMAS in each district. By enabling the marker will appear PUSKESMAS point in the district which can be accessed directly, as shown in Figure15.

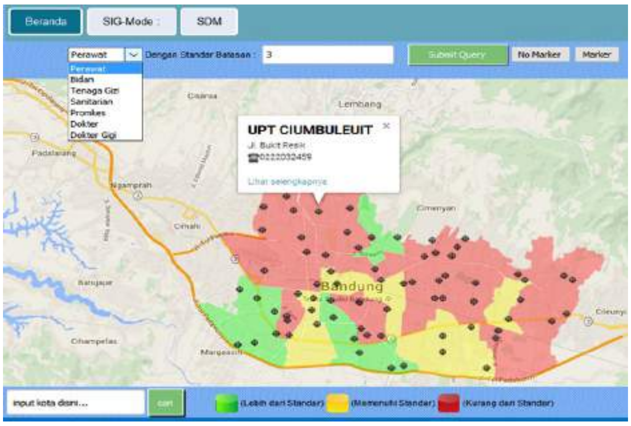


Figure 15. PUSKESMAS level information in Polygon WEBGIS

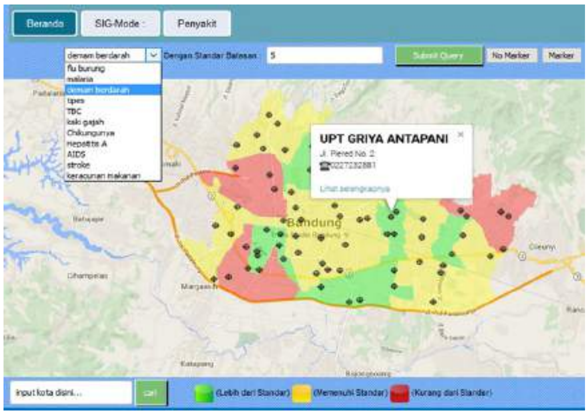


Figure 16. WEBGIS Polygon for desease in puskesmas in distric level

Polygon features display not only the data on human resources, PUSKESMAS types and the real condition but also the information about the spread and treatment for diseases in PUSKESMAS in district area. Polygons diseases monitoring also have the same role as human resources monitoring. The standard display that is in accordance with PUSKESMAS or district takes the standard number of events (related to disease) into account, as can be seen in Figure16. Knowing the spread and incidence of disease management in an area is important.

This application is expected to do better prevention and treatment of a disease.

Figure 17. PUSKESMAS information

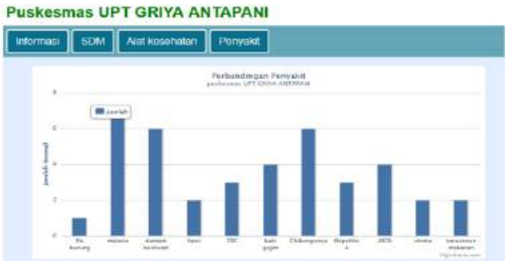


Figure 18.The disease data in the PUSKESMAS

As mentioned previously, the polygon webGIS also provides a marker feature to access any PUSKESMAS. Figure 17 shows the general information of health workers, medical equipments, as well as spread disease that can be accessed through dialog box that appears on WebGIS polygon.

The information of the spread of disease can be accessed directly by comparing the spread of disease in PUSKESMAS, as shown in Figure 18. This function can precisely determine the spread of disease in any PUSKESMAS that can help government to take immediate steps for the prevention and treatment towards the increase number of the disease.

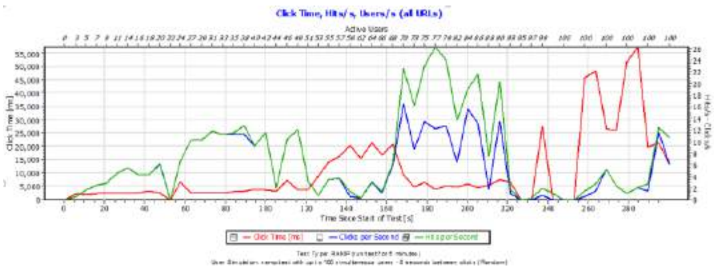


Figure 19. Webstrees Test Result

SIGAPNAS Application was also tested using web stress tool for evaluating performance of web server aplication for giving services to clients. One hundred (100) client was acessing web service at the same time giving the best performance 16 request/second to be respond succsesfully, can be seen at Figure 19. The result is not good enough for web service for large community, it could be improved by using dedicated resource server (dedicated or virtual private server).

V. CONCLUSION

This research tried to create a prototype of web-based application service using a geographic information system with polygon method to mark an area. This application is called SIGAPNAS, aimed at optimizing the function of PUSKESMAS in providing health care and supporting government programs. This application can help people to find information about PUSKESMAS. In the future, This application is also expected to assist the government in monitoring the level of service and the readiness of PUSKESMAS including monitoring the spread of disease. SIGAPNAS in general expected to help the government succeed the national health programs through information technology.

REFERENCES

[1] Undang-undang Repebulik Indonesia nomor 24 tahun 2011 tentang Badan Penyelenggara Jaminan Sosial. http://bpjs-kesehatan.go.id/bpjs/index.php/unduh/index/1.

[2] Shengru Tu, Abdelguerfi M. Web Services for Geographic Information Systems. Internet Computing, IEEE. 2006.

[3] Daoxun Xia, Xiaoyao Xie, Yang Xu. Web GIS server solutions using open-source software. IEEE International Workshop on Open-source Software for Scientific Computation (OSSC), 2009.

[4] Samad A.M., Hussein S.M., Karnadi M.S., Bohari S.N., Suldi A.M., Maarof I . Web GIS solution and 3D visualization towards sustainability of Georgetown as world heritage site. IEEE 8th International Colloquium on Signal Processing and its Applications (CSPA) 2012.

[5] Boonchieng E., Boonchieng W., Senaratana W., Singkaew J. Development of mHealth for public health information collection, with GIS, using private cloud: A case study of Saraphi district, Chiang Mai, Thailand. International Computer Science and Engineering Conference (ICSEC), 2014.

[6] Research and Development Ministry of Health of the Republic of Indonesia, RIFASKES 2011 interim results.

[7] Istikmal, Yuliant S., Ratna M, Tody AW, Ridha M, Kemas ML., Tengku AR.," Rifaskes Geographic Information System Based on Web " EECCIS 2012.

[8] Production of Large Computer Programs. *IEEE Annals of the History of Computing* (IEEE Educational Activities Department)5 (4): 350–361.doi:10.1109/MAHC.1983.10102. Retrieved 2011-03-21. Benington, Herbert D. (1 October 1983).

[9] Istikmal, A. M. Tody, and V. Y. Leanna, "PUSKESMAS Information System based on WEBGIS Case Study City of Bandung". 2nd International Conference on Technology, Informatics, Management, Engineering & Environment, TIME-E 2014.

[10] The JSON Data Interchange Format. Ecma International 2013. 1st Standar ECMA-404. 1st Edition / October 2013. www.json.org.

[11] Prototype of SIGAPNAS. www.sigapnas.org

Throughput Performance of Routing Protocols Based on SNR in Wireless Mobile Ad Hoc Networks

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Abstract— **Routing protocols play important roles in mobile adhoc network performance. The Protocols find and decided the communication path with the routing algorithm to deliver the data along the networks. Channel qualities in wireless communication are an important factor in network performance. Previous research in routing protocol development did not much pay attention in this matter. This paper shows the routing algorithm which not concern with this channel quality cannot adapt well in Mobile Adhoc Networks (MANET). We evaluated reactive routing protocol in deterministic and probabilistic propagation models, which are two-rayground and shadowing. These models are used to predict the received signal power of each packet. The result shows the degradation of routing protocol performance in more realistic environment, which is shadowing propagation model. Then we used mathematical analysis based on throughput performance related to the probability of error and Signal to Noise Ratio (SNR). Routing protocols face uncertainty in link quality when the topology of the network became dynamic. The routing protocols have to consider channel quality in the link transmission to acquire higher throughput performance.**

Keywords—MANET, Routing, Propagation, Throughput, SNR

I. INTRODUCTION

In the future Mobile Adhoc Networks (MANET) is believed to have more significant role in supporting the development of Internet of Think (IoT), such as mobile to mobile [1] and machine to machine communication [2][3]. Some of the important issues in wireless ad hoc networks are energy efficient, congestion, quality of transmission, routing, and adaptive protocol. Routing protocol is one of the protocols that play an important factor in MANET performance. These protocols find and determine the route path to transmit the data from source to destination. Mobility and effects on the transmission of wireless channel will greatly affect the performance of the routing algorithm. Unlike in wired networks where quality transmission paths are more stable, while in the wireless mobile network the quality is influenced by many factors such as speed, mobility, Bit Error Rate (BER), and the path break.

Routing protocols are generally made without considering the quality of the chosen path, but more often used the shortest path search scheme, which is using the smallest number of hops, such as Ad hoc On-Demand Distance Vector (AODV), Dynamic Source Routing (DSR), and Destination-Sequenced Distance Vector (DSDV) [4]. Currently the quality-aware routing (QAR) has evolved to meet the challenges of the routing process in wireless networks.

Several methods QAR has been developed, such as Expected Transmission Count (ETX) [5], Expected Transmissions Time (ETT) using estimates loss rate and bandwidth link [6], Modified ETX (mETX) and effective Number of Transmission (ENT) [7], Medium Time Metric (MTM) [8], development ETX with adding neighborhood heuristics (NH) [9] and bandwidth-aware high-throughput routing with successive interference cancellation [10]. Other approaches, with quality channel aware, such as using SNR to improve multi-hop routing [11], using link quality indicator [12], using channel quality to choose the path of least stable [13],and using the Receive Singnal Strength Indicator (RSSI) as the routing parameter [14].

Previous research did not much pay attention to the routing development with the impact of propagation model. Especially, analysis of throuhgput performance related to route process, channel quality, and probability of error. In this paper, we investigated the performance of reactive routing protocol AODV in differents propagation model. We used two ray-ground as a deterministic propagation model and shadowing for probabilistic model. This investigation to seek the impact of channel quality of the routing performance and the routing algorithm adaptation. Then we make a model route process in AODV to use in mathematical analysis to seek the throughput performance correlation between the route process, SNR, and the probability of error.

The paper is organized as follows. Section II provides an overview of the mechanism of routing protocols and section III describes the propagation models. Section IV describes the routing protocol performance and mathematical analysis. Finally, section V is the conclusion and future research opportunities.

II. ROUTING PROTOCOL

Routing is a function of the network layer to determine the route from the source to the destination node. In wired networks, route failures are rare, whereas in mobile ad hoc networks often occur. The main cause of route failures is the mobility of the nodes, another factor is the contention of the wireless channel. The duration of the route search returned after route failure in adhoc networks depends on the routing protocol, node mobility patterns, and traffic characteristics.

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When a new route has been constructed and longer or shorter than the old route, the congestion control will face the fluctuations in the Round Trip Time (RTT). Routing protocols that often used in ad hoc networks are AODV, DSR and DSDV. AODV and DSR represent reactive routing protocol, while DSDV is a proactive routing protocol [4].

AODV is a reactive routing protocol as DSR, the route search is process when the node will communicate. In AODV, when a route is not used in a certain period of time (lifetime), then it will be removed from the routing table memory [4]. AODV does not require the maintenance of the route from the source node to the destination node when not communicating. AODV only save the route cache or record the route last used to each node in the network. This will minimize the number of active routes that must be monitored and maintained. The lifetime will be updated when each route is used. Fig. 1 shows the AODV routing protocol mechanism.

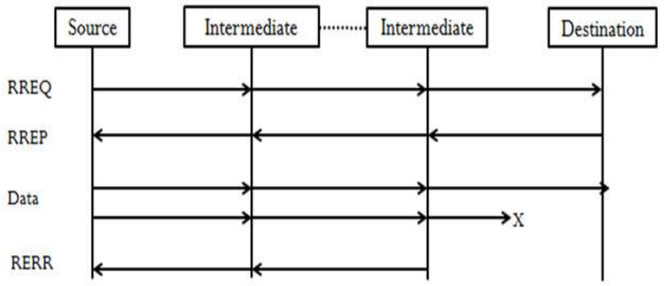


Figure. 1 AODV routing protocol mechanism.

AODV will find a path for communication based on shortest path which is a path with the smallest number of hops. Fig. 2 shows the AODV path between the source (s) to the destination (D) through intermediate node 2 with smaller hop count (hop count 2) compared through intermediate 1 with hop count 3. In this situation, the mechanism works well and the data can transfer from source to destination.

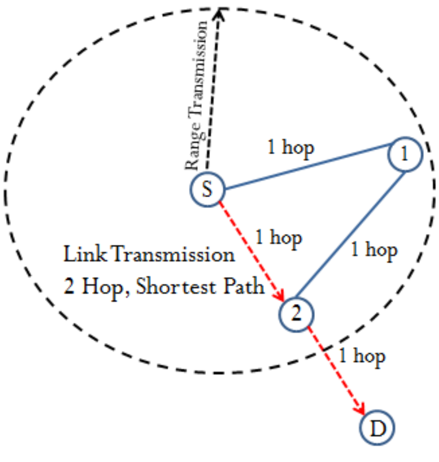


Figure. 2 Shortest paths in AODV

III. Propagation Model

Propagation model used to predict receiver signal power in the receiver. These propagation models can be divided into large-scale path loss models and small-scale path loss model (fading channel) [20]. From the viewpoint of implementation can also classify into deterministic and probabilistic models [15]. The existence of the reflection of some of the objects and movement, can cause a strong signal received by the receiver is varied, and the received signal gets path loss suffered. The path loss will limit the performance of mobile communication systems, so the predicted path loss is an important part in the planning of mobile communication systems. Path loss that occurs in the received signal can be determined through a certain propagation model. In this paper we evaluated the reactive routing protocol with two-ray ground deterministic propagation model, and shadowing as a probabilistic propagation model.

A. *Two-RayGround, Deterministic Propagation Model.*

In free space, transmitter and receiver unhindered, line-of-sight (LOS) path between them, without any other object in the environment. It predicts that received power decays as a function of transmitter-receiver distance raised to some power, the well-known Friis equation is used to calculate the received power [15][16]:

$$P_r(d) = \frac{P_t G_t G_r \lambda^2}{(4\pi)^2 d^\alpha L} \quad \text{...(1)}$$

Where, L is the system loss , λ is the wavelength of the transmitted signal, Pt is the transmitted power, Gt and Gr are transmitter antenna gain, α is the path loss exponent which is 2 for free space and *d* is the separation distance between the transmitter and the receiver.

Two-ray ground model is a more realistic model than the free-space model, this model consider a ground-reflected propagation path between transmitter and receiver, in addition to the direct LOS path. Two-ray ground propagation model is a model that is based on geometrical optics and can be used for the direct path and the reflection of the ground between the sender and the recipient. The power at receiver calculated with this equation [15]:

$$P_r(d) = \frac{P_t G_t G_r h_t^2 h_r^2}{d^4 L} \quad \text{.....................................(2)}$$

This model is considered more accurate than the model of free space to estimate the signal strength on a wide scale from a distance of several kilometers for mobile systems. Based on the above equation, the power loss is faster than the free space propagation model when the distance increases.

B. *Shadowing, Probabilistic Propagation Model.*

Probabilistic model is a more realistic propagation model. This model takes a deterministic model as one of the input parameters to get the average distance transmission. For each transmission, the received power described in the distribution. This leads to more varied results obtained from the successful reception. This happens with different probabilities as two adjacent users cannot communicate, the opposite can occur of probability of two users can communicate outside the range deterministic transmission. Distribution of this effect depends on probability models and parameters used. Fig. 3 shows the path loss, shadowing, and multipath in propagation model.

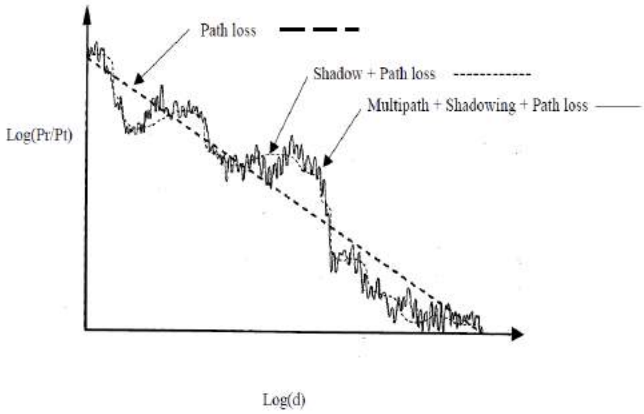


Figure 3. Path loss, shadowing, and Multipath [14]

Model of log-normal shadowing using a normal distribution with variance σ of the distribution of power acceptance in the logarithmic domain [15]:

$$P_r(d;\sigma^2) \sim LN(P_{r_{det}}(d), \sigma^2) \quad \text{……………………..…….(3)}$$

The shadowing model consists of two parts. The first one is known as a path loss model, which also predicts the mean received power at distance d. Deterministic model Pr_{det} as in *free space* and *two-ray ground*. Power receiver can calculate with the equation:

$$P_r(d) = P_t - \overline{PL}(d_o) + 10\alpha\log\left(\frac{d}{d_0}\right) + X_\sigma \quad \text{…………...(4)}$$

The second part of the shadowing model reflects the variation of the received power at certain distances. It is a log-normal random variable. It is about gaussian distribution if measured in dB. Exponent *path loss* α and $PL(d_0)$ are the reference measurement of pathloss close to the transmitter. So the above equation can be rewritten as:

$$P_r = P_{r_{det}}(d_0) \times 10^{PL(d)} \quad \text{………………………..…(5)}$$

In dB, the received power is obtained by multiplying the received power with a scale factor deterministic power loss is:

$$PL(d) = -10\alpha\log10\left(\frac{d}{d_0}\right) + X_\sigma \quad \text{…………..…………...(6)}$$

IV. Routing Protocol Performance

A. AODV Performance with Two-Rayground and Shadowing

In this section we evaluated the performance of the AODV routing protocol with two different propagation model. We evaluated with two-rayground as a deterministic propagation model and shadowing as a probabilistic propagation model. In this experiment we used network simulator [18] with the parameter as shown in table 1.

Table 1. Simulation parameter

Mobility	Random Way Point
Number of nodes	30
Speed of nodes	2-20 m/s
Propagation Model	Two-Rayground and Shadowing
Routing Protocols	AODV
Application	FTP
Loss System (L)	1
Gain Antena (Tx,Rx)	1
High of node (h_t,h_r)	1,5 m
path loss exponent	2
shadowing deviation	4 dB
reference distance	1 m
Power Transmit Tx	0.28183815 W
Frequency	914.e6 Hz

Performance of the AODV routing protocol has significant degradation when used shadowing compare to two-rayground propagation model, as shown in fig. 4. Beside the distance between the source and the receiver the performance influence by the number of hops in the path. Increasing the number of hops will raise the probability of path break. This path break depends on the channel quality of the link. In two-rayground the channel quality depend on distances, while in shadowing the quality link also reflects the variation of the received power at certain distance. The routing protocol face often path break at shadowing because AODV did not pay attention to link quality in route selection.

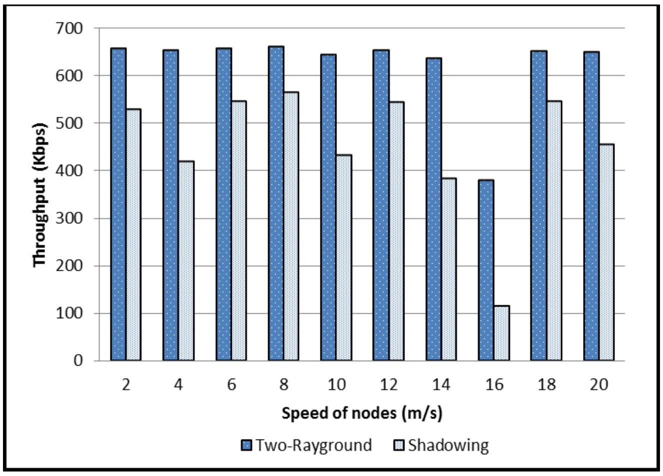
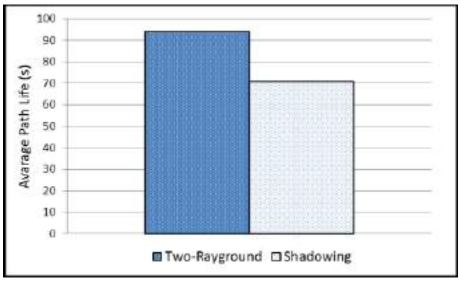


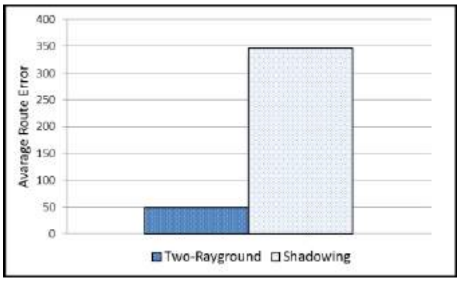
Figure 4. AODV throughput with two-rayground and shadowing

When link break occurs in the transmission path, AODV will generate RERR (Route Error), see fig. 1, this route error sends back to the source to give notification about link break in the path. Source will start again to find the path and send the RREQ (Route Request). The frequent of the path breaks indicate by the number of route error (RERR) generate by routing protocol. When the link breaks occurs, the path will be broken too. The averages duration life of this path is path life.

The average duration of path life as shown in fig 5. AODV in shadowing generate higher RERR and has smaller path life compare to the two-rayground.



(a)



(b)

Figure 5. (a) Avarage path life (b) Avarage route error

When the nodes are mobile, the topology became dynamic. Routing protocol will face often path break and lead to find a new path mechanism. There will be a situation where routing protocol gets high overhead due to route request over and over. The problem is when the routing protocol algorithm chooses shortest path according to the minimum number of hops between the source and destination, while one or more of this hop has minimum channel quality or minimum radio transmission range, as shown in Fig. 6 between node S and intermediate node 2. This hop seems to break and the path communication will break sooner. This situation will lead routing protocol to seek new routes (higher RERR) and smaller path life in data transfer. This will lead networks with low performance in throughput and higher overhead in route process.

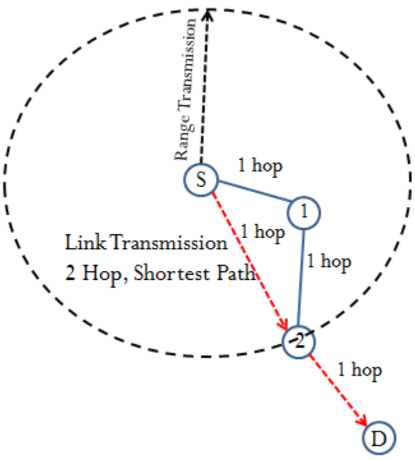


Figure 6. Hop with minimum channel quality

B. Throughput Model and Mathematical Analysis.

From the simulation result in section IV.a, the performance of AODV routing protocol decrease in shadowing compare to two-rayground propagation model. To analysis the impact between SNR and BER to the throughput, we make a model for AODV process, as shown in Fig. 7. Denote λ is the average total number of bits of data are transferred during T, with the total bits to process forward route (RREQ) is Kh. The acknowledgment route process is Ka bits, Route Reply (RREP). The transmission rate is R bps. Where ta is the line turnaround time (RTS-CTS delay) in seconds and the propagation delay between transmitter and receiver is tp seconds. Then average a total of $\lambda - Kh$ bits of data transferred with time T_{total}:

$$T_{total} = \left[\frac{\lambda + Ka}{R}\right] + 2\,(ta + tp) \quad \text{.........................(7)}$$

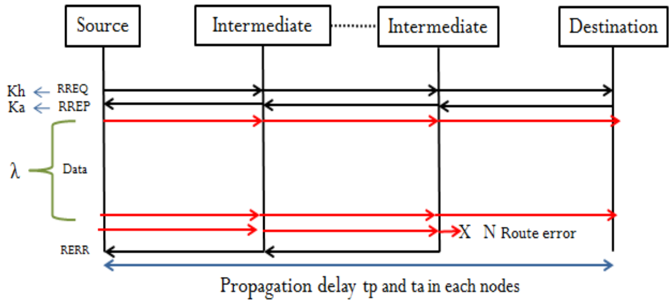


Figure 7. AODV process model

Throughput is total of data transferred during time to transfer those data is $(\lambda - Kh) / T_{total}$. We can express the throughput as follows:

$$Throughput = \frac{\lambda - Kh}{\left[\frac{\lambda + Ka}{R} + 2(ta + tp)\right]} \quad \text{..................... (8)}$$

If the effective rate transmission is R_{eff} as the throughput, and transmission rate is R, than we can express the effeciency $\eta = R_{eff} / R$. If the probability of error P for x transmission can be express as:

$$P\,(x\ transmission) = P^{x-1}\,(1 - P) \text{.........................(9)}$$

Then the probability distribution with mean δ, can be express as:

$$\delta = \sum_{X=1}^{\infty} x\, P^{x-1}(1 - P) = \frac{1}{1-P} \quad \text{.........................(10)}$$

Equation (8) can be written with mean $\delta = 1/(1\text{-}P)$ attempts are made before acceptance as follows:

$$Throughput = \frac{\lambda - Kh}{\frac{1}{1-P}\left[\frac{\lambda + Ka}{R} + 2(ta + tp)\right]}$$

$$= \frac{(1-P)(\lambda - Kh)R}{\lambda + Ka + 2R(ta + tp)} \text{.......................(11)}$$

The equation (8) is for transmission without error. If we consider the propagation delay *tp* and *ta* are small, according no queuing in the buffer, short range transmission, and smooth line turnaround time. Thus, for continues transmission without error the dominator equation (8) is λ/R, if we consider Ka, then the time is (λ+Ka)/R. For routing process with Route Error (RERR) when the path break occurs (transmission error), the number of bits or frames to get accepted can be calculated as follows. Transmission of number of bits or one frame, with probability P (1 - P) transmission of N extra frames or bits (retransmission of N RERR) for route error. Thus the probability transmission of 2N extra frames P^2 (1 - P), so the mean δ can be written as:

$$\delta = 1 + NP(1-P) + 2NP^2(1-P) + \cdots$$

$$= \frac{1+(N-1)P}{1-P} \quad \ldots\ldots(12)$$

Thus, in this case equation (8) can be written as follows:

$$Throughput = \frac{\lambda - Kh}{\frac{\lambda+Ka}{R}\left[\frac{1+(N-1)P}{1-P}\right]}$$

$$= \frac{(1-P)(\lambda-Kh)R}{[1+(N-1)P](\lambda+Ka)} \quad \ldots\ldots (13)$$

For simply the equation, we can express equation (13) as equation (14) if we use a model approach in selective ARQ [17], where:

$$Throughput = \frac{\lambda - Kh}{\frac{\lambda+Ka}{R}\left[\frac{1}{1-P}\right]}$$

$$= \left(\frac{(1-P)(\lambda-Kh)}{\lambda+Ka}\right) R \quad \ldots\ldots(14)$$

From equation (14) we can find efficiency η of the networks, since the throughput is effective sending rate (R_{eff});

$$\eta = \frac{Reff}{R}$$

Then we can write this efficiency from equation (14) as follows:

$$\eta = \frac{(1-P)(\lambda-Kh)}{\lambda+Ka} = (1-P)\left(\frac{\lambda-Kh}{\lambda+Ka}\right) \ldots\ldots(15)$$

From equation (14) and (15), we can see that the throughput and the efficiency will get higher when the probability of error is smaller. To analysis the correlation between throughput with SNR and BER, we used the most common equation approach to compute the probability of error P is [17]:

$$P = 1-(1-\beta)^n \quad \ldots\ldots(16)$$

With β representing the Bit Error Rate (BER) and n is the number of bits in a packet or frame. We can find BER with compute Signal to Noise Ratio (SNR) and modulation technique which is used binary phase shift keying (BPSK), we can express BER [19] as follows:

$$BER = Q\left(\sqrt{2\frac{Eb}{No}}\right) \ldots\ldots(17)$$

If the bandwidth of signal information exactly same with a bandwidth of noise or in the receiver use matched filter to get maximum SNR, then (Eb/(No/2)) = (S/N). For β is the bit error rate (BER), then we can write the probability of error as BER and SNR function, as follows:

$$P = 1-(1-BER)^n$$

$$\frac{S}{N} \approx 2\frac{Eb}{No}$$

$$P = 1-(1-Q(\sqrt{\frac{S}{N}}))^n \quad \ldots\ldots(18)$$

In two-ray ground the power receiver depends on the distance between transmitter and the receiver, eq. (2). While in shadowing which also predicts the mean received power at distance d and the model reflects the variation of the received power at certain distances, eq. (5). From simulation, we can see the relation between SNR and the throughput with the distance change between transmitter and receiver. Signal to noise ratio will decrease as the distance between transmitter and receiver move away, as shown in fig. 8. In shadowing model the degradation SNR higher than two-rayground propagation model.

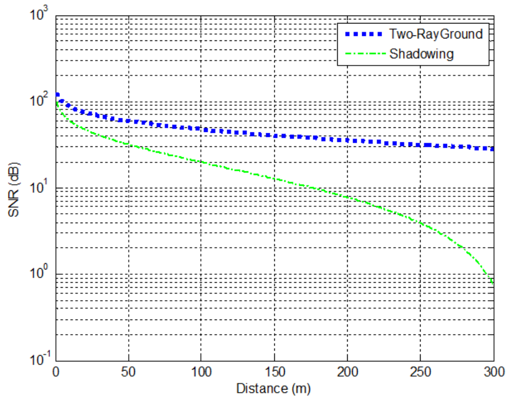


Figure 8. SNR Two-Ray and Shadowing versus distance

From equation (19) and (20), we can see the probability of error and bit error rate will increase along with the degradation of SNR between transmitter and receiver, and the value of SNR will decrease with added difference distance between them. As shown in fig. 9 adding the distance will increase the value of bit error rate.

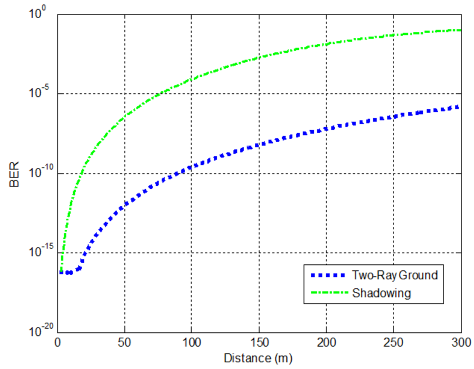


Figure 9. BER Two-Rayground and Shadowing vs distance

We can see that higher SNR will decrease BER and the probability of error. The minimum value of probability of error can gain maximum throughput of the system. A path consists of hop with higher SNR can improve throughput the system. Where in shadowing model the degradation more aggressive compare to two-rayground because effect of the shadowing. From equation (17) and equation (18), we can express the throughput (R_{eff}) for BSPK as follows:

Throughput = R_{eff}

$$= \left(\frac{(1 - Q(\sqrt{\frac{S}{N}}))^{n} (\lambda - Kh)}{\lambda + Ka} \right) R \quad \text{.........................(19)}$$

In general, for other modulation we can use this equation:

$$= \left(\frac{(1 - BER)^{n} (\lambda - Kh)}{\lambda + Ka} \right) R \quad \text{.....................................(20)}$$

From the result in section IV.a, we can see that routing protocol face uncertainty performance, especially in the more realistic propagation model, such as shadowing. Mathematical approach analysis in section IV.b shows the role of SNR and BER to reduce probability of error to gain maximum throughput. AODV choose the path based on shortest path which is the minimum hop count. This mechanism did not pay attention to the channel quality of the hop. With the experiment in shadowing model, we can see this mechanism did not work well, and get performance degradation. In the future, for realistic routing protocol, channel link quality can be considered as metric in routing protocol algorithm, but with minimum overhead and complexity.

V. CONCLUSION AND FUTURE WORK

This paper has shown AODV routing protocol performance in two-rayground and shadowing propagation model. From the simulation AODV has lower performance in a more realistic environment, which is shadowing than two-rayground. This happens because the routing protocol did not pay attention in quality of the link when choose the hop as one of the path. The simulation and mathematical analysis has shown that SNR impact on link quality. Low SNR in the link transmission will cause the higher probability of error and decrease the throughput. In the future the routing protocols have to consider channel quality in the link transmission to achieve higher performance. Cross layer scheme is one of promosing method to improve cooperation between routing protocol and physical layer.

REFERENCES

[1] Muhammad Riaz, Noor M. Khan, Syed Junaid Nawaz. A Generalized 3D Scattering Channel Model for Spatio-Temporal Statistics in Mobile-to-Mobile Communication Environment. IEEE Transactions on Vehicular Technology 2014.

[2] Attwood, A., Merabti, M., Abuelmaatti, O. IoMANETs: Mobility architecture for wireless M2M networks. GLOBECOM Workshops (GC Wkshps), 2011 IEEE

[3] Farah Kandah, Yashaswi Singh, Chonggang Wang. (2011). Colluding Injected Attack in Mobile Ad-hoc Networks. IEEE INFOCOM 2011 Workshop on M2MCN-2011.

[4] E.M. Royer, C.K. Toh. (1999). "*A Review of Current Routing Protocols for Ad Hoc Mobile Wireless Networks*".IEEE Personal Communications, pages 46-55. 1999.

[5] D. De Couto, D. Aguayo, J. Bicket, and R. Morris. (2003). A high throughput path metric for multi-hop wireless routing. International Conference on Mobile Computing and Networking, 2003.

[6] R. Draves, J. Padhye and B. Zill. (2004). Routing in multi-radio, multi-hop wireless mesh networks. ACM Conference on Mobile Computing and Networking (MOBICOM), Sep. 2004, pp. 114–128.

[7] Can Emre Koksal and Hari Balakrishnan. (2006). Quality-Aware Routing Metrics for Time-Varying Wireless Mesh Networks. IEEE Journal on Selected Areas in Communications, vol. 24, no. 11, November 2006.

[8] B. Awerbuch, D. Holmer, and H. Rubens. (2006). The Medium Time Metric: high throughput route selection in multi-rate ad hoc wireless networks. Mobile Networks and Applications, vol. 11, pp. 253–266, 2006.

[9] Declan T. Delaney, Russell Higgs, and Gregory M. P. O'Hare. (2014). A Stable Routing Framework for Tree-Based Routing Structures in WSNs. IEEE Sensors Journal, vol. 14, no. 10, october 2014.

[10] Runzi Liu, Yan Shi, King-Shan Lui, Min Sheng, Yu Wang, and Yuzhou Li. (2015). Bandwidth-Aware High-Throughput Routing with Successive Interference Cancellation in Multihop Wireless Networks. IEEE Transactions on Vehicular Technology 2015.

[11] Ramon Aguero, Jose Antonio Galache and Luis Munoz. (2009). Using SNR to improve multi-hop routing. IEEE 69th Vehicular Technology Conference, 2009.

[12] R. Butt, Javed M.M., Akbar A.H., Taj Q.-u.-A., Chae-Seong Lim, Ki-Hyung Kim. (2010). LABILE: Link quAlity-Based LexIcaL Routing MEtric for Reactive Routing Protocols in IEEE 802.15.4 Networks. 5th International Conference on Future Information Technology (FutureTech), 2010.

[13] B. Amiri, H. R. Sadjadpour, JJ Garcia-Luna-Aceves. (2011). Outage Optimum Routing for Wireless Networks. 7th International Wireless Communications and Mobile Computing Conference (IWCMC), 2011.

[14] Marwan Al-Jemeli, Fawnizu Azmadi Hussin. (2012). A Link-Quality and Energy Aware Routing Metric for Mobile Wireless Sensor Networks. 4th International Conference on Intelligent and Advanced Systems (ICIAS2012).

[15] Andrea Goldsmith, "Wireless Communications," Cambridge University Press, 2005,644 pp.

[16] E.M. van Eenennaam. A Survey of Propagation Models used in Vehicular Ad hoc Network (VANET) Research. Design and Analysis of Communication Systems group, Faculty of EEMCS, University of Twente, The Netherlands.

[17] John D. Spragins, Josep L. Hammaond, and Krzysztof Pawlikowski. Telecommunications protocols and design. November 2006.

[18] Network Simulator NS 2.35. http://www.isi.edu/nsnam/ns/ns-build.html.

[19] Masood Khosroshahy. Study and Implementation of IEEE 802.11 Physical Channel Model in YANS (NS3 prototype) Network Simulator.

[20] Bernard Sklar. Rayleigh Fading Channels in Mobile Digital Communication Systems Part I: Characterization. IEEE Communications Magazine. July 1997.

A Simulation Study of Traffic Agent to Identify Traffic Flow Density Using Modified Traffic Cellular Automaton Model

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***Abstract*—This paper presents the evaluation of traffic agent utilization to identify road traffic flow density. This research also proposes the concept of traffic agent as the new method for traffic monitoring and surveillance.**

***Keywords*—traffic agent; traffic flow estimation; traffic simulation.**

I. Introduction

Nowdays the growth of population has given an impact to the travel demand and traffic congestion. In order to prevent congestion at current levels from getting worse then the option is to increase the road capacity. Another option is to develop alternatives that increase road capacity by optimizing the existing transportation system. The latter option tries to focus on Intelligent Transportation System (ITS) infrastructure. Refer to the US Federal Intelligent Transportation System program, the goals of ITS include the following: enhance public safety, reduce congestion, improve access to travel and transit information, generate cost saving to motor carrier, transit operator, toll authorities, and government agencies, and reduce detrimental environment impacts[1]. ITS covering sensor, communication, and traffic control technologies. Generally, all or part of the data that are used in ITS gathered from the vehicle detection and surveillance technologies. Thus, that technologies become an integral part of ITS. There was a study shows that an investment investment in ITS will allow for fewer miles of road to be built, thus reducing the cost of mitigating recurring congestion by approximately 35 percent nationwide[2].

Nowdays, the technologies of vehicle detection and surveillance are being developed and still improved. Those studies include the capability of speed and presence monitoring, traffic measurement, vehicle classification, and weigh-in-motion data[4]. This paper introduces the concept of traffic agent as a new approach for traffic flow measurement. The traffic agent acts as a mobile traffic surveillance agent to monitor and produce a traffic flow measurement value. This research evaluates the efficiency of traffic agent measurement by using computer simulation.

The modified Traffic Cellular Automaton model introduced by Arai et.al., was used as a model base in this research[3]. This model describes more realistic movement of individual vehicle since the model takes into account the component of scope awareness and spontaneous braking behavior.

This paper is organized as follows. Section 2 describes a theoretical aspect of traffic CA model. Section 3 explains the traffic model that used in the simulation study. Section 4 presents simulation process and the results in the form of fundamental diagrams and space-time diagrams. Finally, a summary and conclusion is described with some discussions in section 5.

II. Traffic Cellular Automaton Model and Traffic Agent Concept

A. Traffic Cellular Automata Model

Traffic Cellular Automata (TCA) model is one of the microscopic models that used for road traffic flow simulation. The Cellular Automata (CA) model was improved to capture the road traffic condition. CA model as the basis of TCA model is a discrete computability mathematical model. In comparison with the other microscopic models, CA based models propose an efficient concept to produce a dynamic system simulations. A cellular space and a set of state are two components that compose a CA model. The nearest neighboring cells determine the state of a cell.

The size of all neighborhood cells is same in the lattice. Each of cells must be in one condition, either be empty or occupied by exactly one car. A set of rule applies to each cell from one time step to another as the system iteration. Then, the parallel updating of local cell interaction can show a global complex behavior[3].

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B. Traffic Agent Concept

The traffic agent concept begins from the idea that an agent is embedded in an environment and relies on the spatial and temporal characters of its environment. In this research, the traffic agent was defined as the subject that involve in the road traffic, receives information about its environment via sensors, interprets and process this information become a value of traffic flow. The sensor may be a dashboard-camera that capture the scenery of the subject car. Figure 1 describes the visual concept of traffic agent.

Figure 1. Traffic Agent Concept

The integration of Global Positioning System (GPS) and dashboard camera inform a subject car position and its scenery images. Each of the cars is equipped with GPS, which act as a sensor for the road network.

III. THE SIMULATION MODEL

This simulation study uses two-lane highway model with unidirectional traffic character in periodic boundaries condition. Two-lane model is necessary in order to accommodate the lane changing behavior of driver in the real traffic condition. The TCA model proposed by Arai et.al. has used as the basis model in this research.

A one-dimensional chain of L cells of length 7.5 m represents each lane. This value is considered as the length of vehicle plus the distance between vehicles in a stopped position. A one-lane consists of 103 cells. There are just two possible states of each cell. Each cell can only be empty or containing by just one vehicle. The speed of each vehicle is integer value between v = 0, 1, . . ., vmax. In this model, all vehicles are considered as homogeneous and have the same maximum speed vmax = 5. The speed value number corresponds to the number of cell that the vehicle proceeds at one time step. The state of a road cell at the next time step, form t to t + 1 is dependent on the states of the direct frontal neighborhood cell of the vehicle and the core cell itself of the vehicle. The schematic diagram of lane changing operation shown by Figure 2.

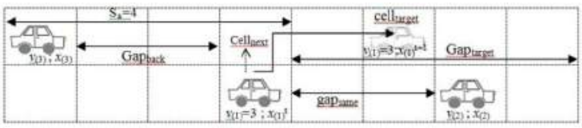

Figure 2. Schematic diagram of lane changing maneuver

IV. SIMULATION RESULTS

The simulation begins with an initial configuration of N vehicles, with random distributions of positions on both lanes. The simulation uses the same initial speed for all vehicle v_{min} = 0 and the maximum vehicle speed v_{max} = 5 cell/time-step. The speed corresponds to the number of cells that a vehicle advances in the iteration. Many simulations are done with the different density ρ. The density ρ is defined as the number of vehicles N along with the highway over the number of cells on the highway L.

A close boundary condition was using during this simulation process. This means during a simulation, the total number of cars on the traffic was fixed. Cars move from left to right. Once a car arrives on the right boundary then it will moves to the left boundary. Since the simulation model assumes symmetry character of the both lanes then the traffic flow characteristics on both lanes are assumed identical.

In this simulation, we analyzed the space-time diagram of traffic condition. Figure 3 shows time-space diagram of road situation captured from registered cars. In this case, the simulation used density 50% and registered car 10%.

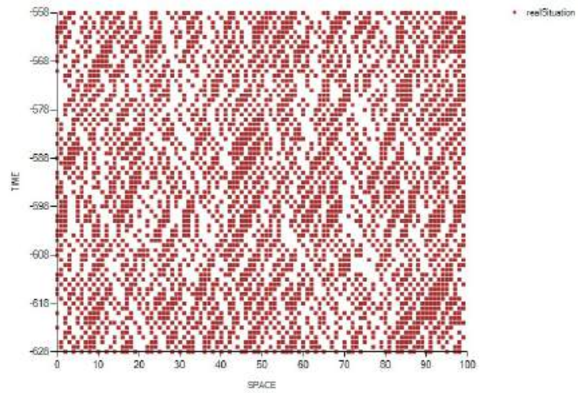

Figure 3. Time space diagram of registered car

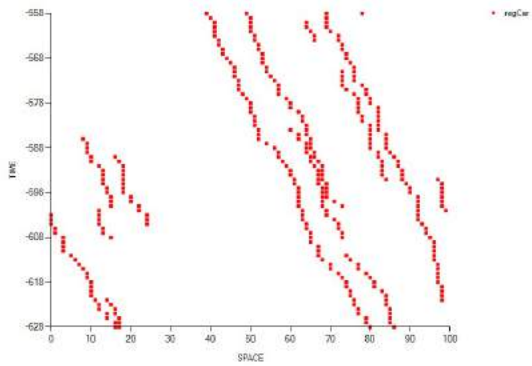

Figure 4. Traffic flow using GPS and camera in Lane 1

Figure 4 and figure 5 show the time space diagram of visualization of traffic flow using information GPS and scenery situation from camera.

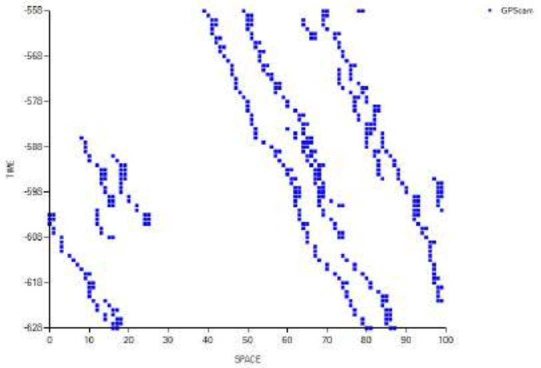

Figure 5. Traffic flow using GPS and camera in Lane 2

The graph of figure 4 and figure 5 present that the accuracy of traffic flow estimation and visualization is closed-related with the number of registered-cars.

The comparison study also conducted to analyze the flow rate that was captured between using only GPS information device and by using the integration of GPS and Camera information. Both conditions have been compared to the real traffic condition. Figure 6 shows the flow rate capture by each condition. Traffic flow is defined as the rate at which cars pass through a point on the certain road, measured in cars/minute.

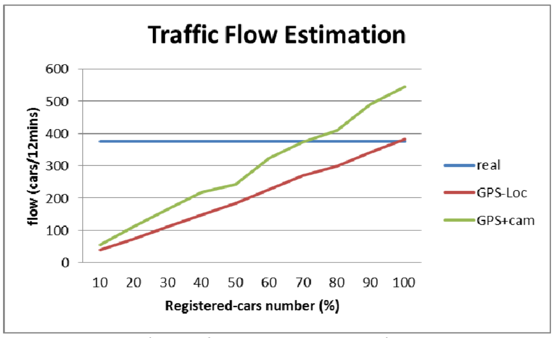

Figure 6. Flow rate comparison

Through this simulation, we also analyzed and compared the traffic flow estimation capability in the situation of using only one traffic agent. This case has tried in three road density conditions: 25%, 50%, and 75% density. Then, the result, include speed and location behavior was plot into a space-time diagram. The comparison results presented by figure 7, figure 8, and figure 9. There is a different degree of its space-time behavior.

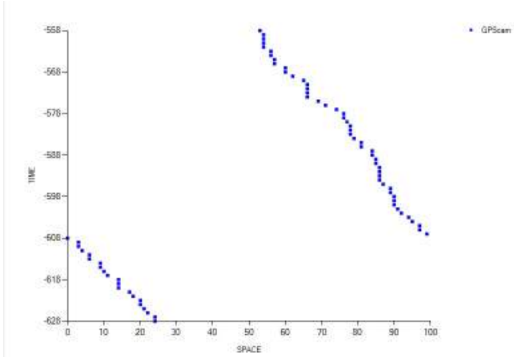

Figure 7. Traffic Density 25%

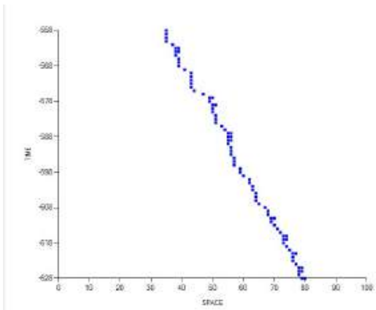

Figure 8. Traffic Density 50%

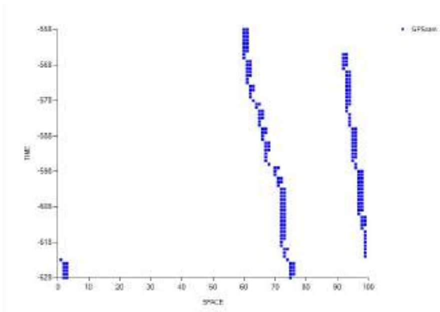

Figure 9. Traffic Density 75%

V. Summary and Conclusion

The simulation study to evaluate the proposed concept of traffic agent utilization is presented through this paper. The followings are concluded through this study:

- Road-length limitation of camera's coverage area makes difficult to estimate number of cars along the road lane. Single snapshot difficult to estimate traffic.
- The traffic flow estimation looks depend on the number of GPS-equipped cars to capture a long portion of road.

- GPS probe data are widely used but suffers from a limited number of vehicles equipped.
- The traffic condition can be distinguished by integrated the information from GPS-camera and convert to space-time diagram.

REFERENCES

[1] L.E.Y. Mimbela et.al, Sumarry of Vehicle Detection and Surveillance Technologies Used, Produced by The Vehicle Detector Clearinghouse, 2000.

[2] Apogee/Hagler Bailly, *Intelligent Transportation Systems: Real World Benefits*, FHWA-JPO-98- 018, January 1998.

[3] K. Arai and S. Sentinuwo, "Driver's awareness and lane changing Maneuver in traffic flow based on cellular automaton model", International Journal of Advanced Research in Artificial Intelligence, Vol. 4 No. 8, 2015.

[4] J. . Wardrop and G. Charlesworth, "A Method of Estimating Speed and Flow of Traffic from a Moving Vehicle," in Proceedings of the Institution of Civil Engineers, Part II, Vol. 3, 1954, pp. 158–171.

DVB-T2 Digital TV Transmitter Infrastucture Optimization

(a case study in the jabodetabek area)

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***Abstract*— the design of digital TV DVB-T2 transmitter infrastructure is done by multiplexing broadcasters in Indonesia, at this time there are many who use the concept of analog TV transmitter infrastructure planning, which in its design avoid two transmitters with the same frequency in the service area as it will cause interference to received signal by the public. With Digital TV DVB-T2 technology allows to design 2 transmitter in the area of service without occurring interference to each other (SFN), it is due to the feature of the guard interval that tolerate varying propagation delay between 2 transmitter antenna when the signal received by public. In this paper carried out the design of a digital transmitter infrastructure that utilizes new features DVB-T2 technology to provide Jabodebatek area TV digital broadcasting. To utilize the DVB-T2 features, three design options was proposed Option I use one transmitter, option II use two transmitters and option III use the four transmitters. All option was designed in order to meet the minimum operational parameters that have been set by the government. After designing all options is done, then compared the results technically and economically. Technical analysis is done by comparing the capacity and coverage of the population of each design, economic analysis is done by comparing the analysis of business each design using a calculation of payback period, NPV (Net Present Value) and IRR (Internal Rate of Return). It was concluded from the results that the design that use 4 transmitters is the most optimal to be implemented in the Jabodebatek area. The optimum design obtained 27.87 Mbps in capacity, 99.9% in population coverage (29,408,063 inhabitants) and IRR 9.64%**

Keywords—Digital TV DVB-T2; transmitter; optimization; design

I. Introduction

In the deployment of digital TV infrastructure in Indonesia, the current television broadcasters still using the concept of infrastructure development with the concept of analog broadcasting. By using analogue broadcasting technology, it is not possible to operate two transmitters or more in a service area which use the same frequency, as it will cause the transmit signal interference with one another. Therefore broadcasters in Indonesia has built a very high towers, as well as using a transmitter with a large transmit power.

Technology digital TV - DVB-T2 allows the television broadcasters to build more than one transmitter at a service area using the same frequency (Single Frequency Network). The presence of a feature in the form of guard interval, will allows the the incoming signal at the receiving antenna are not accepted at the same time, so as not to interfere with each other. Thus, a broadcaster can choose the option to build multiple towers with sufficiently high at some point in a service area and use the transmitter with less power. With proper planning, the forward blind spot at the service area can be minimized.

Therefore, there are several options of a digital TV infrastructure designing new models that can be selected by the broadcasters. It takes a techno economic analysis related to investment of each design option using the concepts and features of digital broadcasting. The design of each option must first meet the operational parameters set by the government.

The contribution of this research is to get a a digital TV transmitter infrastructure design DVB-T2 which can be implemented in a service area that meets the operational parameters of the government and have an optimal quality. Feasibility analysis of business generated in this paper can accelerate the implementation of infrastructure TVdigital transmitter in Indonesia.

The target in this study are as follows. (a) To design three options for digital TV transmitter infrastructure - DVBT2 to the Jabodetabek (Jakarta, Bogor, Depok, Tangerang, Bekasi) service area; (b) to test the design using chirplus BC - software for the fulfillment of minimum operational parameters of the government; (c) to determine the optimal design of the most technically and economically to be implemented in the Jabodetabek service area

II. Literature Review

A. *Regulation of Digital TV Broadcasting Services in Indonesia*

There has been a situation that is less certain in the regulation of digital TV broadcasting in Indonesia. The government has set DVBT2 use of technology, with the result

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that a transmitter (and multiplexer) may be used by some broadcasters. Then the government also established that the company that built the television transmitter is different from companies that do broadcasting. This decision was canceled by the Supreme Court. The solution of this problem is actually very easy, should be made a regulation that stocks of companies which build and has a television transmitter is shared by all broadcasters who use it.

B. Variable Design of Infrastructure Digital TV Transmitters[1],[2],[3]

Geographic location. Determine the geographic location should consider where the population is aiming to reach the transmitter broadcasts. Contours of the earth in the area of population who want to reach a consideration, it will determine the height of the tower is necessary. Line of sight of the transmitter and the population who want to reach a major consideration in the selection of the geographical location of the transmitter.

Direction & Total Panel Antenna. Direction and number of panels will have an effect on beamwidth antenna is formed. Beamwidht shape formed from the number of panels and antennae direction can be as follows

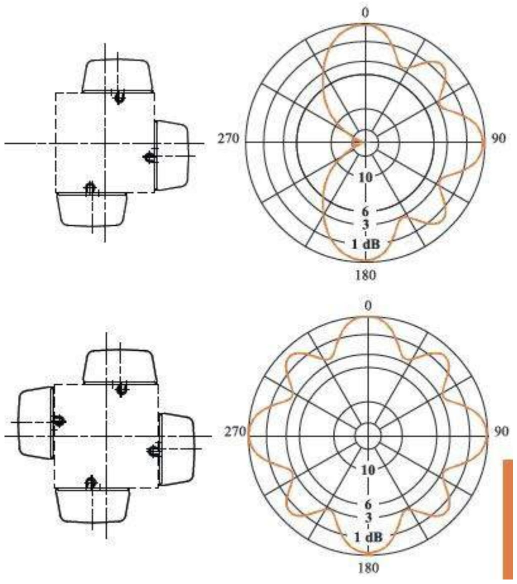

Fig. 1. Beamwidth antenna pattern corresponding number of directions [1]

Power & Gain Antenna. Gain power and antenna will be directly proportional to the range of the transmitter. The greater the power and antenna gain, the more is the range of the transmitter. Power and antenna gain will be used to calculate ERP (effected Radiated Power). ERP = Power Transmitter (dB) + Antenna Gain (dB) -Feeder Loss (dB)

Variable Modulation. The variables are variables that affect the modulation coding digital signals used in digital TV broadcasting. In this case the variables used is variable DVB-T2 signals. These variables will affect the emission range and network capacity. This modulation variables are:

TABLE I. VARIABLE MODULATION OPTION OF DVB-T2 SIGNAL [2]

Variable	*Option*
Constellation Mode	QPSK, 256QAM, 64QAM, 16QAM
Guard interval	¼, 19/128, 1/8, 19,256, 1/16, 1/32, 1/128
Code rate	½, 3/5, 2/3, ¾, 4/5, 5/6
FFT size	1k, 2k, 4k, 8k, 16k, 32k
Pilot pattern	PP1, PP2, PP3, PP4, PP5, PP6, PP7

Rent Price Determination. Broadcast Channels. Determination of rent price of the broadcast channel has been set up by the government in the Regulation of Minister of Communications and Information Technology Number 18/2012, on the Procedure for Calculation of Rent rates on the Implementation of Broadcasters Broadcast Channels Multiplexing. On the regulation stipulated that each broadcaster can rent a broadcast channel to LPS with the following formula: The rental fee = BP (1 + CM) * (Ki / Kt). BP = Operation Cost, CM = Corporate Margin, Ki = rent Channel Capacity (of one broadcaster) and Kt = Total Channel Capacity of the transmitter.

Finance Evaluation. In this paper performed by calculating the payback period and IRR (internal return rate)

III. DESIGN AND SIMULATION [1],[2],[3]

In this chapter proposed three options on the TV transmitter infrastructure design - DVBT2 in terrestrial that can be used by broadcasters to transmit digital TV broadcasting in the service area of Jabodetabek. The three designs must meet minimum operational parameters established by the government. The government stipulated that each multiplexing broadcasters can broadcast digital TV broadcasts that reach more than 70% of the population in each service area, a minimum should be able to accommodate 9 broadcast channel to each broadcast channel capacity of 3 MB. [10]. to analyze the reach of the population and the capacity of their designs, in this paper used helper software that is Chirplus BC. After each draft proposed has met the operational parameters of the specified bill of quantity of each design so that it can determine the cost of the investment.

A. Option 1: Design of Infrastructure of TV - DVB T2 using one transmitter

TABLE II. VARIABLE TECHNIQUES USED IN OPTION 1

Coordinates	6°13'7.00"S 106°43'42.00"T
Tower height	300 m
Directional of antennas	4 (0, 90, 180, 270)
The number of panel antennas	16 panels (= 18.4 dB of gain)
Pilot pattern	10 kW

TABLE III. VARIABLE MODULATION OPTION OF OPTION 1

Variable	***Option***
Constellation Mode	64QAM
Guard interval	1/128
Code rate	2/3
FFT size	8k
Pilot pattern	PP7

From the calculation results obtained that the transmitter ERP in the design of this first option is 436 515 kW. Network capacity is 29.79 Mbps. Plotting the results of the first draft of the options shown in Figure 2.

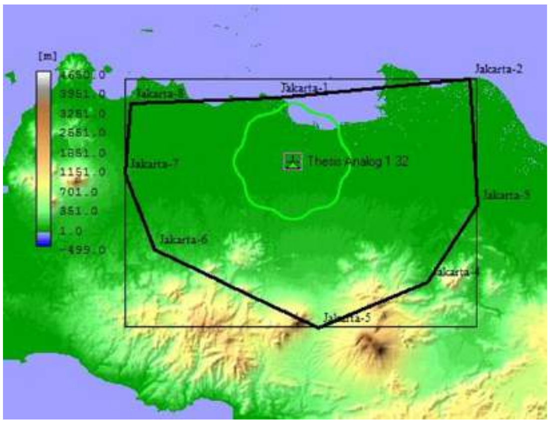

Fig. 2. Transmitter Plotting Design of Option 1

B. *Option 2: Design of Infrastructure of TV - DVB T2 using two transmitter*

TABLE IV. VARIABLE TECHNIQUES USED IN OPTION 2

Coordinates	Transmiiter I : 6°18'25.644"S 106°34'10.665"T Transmiiter II : 6°18'00.684"S 106°43'42.00"T
Tower height	15 m
Directional of antennas	4 (0, 90, 180, 270)
The number of panel antennas	8 panels (= 15.3 dB of gain)
Pilot pattern	5 kW

TABLE V. VARIABLE MODULATION OPTION OF OPTION 2

Variable	***Option***
Constellation Mode	64QAM
Guard interval	¼
Code rate	5/6
FFT size	8k
Pilot pattern	PP1

From the calculation results obtained that the transmitter ERP in the design of this option-2 is 106.898 kW kW. Network capacity is 27.87 Mbps. Plotting the results of the first draft of the options shown in Figure 3.

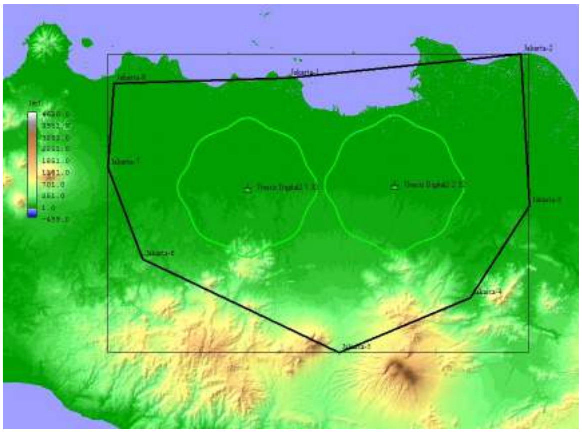

Fig. 3. Transmitter Plotting Design of Option 2

C. *Option 3: Design of Infrastructure of TV - DVB T2 using 4 transmitter*

TABLE VI. VARIABLE TECHNIQUES USED IN OPTION 1

Coordinates	Transmiiter I : 6°12'24.616"S 106°46'39.023"T Transmitter II : 6°17'30.619"S 106°30'01.212"T Transmiiter III : 6°34'14.772"S 106°46'57.732"T Transmitter IV : 6°19'29.176"S 107°02'14.472"T
Tower height	8 m
Directional of antennas	4 (0, 90, 180, 270)
The number of panel antennas	8 panels (= 12.3 dB of gain)
Pilot pattern	5 kW

TABLE VII. VARIABLE MODULATION OPTION OF OPTION 1

Variable	***Option***
Constellation Mode	64QAM
Guard interval	¼
Code rate	5/6
FFT size	8k
Pilot pattern	PP1

The variables used in the planning of option 3 is the same as that used in option 2; ERP in the design of this option-2 is 106.898 kW kW. Network capacity is 27.87 Mbps. Plotting the results of the first draft of the options shown in Figure 4.

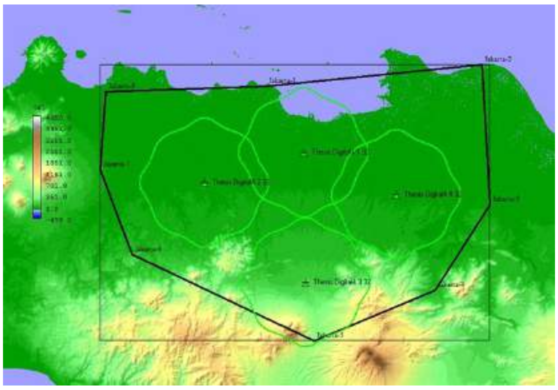

Fig. 4. Transmitter Plotting Design of Option 3

D. *The calculation of the value of the investment*

The calculation of the value of investments are shown in table VIII

TABLE VIII. CAPITAL EXPENDITURE (CAPEX) AND OPERATION EXPENDITURE (OPEX) CALCULATION

Capex (Rupiah)	***Opex (Rupiah)***
Option 1 (1 transmitters): 37,800,303,799	5,649,000,000
Option 2 (2 transmitters): 36,403,932,015	5,906,000,000
Option 3 (4 transmitters): 36,365,513,017	6,428,000,000

E. *The calculation result of the rental rates per channel*

The calculation result rental rates per channel and assumptions used re shown in table IX and X

TABLE IX. THE ASSUMPTIONS USED TO CALCULATE THE RENTAL PRICE

Inflation	6%
Interest rate	7.5 %
Debt / Equity	0
Tax	25 %
Profit margin	35 %
Depreciation	Flat 10 years
Capacity / Channel	3 Mbps

TABLE X. THE CALCULATION RESULT OF THE RENTAL PRICE PER CHANNEL

Option	***Rental Price/Channel (Rupiah)***
Option 1 (1 transmitter)	2,223,530,677
Option 2 (2 transmitters)	2,149,037,159
Option 3 (4 transmitters)	2,161,202,376

IV. ANALYSIS

Resume of the design shown in Table XI. It can be concluded that each option has advantages of each. If that be the primary consideration is the capacity of the channel, then option 1 can be selected. Option 3 (4 transmitters) be the choice when to consider is the percentage of the population that can be reached by digital television boradcast and business benefits achieved.

TABLE XI. COMPARISON BETWEEN THE THREE OPTION DESIGN

Variable	***Option 1 (1 TX)***	***Option 2 (2 TX)***	***Option 3 (4 TX)***
Capacity (Mbps)	29,79	27,87	27,87
Population Reach (%	99,84	99,89	99,90
IRR (%)	88,85	9,12	9,64
Capex (rupiah)	37,800,303,799	36,403,932,015	36,365,513,017
Opex (rupiah)	5,649,000,000	5,906,000,.000	6,438,000,000
Rental Price /Channel (Rupiah)	2,223,530,677	2,149,037,159	2,161,202,376

REFERENCES

[1] Sistem Radio. “Data Sheet UHF Panel Antenna UTV-01/UTV-05”

[2] Lstelcom. “Advanced Broadcast Planning Training Material”.

[3] ETSI (2009). “Digital Video Broadcasting (DVB); Frame structure channel coding and modulation for a second generation digital terrestrial television broadcasting system (DVB-T2)” ETSI : Perancis

Security Requirements Planning To Anticipate The Traffic Flooding On The Backbone Network

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***Abstract*—Along with the increased use of telecommunications networks for social media activities, the risk to information security increases, including the risk of flooding traffic. This paper studied the effect of flooding on the QoS traffic in the backbone network, and needs to be met by security to face the traffic flooding. Security requirements required include: Centralized resource, identity management that supports Identity Recovery, Transaction Logging, Policy Management, mechanisms of basic security (authentication, confidentiality, etc.), minimization of data, and access control, monitoring system , secure Intranet, secure Internet and Extranet, and physical security.**

Keywords—traffic flooding; QoS; security reqiuirements; social media; centralized resource

I. Introduction

With the widespread use of social media services, telecommunications network usage trends shifted resulting surge of data traffic. For telecommunications network operators, network traffic characters changed drastically is increasingly difficult to control, especially with the increasing traffic flooding.

With such a trend, the risk to information security increased, but so far Indonesia has also recorded a poor performance as the number one source of cybercrime attack [1] and most at risk of IT security attacks. [2]

This paper was formulated on security requirements to face the traffic flooding, in order to maintain QoS (Quality of Services).

II. Literature review

A. The Tendency of Cyber Vulnerabilities

The era of smart phones increase the number of cyber vulnerabilities to levels never experienced before. Specifically in Indonesia, which has a bad record of achievement as the country's number one source of cybercrime attack [1] and most at risk of IT security attacks. [2]

Among the disturbances of information security, which took effect on costs incurred operators are attacks that make the operator's network channels flooded traffic undue. This is what will be the focus of security in this paper. Examples of disruption of network traffic caused by the flooding, which once felt by Indonesia of which are errors in BGP (Border Gateway Protocol), which resulted in the collapse of Indosat's network and disrupt communications networks in other countries [3]. Although the disorder is caused by BGP configuration errors because of human error. But the true weakness of BGP can be utilized by the pure wish to attack the network.

This type of attack BGP among Denial-of-Service (DoS) attack on the type of network that is designed to create a network down by flooding the network with useless traffic. Many DoS attacks such as Ping of Death and Teardrop attacks, exploit limitations in the TCP / IP protocol. For all known DoS attacks, there were improvements in software that can be performed by system administrators to limit the damage caused by the attack. However, like viruses, new DoS attacks are constantly developed by hackers. [4] One of the dangers of DoS attacks were the center of attention in this paper is the traffic flooding can disrupt the maintenance of QoS.

B. Effect of Traffic Flooding on QoS (Quality of Service)

In the real conditions of a telecommunications network, each network element typically a queuing network, each node and any route of transmission is essentially a queuing network unit. GoS (Grade of Service), a variable number of traffic engineering is used to provide a measure of the adequacy of the resource group under certain conditions, is essential in the network queue. Although different in definition (QoS defined by the ITU E.800 series, GoS defined by ITU-T E.600 series) some GoS parameters may also be the QoS parameters, such as end-to-end blocking probability, essentially these parameters

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indicate the reliability of the network , GoS parameters to be calculated is the probability and time delay queuing / waiting time. The probability of a query that comes will wait (to experience the waiting time t> 0) can be calculated using the following formula.

$$P\ (t > 0) = D_N(A) = \frac{R.N}{\lceil A(N-A+R)\rceil} \quad (1)$$

By using probability theory, we can find other parameters, including the amount of inquiry that wait before being served (nq), the average time inquiry in the queue before it is served, including those not wait (TQ), the average time inquiry in queue before being served, without which no waiting (tqm), and the average length of time in the system (ts).

$$n_q = D_N(A)(\frac{A}{N-A}) \quad (2)$$

$$t_q = D_N(A)\left(\frac{h}{N-A}\right) \quad (3)$$

$$t_{qm} = (\frac{h}{N-A}) \quad (4)$$

$$t_S = h + t_q \quad (5)$$

The presence of traffic flooding will increase the processing delay of information, which is one of the most important QoS parameters.

TABLE I. CALCULATION OF NETWORK PERFORMANCE (QOS) IN THE EVENT OF TRAFFIC FLOODING IN THE BUSY HOUR

Traffic flooding (%)	***Total traffic (inquiries/sec)***	**P(t>))**	**Nq (Inquiries)**	**Tq (sec)**	**Ts (sec)**
0	22.725	0.1018	0.3055	0.0407	1.0407
5	23.625	0.1490	0.4471	0.0596	1.0596
25	28.125	0.6405	1.9215	0.2562	1.2562
50	33.75	2.0198	6.0594	0.8079	1.8079
100	45	7.1976	21.593	2.8790	3.8790
1000	247.5	193.24	579.73	77.297	78.297

The following will illustrate what is happening to network traffic in the event of flooding on many levels at peak hours. Suppose the total number of servers (N) which will exist in the system is 30. These servers takes an average of one second to process the query, so that the full utilization, the server can process 30 queries per second. Then at rush hour, network utilization rose to 75%, or in other words the offered traffic (A) at rush hour with the service time of one second per query is 75% of 30, which is 22.5 erlang. This causes the average waiting time for inquiry waiting (tqm) only 0133 seconds, the network performs well. But what happens when the traffic flooding?

At the table I presented the results of calculations using the formula above. It appears that the 50% level of flooding is already occurring congestion (inquiry certainly ter-delay). Users already have to wait until more than a second for the inquiry out of the system. At the 100% level of flooding has passed the waiting time of three seconds, the authors think is generally felt on the network Indonesia, which is not even the worst case that might occur.

This further confirms that the implementation of a good security system is needed to reduce the possibility of attack traffic flooding

C. Regulations in Indonesia Related to Cyber Attacks

Act No. 11/2008 on Information and Electronic Transactions

Article 26: (1) Unless otherwise stipulated by legislation, the use of any information through electronic media concerning personal data should be done with the consent of the person concerned; (2) Any person whose rights are violated as referred to in paragraph 1 may file a lawsuit for damages caused by this Act.

Government Regulation No. 82/2012 on Implementation System and Electronic Transactions. Article 6, paragraph 1: Electronic System Operator shall ensure: (a) Availability of information security agreement on information technology services used; (b) The availability of a service level agreement; (c)

Security of information and means of internal communication are organized. Article 55, paragraph 3: Data Preparation of Electronic Signatures as referred to in paragraph (1) and (2) must meet the following conditions: (a) The whole process of data Making Electronic Signatures guaranteed security and confidentiality by the organizers of the Electronic Signatures or Support Services Signs Electronic hand; (b) If using a cryptographic code, Data Preparation of Electronic Signatures must not be easily known from Electronic Signature verification data through a certain calculation, within a certain time, and with a reasonable; (c) Electronic Signature Creation Data stored in an electronic medium that is in the possession of Hand markers; (d) Data relating to the Bookmarks Hands mandatory stored on the premises or means of data storage, which uses the system reliable belonging Organizer Electronic Signatures or Support Services Electronic Signatures which can detect the change and meet the following requirements: (a) Only persons authorized which can enter new data, alter, exchange, or replace the data; (b) the identity information can be checked for authenticity markers Hands; (c) other technical changes that violate security requirements can be detected or known by the organizers; (d) Hand markers shall be responsible for maintaining the confidentiality and Data Preparation of Electronic Signatures.

Bank Indonesia Regulation No. 9/15 / PBI / 2007; on the application of risk management in the use of information technology by the commercial banks.

Article 18 paragraph 2 section B for service providers Information Technology: (a) As an affiliated party, the service provider shall ensure the security of all confidential information including bank and personal data of customers; (b) Part of the regulation, indirectly supporting the existence Huber, for example, regulations related to information and electronic transactions such as UUITE, PPITE, PBIMRTI09 / 15/2007.

Emphasis on the importance of information security and the necessity Data Center and Disaster Recovery Center in Indonesia which apply to all providers of electronic services to the public to encourage the need for a reliable security system.

With the current regulations, each telecommunications network operator that has access to customer information, either individual or corporate customers, shall maintain the confidentiality of such information, except on the orders of a judge or other legitimate authority in accordance with the provisions of the legislation. In relation to security of data, the current telecom service providers as third party service providers are not bound to the law or the obligation to maintain the security of user data in Indonesia. [8] So a good security system can be the answer to the application service users are often concerned about the security and confidentiality of their data.

III. Security Requirements Planning To Face The Traffic Flooding

A. *Research Scenarios*

The methodology used in planning security requirements to face the traffic flooding is as follows: (1) Defining the security system and the objectives to be achieved, is done by specifying the conditions to be achieved where the security is created for the business and the network between telecommunications network operators and service providers social media; (2) Understanding the functions and workings of the components of network security architecture, carried out by gathering the facts related jaringann security architecture; (3) to analyze and formulate requirements network security components are assessed for compliance with the security needed to deal with traffic flooding.

B. *Planning and Implementation Process*

In the planning stage, there are three processes that occur in parallel. Then the process of implementation is derived from the plan. Here's an explanation of each process.

- Designing a national backbone network. In this process analysis in determining the network link by considering the capacity and efficiency of the network infrastructure, which is included in the field of telecommunications engineering. The process is reported in paper [XXX]
- Strategy and Regulation at national backbone network. Analysis in determining the functionality and business model of network Huber which is the embryo of a national backbone network. The process is reported in paper [XXX].
- Security Requirements Planning of the national backbone network. In this process analyzes the effect of flooding on the QoS traffic, as well as an analysis of the security components that must be met to maintain QoS

  Implementation process. In this paper not reported, consists of two things:

- Planning and implementation of data-server and backbone. In this process involves planners of links required, along with strategies and regulations.
- Implementation of the system software security

C. *Security Requirement Planning of National Backbone Networking*

1) **Resources Planning.** At the national backbone network, the existing services must be guaranteed by a QoS targets, therefore the recommended network security model is a model with a centralized resource. Each policies, procedures, and processes defined by the generic network security. In the case of the fulfillment of QoS, flexibility and autonomy is less needed with similar objectives. All safety aspects will be organized to meet a particular vision, and meet a collection of policies.

2) **Identity Management.** Common identity management applied in the least privileges principle, the principle of practicing restrictions on access to minimal resources in each module, in this case the identity that has been built, without disrupting the work and the right to identity. For the implementation of identity management to the principle of least privileges, there are several aspects that need to be considered and met. (a)**Functional Requirements**. On identity administration, in addition to the standard features, Identity Recovery mechanisms need to be applied so that when a part of the digital identity stolen by vandals, the identity can be recovered immediately. Then of course there needs to be management history for checking trace data, as well as applied too Transaction Logging. Control must also be met with Policy Management to manage identity / profile which is right for a particular transaction. (b)**Security**. Basic security mechanisms must be met, such as authentication, confidentiality, integrity, and non-repudiation. Because the core mechanism of the identity management system is to ensure the authenticity of the user. For more services require more security, multifactor authentication or authentication by a factor of more than just the user-id and password to use. (c)**Privacy**. One guard is the minimization of data privacy. Necessary to ensure that the data is stored and processed in the service provider data is needed. This can prevent the release of data that is unnecessary or sensitive to an unauthorized person., So that the risk of privacy breaches is reduced significantly. (d)**Access Control**. Restrictions on access to resources. Examples of access control list that contains the subject, object, and access privileges contained in the table II.

3) **Monitoring System.** Recording activity (log) that utilizes information technology, let alone assault, unauthorized scanning behavior must be monitored because it can lead to security breaches. Centralized log management system or security event management is an important thing because one of the

steps taken by the intruder when successfully entered into a system is to remove the entry in the system log. Performed in the monitoring system include:

• Host-based intrusion detection

• Network-based intrusion detection

• Event-logging management

• procedures review & analysis log

• Conduct a review of security events

• Compliance assessment evaluation

• Penetration testing

4) **Secure Intranet.** In securing intranets, there are several things that can be applied, including physical separation, segmentation, VLAN, Firewall and IDS. These things are applied to the principle of least privileges. As well as efforts to minimize the risk of infiltration of the intranet National Network.

5) **Secure Internet and Extranet .**Security aspects of Intranet and extranet these include :)(i) **Perimeter Defense**. Parts are connected to the outside world (Internet), must be separated from the part that is connected to the (intranet). Generally done by using a firewall and IDS. (ii)**Demilitarized Zone.** Services that can be accessed by the public should be put in place that do not interfere with the internal network. (iii) **Extranet.** Extranet access services implemented with VPN (Virtual Private Network). VPN must use encryption to secure the data. (iv)**Deep Packet Inspection.** Packet inspection methods applied in China need to be applied to national backbone network to monitor incoming and outgoing traffic Indonesia. This is the answer to improve Indonesia as the number one source of cyber-attacks.

6) **Physical Security.** Security information technology infrastructure physically, that security in access to the office environment and buildings, access to a particular room in a building (egg data centers), access to the device (egg servers) within data center, and access to the workstation and its peripheral. Examples its security by implementing smartcard, clear desk policy, surveillance cameras and biometric locks.

IV. CONCLUSION

Traffic flooding to some extent will affect the reliability of the network and degrade QoS, so it is necessary to formulate requirements in security systems that minimize the risks and impacts.

The various needs of the Huber system security including:

• Centralized resource

• Identity management

• Monitoring System

• Secure Intranet

• Secure Internet and Extranet

• Physical Security

REFERENCES

[1] H. Sigit, "Telecommunications Traffic: Technical and Business Considerations", 2012. (*references*)

[2] H. Sigit, "Telecommunication Service and Experience Quality", 2012. (*references*)

[3] I.S. Jacobs and C.P. Bean, "Fine particles, thin films and exchange anisotropy," in Magnetism, vol. III, G.T. Rado and H. Suhl, Eds. New York: Academic, 1963, pp. 271-350.

[4] K. Elissa, "Title of paper if known," unpublished.

[5] R. Nicole, "Title of paper with only first word capitalized," J. Name Stand. Abbrev., in press.

[6] Y. Yorozu, M. Hirano, K. Oka, and Y. Tagawa, "Electron spectroscopy studies on magneto-optical media and plastic substrate interface," IEEE Transl. J. Magn. Japan, vol. 2, pp. 740-741, August 1987 [Digests 9th Annual Conf. Magnetics Japan, p. 301, 1982].

[7] M. Young, The Technical Writer's Handbook. Mill Valley, CA: University Science, 1989.

[8] E. Vinietta and S. Haryadi, "Hubber Strategy and Regulation," in ICWT Conference, Manado, 2015.

Design of Indonesian Hubber Communication Network

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***Abstract*—Hubber [5] Communication Network in Indonesia currently has never existed. This paper reports the design of the national backbone network that can be used as a hubber communication network.**

Design and simulation using software Net2plan, based on abstract components such as nodes, links or route traffic. The design allows to be applied in almost every technology.

Results of the simulation, a hubber communication network in Indonesia require 15 vantage point, namely Medan, Palembang, Pontianak, Banjarmasin, Jakarta 1, Jakarta 2, 3 Jakarta, Bandung, Yogyakarta, Surabaya, Denpasar, Makassar, Manado, Ambon and Biak. The offered traffic of 18.372 Tbps, delay point-to-point with the largest is 32 ms average network delay is 7.71 ms that makes this network is able to follow the growth of demand for real time voice and video over IP

Keywords—hubber; national backbone; vantage point; Net2plan; offered traffic

I. INTRODUCTION

Referring to the Presidential Decree No. 96/2014 on the Broadband Development Plan Indonesia development program instructs the data center and network integrated government agencies to establish a secure intranet and integrate government data center. Presidential Decree implemented by the Ministry of Communications and Information Technology in the form of program Network Development Government Agencies Integrated to integrate network and datacenter all government agencies for purposes of e-Education, e-Logistics, e-Health, e-procurement, e-government and telecommunications for emergencies and disasters / Public Protection and Disaster Relief (PPDR).

Existing conditions today, government agencies utilizing the backbone owned mobile operator, in other words, using the same communication medium used by the public. Based on the statistical report released by the ITU, the development of the use of information technology and telecommunications, in particular on access and ownership of technology-based cellular mobile telephone, reaching the point of being awesome and increasingly rising from year to year. In the context of Indonesia, is experiencing growth as well as access to significant growth in the availability of information and communication technologies in the access category-based mobile telephone services cellular network. But in recent developments, the mobile operators we are facing problems due to the development of business networks and the proliferation of services OTT (over the top service). Nowadays, more and more Internet use cannot be separated from public life. The development of some technologies that are interrelated simultaneously as a mobile internet connection, smartphone, telecommunications network infrastructure, the realization of hubber communication network, and the development of various applications bring a new trend that is OTT (over-the-top) service.

The purpose of this study was

- Designing a national backbone network for the purpose of hubber communication network in Indonesia.
- Comparing the centralized network planning scenarios (Vantage Point) and decentralized to be applied in Indonesia.
- Designing a national backbone network that meets the requirements of ITU G.114
- Designing the network capacity needs to be installed for network planning until 2020

II. LITERATURE REVIEW

A. *Internet Access Network Hierarchy*

Internet architecture which is implemented since the early 2000s based on a multi-tier hierarchical structure. Tier 1 ISP (Internet Service Provider) located at the top of the hierarchy, followed by Tier 2, regional ISPs and ISP Access at the bottom of the hierarchy that connects to the end users. In this scheme, Tier 1 ISP connected to another ISP and offer transit services to other ISPs in the layer below it. Content is distributed through ISP Access or, in some cases, through an ISP located at the points were favorable. In the hierarchy, traffic required to

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ascend and then descend to reach the end users [1].

Tier 2 ISPs collect payment from a Tier 3 ISP's bandwidth and traffic volume passing through, this relationship is very expensive and inefficient also add latency is not required even if the source and destination of traffic in close proximity geographically. As an alternative to the link-based transit, ISPs also use the connective tissue of the Internet Exchange Point (IXP) to interconnect. Internet Exchange Point serves a point location where ISPs may exchange traffic without bandwidth or volume limits. Both networks are usually located in the same city, thereby reducing the latency interconnection occurs. Internet Exchange Point in Indonesia is Indonesia Internet Exchange (IIX) managed by APJII residing on the classification of the National Backbone Network Operators in the hierarchy of the Internet as shown below:

Global Internet Exchange (IIX) is a global network-based peer Internet exchange points (IXP or IX) used for traffic exchange point between Proviers Internet Service Provider (ISP) and a larger network again. GIX serving infrastructure (routers, Switches, and other support tools), which serves as the association of the exchange of traffic between ISPs. This reduces cost, redundant, fail-safe, route-based and low latency alternative to the more expensive transit links from ISP higher. Global Internet Exchange is a higher classification from the National Backbone Network Operators on the hierarchy of the Internet.

III. BASIC THEORY FOR INTERNET BACKBONE NETWORK PLANNING

The study began with a search for data in the form Palapa Ring network scheme to be built in Indonesia, and user data internet services in Indonesia. Which would then be used to calculate traffic and determine placement Huber in order to meet the predetermined criteria.

A. *Net2plan*

Net2plan is an open-source software based on Java devoted to planning, optimization and evaluation of communication networks. Originally used as a tool to help teaching courses communications network planning. Has now been converted into a network planning tool qualified for use by academia and industry.

Net2plan built on an abstract representation of the network, called network plan, based on seven abstract components, namely: the node, link, route, traffic needs, protection segment, shared-risk group and the network layer. Representations network-agnostic technology, which means it does not lead to a vendor, so it can be adjusted for Net2plan network planning in any technology. Information specific technologies can be introduced in the representation of the network through a user-specified attributes attached to one of the abstract components mentioned earlier. Some attribute names have been corrected to facilitate adaptation of known technology that is an IP network.

Net2plan implemented as a Java library with the command line and graphical user interface (GUI). This GUI is used for laboratory sessions as an educational resource, or for a visual inspection of the network. Command line interface devoted to in-depth research studies, utilizing batch processing or large-scale simulation features. Therefore Net2plan is a tool that can be used in a broad spectrum of users, the user industry, research and academia.

Some of the tools provided by net2plan to perform network planning is:

Offline network design: designed to evaluate network planning generated by an algorithm that has a built-in or user-defined, evaluate several aspects such as network topology, the traffic, the capacity of the link, route protection and others.

Traffic matrix generation: designed to help users in the process of creating and normalize the traffic matrix.

Resilience simulation: simulates the operation of the network, where the failure of link and node appear randomly to evaluate the performance of availability of the scheme of protection / restoration are built-in or user-defined.

Time-varying traffic simulation: simulates the operation of the network, where the volume of traffic that are required will change with time depending on the pattern of change built-in or user-defined.

Connection-admission-control simulation: simulates the operation of the network, where traffic demand is the source of the connection request. Algorithms used to evaluate the connection-admission-control that dynamically allocates resources to the connection request.

Reporting: Net2plan allows creating reports based algorithm built-in or user-defined, from any network design. Report generation tool is integrated in all the previous functions, so it is possible to create reports collecting measures of performance in every aspect.

Calculation traffic is using teletraffic theory which usually requires the capacity of a network using Erlang. Some things that need to be considered in the simulation using this Net2plan are:

- The average length of the packet is 340.46 bytes
- Binary rate per Erlang is 1 Gbps
- Precision factor is 3 digits after the decimal point
- Follow the propagation speed of light in fiber propagation speed is 2*(10^ 8) m / (s^2)

A national network for the purpose of hubber communication network is a network that is built based data communications based on IP, technically could be interpreted as a network that uses a lot of data servers in various locations spread is also supported by the telecommunication networks appropriate, so can offer the best quality to end users.

Network platform closer to the existing content on the primary server to the server that is closer to the end users. Because there is a cache in Huber network server that will store the content that was originally located on the main server. The content of the main server does not need to be transferred over the network whenever end users make requests, the Cache contents are copied to the cache servers located near end users who make requests without affecting other tissues, then every request over the content will be served by the cache server.

There are two alternative strategies for implementing national hubber communication network network. The first strategy, built a large content distribution centers in strategic locations and linking centers of high-speed connection to the ISP, which formed a distribution center at Vantage Point (VP) are at once close to the major ISPs. And the second strategy, the network using a cache server that is in the ISP.

The first strategy has a simple management overhead, but at the expense of response time. The second strategy has optimal performance in terms of response time because of its location so close to the end users, but management and more complex server deployment. Modeling in this study using a vantage point, where each vantage point has a coverage area that will be served. So we get the advantage in terms of a very high QoS, Cost of Network Infrastructure that is more efficient than putting in the ISP cache.

B. *Traffic Forecast*

Results forecast of telecommunication traffic in Indonesia is shown in the graph below

TABLE I. INTERNET TRAFFIC FORECAST

Year	*# Data (GB)*	*# User (Million)*	*Data per User per Year (GB)*
2010	22,927	86.1	0.266
2011	50,887	111.5	0.457
2012	121,747	127.0	0.958
2013	251,204	148.3	1.693
2014	482,207	156.6	3.080
2015	773,191	182.0	4.248
2016	1,161,422	200.8	5.783
2017	1,636,350	219.6	7.450
2018	2,197,977	238.4	9.218
2019	2,846,300	257.3	11.064
2020	2,912,154	276.1	10.548

IV. PLANNING A NATIONAL NETWORK FOR THE PURPOSE OF HUBBER COMMUNICATION NETWORK IN INDONESIA

A. *Vantage Point Determination*

Fifteen locations were chosen vantage point is:

1. Medan, which will serve traffic from Aceh, North Sumatra, West Sumatra and Riau.

2. Palembang, which will serve traffic from Jambi, South Sumatra, Bengkulu, Lampung, Bangka Belitung and Riau Islands.

3. Jakarta 1, which will serve the traffic of Jakarta and Banten as well as being a liaison with the network of Singapore.

4. Jakarta 2, which will connect traffic from the central and eastern part of Indonesia with a network of Singapore.

5. Jakarta 3, which will connect traffic from the western part of Indonesia with a network of Singapore.

6. Bandung, which will serve traffic from West Java

7. Yogyakarta, which will serve the traffic of Central Java and Yogyakarta.

8. Surabaya, which will serve the traffic from the area of East Java.

9. Denpasar that will serve traffic from the area of Bali, NTT and NTB.

10. Pontianak, which will serve traffic from West Kalimantan, Central Kalimantan.

11. Banjarmasin, which will serve the traffic from South Kalimantan and East Kalimantan.

12. Makassar, which will serve the traffic from South Sulawesi, West Sulawesi and Southeast Sulawesi.

13. Manado, which will serve the traffic from North Sulawesi, Gorontalo and Central Sulawesi.

14. Ambon, which will serve the traffic of Maluku and North Maluku.

15. Biak, which will serve the traffic from the area of West Papua and Papua.

Selection of points is based on the vantage point Palapa Ring network to be built in Indonesia, along with the number of users who will be using OTT services.

Palapa Ring is shown in figure 2 and the national backbone topology is shown in figure 3

Fig. 1. National Government Network (Palapa Ring)

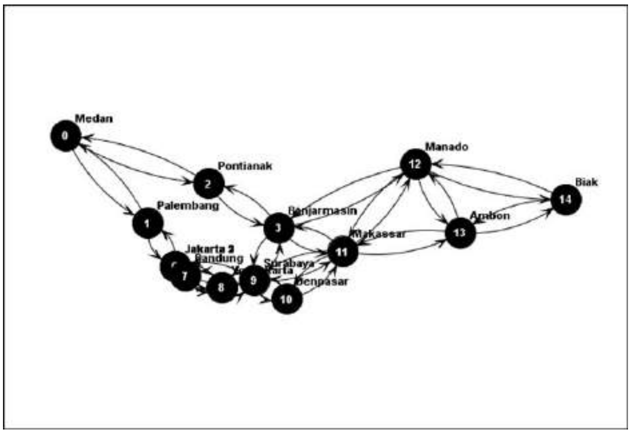

Fig. 2. National hubber communication network

B. Topology & Network Capacity Analysis

Design a national hubber communication network is a network that is a point of liaison between the various points of the Internet network Indonesia and also the global Internet network, so the network is designed to be able to accommodate all the traffic in Indonesia.

Number of vantage points in the design there are 15 dots, each representing their respective areas as well as some surrounding areas is still possible to be covered within a point of vantage point. The number of links is installed in the network are 46 pieces of link, which connects each point vantage point is bidirectional. The average number of links per node (average node in / out degree) is 3.067. Average length of biggest roads between all pairs shortest path (network average diameter) is 5 hops, 5330.048 km, and 26,7ms. The average total installed capacity links on the entire network is equivalent to the Erlang 58437.376 5,84E + 4Gbps. An average of 805.450 km link distance, the average distance between the nearest 2,514 hops hop is equivalent to 1886.905 km

TABLE II. NETWORK ANALYSIS

Metric	*Value*
Number of nodes	15
Average number of links	46
Average node in/out degree	3.067
Average network diameter (ms)	26.7
Average total capacity installed (Erlangs; bps)	58437.376; 5.84e+13
Average link distance (km)	787.304
Network diameter (hops)	5
Network diameter (km)	5330
Average shortest path distance (hops)	2.514
Average shortest path distance (km)	1886.905

C. Traffic Analysis

Design a national hubber communication network is a network that is a point of liaison between the various points of the Internet network Indonesia and also the global Internet network, so the network is designed to be able to accommodate all the traffic in Indonesia.

Number of vantage points in the design there are 15 dots, each representing their respective areas as well as some surrounding areas is still possible to be covered within a point of vantage point. The number of links is installed in the network are 46 pieces of link, which connects each point vantage point is bidirectional. The average number of links per node (average node in / out degree) is 3.067. Average length of biggest roads between all pairs shortest path (network average diameter) is 5 hops, 5330.048 km, and 26,7ms. The average total installed capacity links on the entire network is equivalent to the Erlang 58437.376 5,84E + 4Gbps. An average of 805.450 km link distance, the average distance between the nearest 2,514 hops hop is equivalent to 1886.905 km.

TABLE III. TRAFFIC ANALYSIS

Metric	*Value*
Number of demands	210
Average total offered traffic (Erlangs; bps)	6068.788; 6.07e+12
Average offered traffic per node pair (Erlangs; bps)	28.899; 2.89e+10
Average blocked traffic (%)	0
Average over-subscribed traffic (%)	0.038
Total offered traffic (Erlangs; bps)	5340.982; 5,340,981,578,348.355
Total carried traffic (Erlangs; bps)	5340.982 (5340981578348.355)
Average offered traffic per demand (Erlangs; bps)	25.433; 25,433,245,611.183
Average carried traffic per demand (Erlangs; bps)	25.433 25,433,245,611.183
Blocked traffic (%)	0
Avg. number of hops	2.447

D. Delay Analysis

As seen from the table III, the average delay of the network designed is 8.27 ms which is an end-to-end average delay in getting by a randomly selected packet in the network. Delay a link of this network is the largest delay from Banjarmasin to Manado with delay reaching 9.87 ms.

Delay end-to-end delay of the largest Biak to Palembang is 35.6 ms. According to ITU recommendation G.114 IP network delay one-way trip from a point to another point for real-time communication quality is less than 150 ms or 300 ms for international calls [2], one-way latency delay from a point to another point in interactive video communication also requires no more than 150 ms [3].

Delay of 35.6 ms for IP networks is now entering the comfort standard voice and video communication in realtime recommended by the ITU. Because the convenience of the users as well as video phone service will be interrupted when the round trip delay of 250 ms, and the phone will begin to talk to each other simultaneously [4]

TABLE IV. DELAY ANALYSIS

Metric	***Value***
Number of demands	210
Average total offered traffic (Erlangs; bps)	6068.788; 6.07e+12
Average offered traffic per node pair (Erlangs; bps)	28.899; 2.89e+10
Average blocked traffic (%)	0
Average over-subscribed traffic (%)	0.038
Total offered traffic (Erlangs; bps)	5340.982; 5,340,981,578,348.355
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Average offered traffic per demand (Erlangs; bps)	25.433; 25,433,245,611.183
Average carried traffic per demand (Erlangs; bps)	25.433 25,433,245,611.183
Blocked traffic (%)	0

V. PLANNING A NATIONAL NETWORK FOR THE PURPOSE OF HUBBER COMMUNICATION IN INDONESIA

National hubber communication network architecture with the following results.

1. The number of vantage points is 15 points, namely Medan, Palembang, Pontianak, Banjarmasin, Jakarta 1, Jakarta 2, 3 Jakarta, Bandung, Yogyakarta, Surabaya, Denpasar, Makassar, Manado, Ambon and Biak.

2. Diameter Network 5 hops which is equivalent to 5330 km.

3. Total offered traffic is Erlang 5340.982 equivalent to 5,34Tbps.

4. Total Carried traffic is Erlang 5340.982 equivalent to 5,34Tbps.

5. Blocked traffic 0%.

6. Delay biggest link is 9.87 ms from Banjarmasin to Manado and vice versa, delay end-to-end network is the largest in 35.6 ms from Biak to Palembang. End-to-end delay is still meet the standards of real-time communications using IP such as VoIP and video conferencing.

REFERENCES

[1] C. S. a. E. W. B. Pablo Rodriguez, “Analysis of Web Caching Architectures: Hierarchical and Distributed Caching,“ ACM TRANSACTIONS ON NETWORKING, Bd. 9, Nr. 4, pp. 404-418, 2001.

[2] Cisco, “Quality of Service for Voice over IP,“ 16 April 2001. [Online]. Available: http://www.cisco.com/c/en/us/td/docs/ios/solutions_docs/qos_solutions/QoSVoIP/QoSVoIP.pdf. [Diakses pada 1 Juni 2015].

[3] Ciscopress, “Ciscopress Quality of Design Overview[,“ 17 Desember 2004.[Online].Available: http://www.ciscopress.com/articles/article.asp?p=357102&seqNum=2.

[4] Sans Institute, “Voip Latency QoS voice IP,“ 2004. [Online]. Available: http://www.sans.org/reading-room/whitepapers/voip/latency-qos-voice-ip-1349. [Diakses pada 2015 Juni 1].

[5] E. Vinietta and S. Haryadi, “Hubber Strategy and Regulation,” in ICWT Conference, Manado, 2015.

Hubber Strategy and Regulation

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***Abstract*—Hubber system is proposed as a solution for Indonesian Telecommunications problem in the era of global trend, where the sustainability of Indonesian Telecommunications operators is threatened as the result of the evolvement of OTT communication service (IP-based) that substitutes the traditional operator's communication services (non IP-based), SMS and telephony. The definition of hubber as proposed is: a hub for IP-based communication between OTT provider or global CDN with Indonesian network provider that is embodied in Telecommunications network and distributed data center infrastructure. As a Point of Interconnection, hubber has administrative aspect, business aspect, and technical aspect. Revenue forecasting has been done to Telkomsel – the largest Indonesian operator, and the result shows that Telkomsel's revenue will go down for the first time in 2015. To achieve positive 7% revenue growth, Telkomsel has to set its revenue target from hubber of 4.83 trillion Rupiahs in 2015 with a steady increase in annual growth and reaching almost a tenfold increase in value, 46.14 trillion in 2020.**

Keywords— hubber, OTT, Telecommunications operators

I. INTRODUCTION

The use of Internet has been growing rapidly in recent decades. The development of technologies has led to a trend of new service: OTT (over-the-top) service.

While hubber concept has not been implemented in Indonesia, most Telco services through OTT can be consumed with low price or even free of charge because OTT providers rely their profits from advertisement, application sales, and in-app purchase. OTT services have been widely substituting the legacy services of telecom operators, especially telephone and SMS. Meanwhile, operators need to invest heavily to comply the traffic demand as the result of OTT trend, because the use of OTT is very intense and the delivered contents need very high bandwidth. In contrast, most Indonesian Internet services that are offered by Telco operators and popularly used are time based, for example, unlimited volume per month, so traffic from OTT does not contribute much to Telco operators' revenue.

If this trend continues, the national Telco operators will suffer losses, even though they are also needed by OTT providers to distribute their services to their end consumers. Meanwhile, in Indonesia, there has not been an effective regulation related to the relationship between the Telco operators and the OTT service providers.

II. STUDY OF THE OTT

Over-the-top (OTT) service refers to the form of services provided via the Internet, by utilizing the traditional service provider network canal [1]. In line with the above definition, reference [2] defines OTT providers as service providers offering only Telecommunications services without dealing with Telecommunications networks needed to distribute them, and only rely on the worldwide Internet network.

OTT services can be classified into communication services and content services [1]. OTT providing communication services is currently a problem for the Telco because those services substitute Telco's legacy communication services such as telephony and SMS.

With the presence of OTT, traffic on the network is no longer dominated by the voice traffic, but by the data [3]. These changes in characteristics of the traffic changes the revenue of Telco operators. Telco operators must continue to pay to increase data capacity in order to maintain service quality. Meanwhile, the gain by Telecommunications operators tend to have linear growth. Increased costs are not proportional to the increase in revenue. This makes Telco's existences are being threatened by OTT.

The emerging campaign of net neutrality principle became one factor that makes Telco operators do not have many options to react to the threat of OTT. The definition of net neutrality proposed by Tim Wu is "a principle behind an anti-discrimination in the networks that give users the right to use any network attachments or applications as long as they are not harmful." There is a contradiction between Telco's business benefit and neutrality of the network [4].

III. STRATEGY PLANNING PHILOSOPHY

Increasing use of the Internet service in Indonesia in recent years turned out to be contrary to the national Telco's revenue growth. According to data from Indonesian ICT ministry in 2013, the three major cellular operators in Indonesia, Telkom, Indosat and XL has been experiencing a dominant negative total revenue growth from 2007 to 2011 [5].

In particular, Telkomsel as the largest mobile Telco operator in Indonesia also experienced a decline in revenue growth of a few years back. Analysis of operational data from

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the annual report in the period of 2008 to 2014 revealed that the cause of the decline in revenue growth of Telkomsel is the same as the cause of a decrease in revenue of Telco operators in many countries. The growth of SMS and telephony service usage has been decreasing, on the other hand, data services usage has been experiencing rapid growth. As a consequence, a high investment for data service infrastructures should be performed, and that can be seen from the growing number of BTS 3G/4G from year to year, while revenue from data services, which generally are time based and not volume based, has been slowly growing, and those conditions cause a decrement in ARPU. Because the growth of users tends to decline, the growth of revenue is predicted to be declining as well.

OTT is a service that occupies the data pipe belonging to Telco operators. To access global OTTs, the access to OTT data centers abroad is needed. Communication with the OTT which mostly have a client-server protocol (with or without CDN) requires that a message first sent OTT server and after that is sent to the destination, although the location of the sender and recipient are much closer geographically compared to OTT server location. That is one factor that makes communication in Indonesia inefficient.

OTT-related problems are existed not only in Indonesia, but also in most countries in the world that have Internet access, both developed and developing countries. According to the reference [6], the countries in the world have been making and implementing national policies and regulations regarding OTT services according to their jurisdictions, which tries to answer the problems posed by the presence of OTT. Regulation that is applied in various countries has its respective context, motivation, and environment.

From the philosophy explained above, hubber system design has a main objective to maintain Telco operators' revenue growth by creating new sources of revenue and reducing cost.

In addition to the main objective, hubber is also created to meet additional objectives, namely:

- Enhancing information security and national sovereignty
- Enhancing the resilience of the national economy
- Improving the quality of national Telecommunications

IV. Strategy and Regulation Planning for Hubber System

A. Strategy Planning in Implementing Hubber

Hubber system is required to bridge the traditional Telecommunications era to the future Telecommunications era. In the era of traditional Telco, access and services are provided solely by Telco operators, and customers only can use services that is provided by the operator which they are subscribed to. In the future of Telecommunications era, customers can use various services from various service providers (SP) as well. This means that OTT services will be shifted upward and become the 'top service'. These changes are illustrated with figure IV.1 and IV.2.

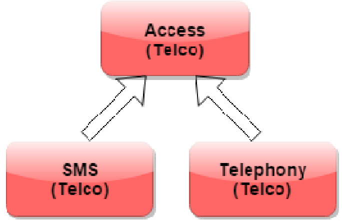

Figure IV.1 Traditional Telecommunications

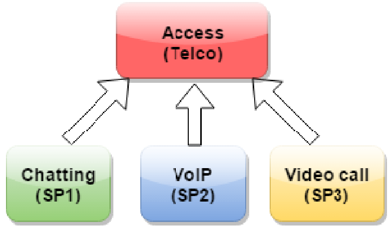

Figure IV.2 Future Telecommunications

B. Regulation Planning of Hubber System

Hubber network regulation proposed in this paper covers: the scope of hubber, hubber definition, hubber roles and responsibilities, and hubber business model.

- *Scope of Hubber*
  1. Hubber is an IP-based communications link
  2. Hubber connects Indonesian Telco operators with OTT or global CDN
  3. Hubber is a local CDN
- *Hubber Definition*

Definition of hubber proposed in this paper are: a hub for IP-based data communications between OTT or global CDN provider with Telecommunications providers that provide Internet access to users in Indonesia, which has a special infrastructure consisting of Telecommunications infrastructure and distributed data center. In hubber system, OTT service users in Indonesia subscribe to internet service from Telco operators to get access, and Telco operators connect with OTT services provider through the hubber. Basically, hubber does not manipulate or interpret the data packets communicated between users and OTT providers.

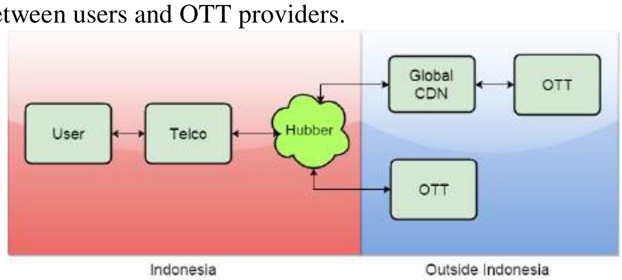

Figure IV.3 Hubber System

In this paper, hubber is also described with the term PoI (Point of Interconnection) between Telco Indonesia with OTT or global CDN. Hubber as a PoI mentioned in this paper has administrative, business, and technical aspect. Physical network connection which supports hubber system is just a part of the technical aspect of the PoI.

- *Roles and Functions of Hubber*

Roles and functions of the hubber in proposed regulation is as follows:

1. Managing the relation between OTT provider and global CDN with national Telco operators
2. Running the local CDN functionality
3. Managing network security management
4. Maintaining QoS and GoS to the customer together with Telco operators

- *Hubber Business Model*

In terms of business, the presence of hubber changes best effort service distributed via the Internet into a new form of service with the QoS. Some OTT providers that choose to connect to their users in Indonesia via hubber can guarantee the quality of their services to their customers in accordance to the selected class of service.

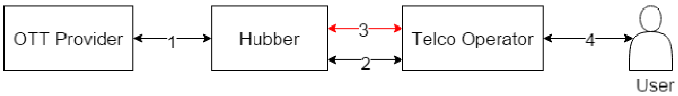

Figure IV.4 Traffic Flows in Hubber System

1. OTT providers can put the contents, data, and applications that are frequently accessed by Indonesian customers on the hubber server. In addition, the traffic from OTT providers to Indonesian users and vice versa are passed through hubber network.
2. Hubber network distributes the traffic from OTT to network operators in Indonesia and vice versa.
3. For communications between users in Indonesia, OTT traffic especially static content is served by hubber server.
4. Customers are connected to the Internet, especially OTT services only through Telco operators.

Furthermore, in the figure IV.5 and IV6, the service exchange and expected fee are illustrated and explained.

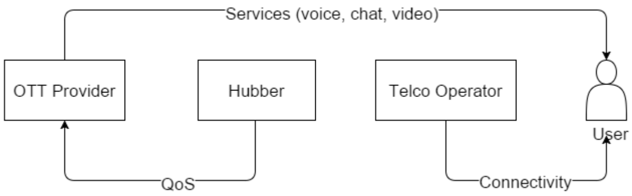

Figure IV.5 Service Interaction in Hubber System

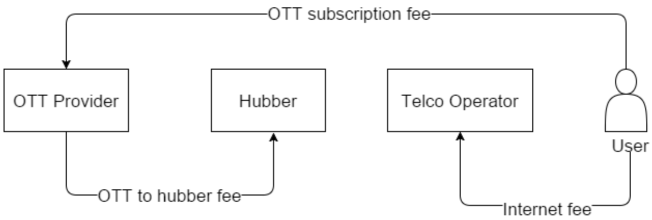

Figure IV.6 Billing in Hubber System

In hubber system, the concept of OTT service subscription fee is introduced. The subscription fee is paid by the customer and then divided to both OTT providers and Telco operators as an additional revenue. The proposed payment mechanism of the OTT subscription fee is using the prepaid or postpaid system that conforms to the Internet subscription mechanism for respective customers. This mechanism chosen in order to facilitate the implementation and supervision of the payment. The payment mechanism is illustrated in figure IV.7.

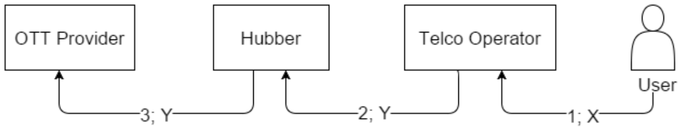

Figure IV.7 Payment mechanism in Hubber System

1. Users make payment with prepaid or postpaid mechanism for Internet services like they do in pre-hubber system. In addition, users also pay a subscription fee for OTT services that they want to use, symbolized as a number X. The amount of X depends on the service level (class of service) selected by the customer.
2. From the payment X, Telco operators received some portion of it as a consequence of the use of their resource in delivering the OTT services. The other portion, called Y, is given to hubber.
3. Y is received by OTT providers as a subscription fee for their services.

The existence of the class of service is also proposed in hubber system, allowing customers to choose the level of service from one OTT provider according to the customer's needs and capability. Examples of the class of service implementation in hubber system is as follows:

Table IV.1 The Example of Class of Service Proposed in Hubber System

CLASS	AVAILABLE SERVICES	CHARGE
3	Only reading	Free
2	Class 3 service + *texting*	Charged
1	Class 2 service +*sharing file*	Charged (higher price)

C. *Hubber Business Simulation*

In this business simulation, a forecast is conducted using data from Telkomsel as a representative Telco operators of Indonesia to determine the Telco operators' revenue target from hubber. The principle used in this forecast is that the loss of income from the Telco operators' legacy services – telephony and SMS – are replaced by income from a new source, in this case, hubber. Telkomsel's total revenue growth determined in advance in accordance with the national GDP growth target, and Telkomsel's revenue target from hubber is determined to cover the loss of revenue from the SMS and telephony, so that Telkomsel's revenue growth target can be achieved. Using national GDP as a reference, the value of 7% is used as the linear target value of the revenue growth.

Forecasting in this simulation is done with time-series extrapolation method of the existing annual growth values [7].

Business simulation calculations yield predictions and targets as shown in the figure IV.8.

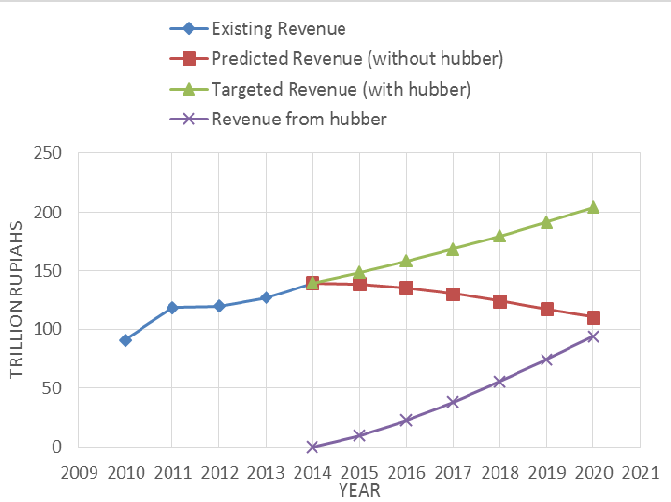

Figure IV.8 Result of Hubber System Business Simulation

From figure IV.8:

- Existing Revenue: revenue value in the annual report.
- Predicted revenue: revenue calculation as the result of predicting and summarizing the values of SMS, telephony, and Internet volume from year to year. This is calculated with assumption that hubber is not implemented.
- Targeted revenue: target values that is obtained by setting the annual growth rate of 7%.
- Revenue from hubber: additional revenue for Telkomsel with assumption that hubber system is applied, originated from the OTT subscription fees from the customers.

From the business simulation that has been done, the revenue target of hubber operator is calculated, and the result is indicated by the the cross-marked line in figure IV.8. It can be seen that the value of operator's revenue from hubber implementation is increasing every year. This is in line with the trend that the use OTT service (IP-based communications) as the main communication services will continue to increase while SMS and telephony will be gradually decreased until completely replaced by IP services.

Table IV.2 Determining Telkomsel's Revenue Target from Hubber

Year	Targeted Revenue (Trillion Rupiahs)
2015	4.83
2016	11.22
2017	18.81
2018	27.32
2019	36.48
2020	46.14

For the next calculation, proposed a 50:50 profit sharing between OTT and Telkomsel. This assumption is used because of the initial idea where OTT and Telco have the same significant role in operating communications services. Thus, total revenues from hubber implementation can be determined by summing Telkomsel's revenue target from hubber and OTT provider's target revenue from huber.

Table IV.3 Determining Hubber ARPU

Year	Total revenue from hubber implementation (Trillion Rupiahs)	Data enabled customer (thousands)	Hubber ARPU (Rupiahs per months)
2015	9.65	71791.30	11204.10
2016	22.43	74539.11	25080.57
2017	37.63	76518.46	40977.48
2018	54.63	77983.57	58379.77
2019	72.97	79094.35	76880.18
2020	92.27	79954.30	96173.13

Hubber ARPU as shown in the rightmost column of table IV.3 is the cost that a customer is expected to pay in addition to Internet cost each month. Hubber ARPU value is expected to increase every year. This ARPU hubber is the expected average of the total subscription cost of a customer for all OTTs that is used, because in the OTT era a customer can subscribe to more than one OTT providers. Therefore, the more OTT providers are connected via hubber, the more affordable is target subscription fee per OTT.

The increasing hubber ARPU target is in line with the increasing hubber deployment target, so that OTT services with guaranteed QoS can be enjoyed and the better quality of OTT communication services can be experienced by more customers in Indonesia.

V. CONCLUSION

Hubber system is needed in Indonesia to maintain the existence of Indonesian telecom operators facing the global trend where non IP-based communication services are gradually replaced by IP-based communication services provided by OTT providers.

Hubber system in Indonesia is proposed to comply these regulation points:

- Huber definition: a hub for IP-based data communications between OTT or global CDN provider with Telecommunications providers that provide Internet access to users in Indonesia, which has a special infrastructure consisting of Telecommunications infrastructure and distributed data center.
- Roles of hubber: managing the relationship between OTT or Global CDN with national Telco, performing the function of a local CDN, managing network security management, and maintaining QoS and GoS services to the customer.
- In the new business model, hubber system offers changes from today OTT services which is best effort via the Internet into a service that has a QoS. This is manifested by the OTT subscription fee which is then divided as revenue for both OTT providers and Telco operators. Meanwhile, traffic from OTT providers who do not cooperate with hubber still pass through the network as best effort traffic.

Based on the results of business simulation done on Telkomsel, it is predicted that Telkomsel's revenue will go down for the first time in 2015. To achieve positive 7% revenue growth, Telkomsel has to set its revenue target from hubber of 4.83 trillion Rupiahs in 2015 with a steady increase in annual growth and reaching almost a tenfold increase in value, 46.14 trillion in 2020.

References

[1] M. K. John Ure, "Discussing the Grey Areas in Regulating OTT Services," in *Telecom Regulatory Affairs Asia*, Singapore, 2013.

[2] E. Bertin, "A Few Myths About Telco and OTT Models," in *Intelligence in Next Generation Networks (ICIN)*, Berlin, 2011.

[3] R. van den Dam and E. Nelson, "Telco 2015: Re-envision the future of mobile communications," in *Wireless Telecommunications Symposium (WTS)*, New York, 2011.

[4] F.-Y. Ling, "Research on the Net neutrality: the Case of Comcast Blocking," in *International Conference on Advanced Computer Theory and Engineering (lCACTE)*, 2013.

[5] M. B. Setiawan, "Pemanfaatan Spektrum Frekuensi Radio dan Orbit Satelit Dalam Mendorong Pembangunan Broadband," Kementrian Komunikasi dan Informatika, Jakarta, 2013.

[6] K. Jayakar and E.-A. Park, "Emerging frameworks for regulation of Over-The-Top services on mobile networks: An international comparison," 2014.

[7] J. S. Armstrong, "Extrapolation for Time-Series and Cross-Sectional Data," in *Principles of Forecasting: A Handbook for Researchers and Practitioners*, Kluwer, 2001.

BER Performance Analysis of APD-based FSO System for Optical Inter-HAPS Link

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***Abstract*—This paper aims at evaluating bit error rate (BER) performance of optical inter platform link using intensity modulation/direct detection (IM/DD) with on-off keying (OOK) and an avalanche photodiode (APD) receiver based on analyzed theoretically. The atmosphere is not an ideal communication channel. Atmospheric turbulence is one of the main impairment affecting the operation of optical inter platform link. The log normal distribution is used as inter-HAP channel model to describe the effect of atmospheric turbulence. The BER performance is theoretically derived with taking into account various FSO link conditions and receiver parameters. The numerical results ilustrate that the optimal APD gain to achieve minimum BER depends on data rate, the link distance, and atmospheric turbulence strength.**

***Keywords*—component; avalanche photodiode (APD), optical interplatform link, atmospheric turbulence, log normal distribution.**

I. INTRODUCTION

High altitude platform station (HAPS) is proposed as a new means for wireless system delivery method [1]-[2]. HAPS has many advantages compared with existing terrestrial or satellite system. Therefore, HAPS is envisaged as a complimentary wireless system that the terrestrial or satellite cannot be cost efficiently deployed anymore. With the position of approximately 17-22 km above the ground, HAPS is very promising to deliver many kind of services. For example for earth observation and location based monitoring system, early warning system, broadcasting system, and of course for mobile telecommunication services. HAPS is also able to be connected with terrestrial system and satellite system for backhaul communication.

Performance of HAPS channel for mobile communication has been addressed in some literatures [3]-[7]. The channel characteristic of HAPS for downlink and uplink tend to follow Ricean fading channel rather than Gaussian or Rayleigh [1]. The propagation signal in HAPS communication is dominated by direct line of sight signal. However, for inter-platform link performance, there is no so much result has been published.

Free space optical communication is an optical communication technology that offers extremely high bandwidths, implying that it can suppot high data rates. FSO communication technology has many advantages compared to microwave technology. Because of that,FSO systems is considered as attractive solution for high-data rates last-mile link communications. Some possible applications of FSO are inter-satellite links (ISL) in satellite networks, links for deep space missions, links between unmanned aerial vehicles (UAV) and High Altitude Platforms (HAPs), and data links from geostationary satellites (GEO) to earth ground stations [8]. Optical Inter Platform Links (OIPLs) is a term for optical link that connects two HAPS (High Altitude Platform Stations). Optical inter platform links can be used to build up stratospheric back haul network independent from terestrial ground network [9].

In FSO communication, the atmospheric turbulence is a major impairment over FSO links. Atmospheric turbulence is a phenomenon caused by inhomogenities in the index of refraction known as scintillation, which could deteriorate the quality of the the received signal and lead to an increase in the bit error rate of the FSO systems [10]. Beside the atmospheric turbulence, performance of optical interplatform link are degraded by the geometric loss and atmospheric attenuation caused by the absorption and scattering processes [11]-[12].

The intensity modulation with direct detection and on-off keying (OOK) scheme is mostly deployed in commercial FSO systems because of the simplicityin design and implementation [12]. In long distance optical communication, the AvalanchePhoto Diode (APD) is generally used as photodetector owing to its internal gain. Performance of APD is better than PIN diode when the received power levels are low [13].The higher gain of APD doesn't imply the better performance. Since the performance will degrade after beyond a certain gain. So, the optimum gain of APD has to be determined to achieve minimum BER.

In this paper, optical inter platform link with IM/DD and OOK scheme using APD receiver has been observed.Bit error rate has been chosen as performance parameter of optical inter platform link in the presence of atmospheric turbulence. For inter-HAPS link, because of lower density of particles and gas, the random attenuation due to atmospheric turbulence is weak [13]. In case of weak turbulence, log-normal distribution can be used to model random process of atmospheric turbulence. The impact of the receiver noise, such as APD shot noise and thermal noise are taken into account.

The remainder of the paper is organized as follows. Section II describes the system description. In section III, channel model, including atmospheric turbulence, attenuation and geometric loss on BER analysis is presented. In section IV, BER expression are derived. The numerical results is discussed in section V. The conclusions are presented in section section VI.

II. System Description

Current technology that used for optical inter sattelite link (OISL) should be suitable for atmospheric OIPLs. Fig. 1. ilustrates optical link inter platform using IM/DD with OOK scheme and APD receiver. At the transmitter terminal, block modulator IM/OOK modulates information bits in electrical signal and generates output intensity of the laser source that represents information bits. Then, block transmitter telescope directs and determines the size of the laser beam. Laser beam travels through the atmospheric channel to the receiver terminal. At the receiver terminal, telescope narrows the laser beam and forwards it to the APD photodetector. APD converts optical signal into electrical signal [14].

The APD photodetector receives optical signal and can be expressed as,

$$r = axI = \begin{bmatrix} r_{on} \\ r_{off} \end{bmatrix} = \begin{bmatrix} a2P_t I \\ 0 \end{bmatrix} \qquad (1)$$

where$x$ represents the information 'on' ($2P_t$) or 'off' (0) bit, $a$is total link loss and I is the received field intensity in presence of atmospheric turbulence, and P_t is the average transmitted optical power .

The output of APD photodetector generates the electrical signal in 'on' and 'off' states which is expressed by,

$$r_e = \begin{bmatrix} gRr_{on} + n_{on} \\ r_{off} + n_{off} \end{bmatrix} = \begin{bmatrix} gRa\,2P_t I + n_{on} \\ n_{off} \end{bmatrix} \qquad (2)$$

where g represents average gain APD and R is responsivity of APD. The total APD receiver noise is stated by n and differen is 'on' and 'off' states ($n_{off}$and n_{on}).

Noise receiver can be modeled as the stationary Gaussian random process with the zero-mean value and variance [15]. The total receiver noise is caused by shot noise, thermal noise and dark current.With dark current is negligible, so it can beexpressed as,

$$\sigma_n^2 = \sigma_{th}^2 + \sigma_{sh}^2 \qquad (3)$$

where σ_{th}^2 represents the variance of thermal noise and σ_{sh}^2 is shot noise variance.

Variance of thermal noise σ_{th}^2 can be expressed as,

$$\sigma_{th}^2 = 4k_B \frac{T}{R_L} F_n \Delta f \qquad (4)$$

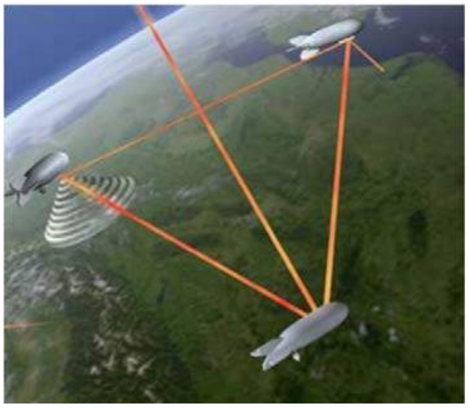

Fig. 1 Inter-platform link.

where $k_B$is Boltzmann constanct, T is the receiver's temperature in Kelvin, $R_L$denotes APD's load resistance, F_n is he amplifier noise figure and Δf is symbol's effective noise bandwidth, here stated that $\Delta f = R_b/2$, $R_b$denotes the bit rate.

Different from thermal noise, shot noise depends on the APD parts. The atmospheric turbulence causes the fluctuation in the received field intensity leads to uncertain in the shot noise variance. The shot noise variance can be expressed as,

$$\sigma_{sh}^2 = \begin{bmatrix} 2qg^2 F_A RI2P_t \Delta f \\ 0 \end{bmatrix} \qquad (5)$$

where q is an electron charge and F_A is the excess noise factor and can be stated,

$$F_A = k_A g + (1 - k_A)(2 - 1/g) \qquad (6)$$

where k_A is the ionization factor.

The variance of the total noise APD receiver can be expressed as,

$$\sigma_n^2 = \begin{cases} 4k_B \frac{T}{R_L} F_n \Delta f + 2qg^2 F_A RI2P_t \Delta f & \text{on} \\ 4k_B \frac{T}{R_L} F_n \Delta f & \text{off} \end{cases} \qquad (7)$$

III. Inter HAPS Link Channel Model

A. Structure Parameter Index Refractive (C_n^2)

In atmospeheric turbulence, the structure parameter, $C_n^2(h)$, represents the total amount of energy contained in the stochastic field of the refractive index fluctuations [16]-[20]. It is a measure of optical turbulence strength and required for the calculation of important fading related parameters like the scintillation. In this paper, the altitude profile for the parameter C_n^2 is generated using the Hufnagel-Valley (H-V) model. The H-V model is expressed as a function of height as follows,

$$C_n^2(h) = 5.94\times 10^{-3}\times (v_{rms}/27)^2 \times (h/10^5)\times \exp(-h/1000) + 2.7\times 10^{-16}\times \exp(-h/1500) + C_n^2(0)\exp(-h/100) \quad (8)$$

where h is the altitude in meters, $v_{rms}$is the rms wind speed in m/s and $C_n^2(0)$ is the structure constant at level ground.

B. Scintillation Index

The intensity Scintillation Index (SI) is defined as the normalized variance of irradiance fluctuations. For such a high divergence and a long propagation paths, the Gaussian beam wave is close to a speherical wave in terms of scintillation [2]. Scintilattion index is expressed as,

$$\sigma_I^2 = \exp\left[\frac{0.49\sigma_{I,sp(0)}^2}{(1+0.18d^2+0.56\sigma_{I,sp(0)}^{12/5})^{7/6}} + \frac{0.51\sigma_{I,sp(0)}^2(1+0.69\sigma_{I,sp(0)}^{12/5})^{-5/6}}{1+0.9d^2+0.62d^2\sigma_{I,sp(0)}^{12/5}}\right]-1 \quad (9)$$

where $d=\sqrt{kD^2/4L}$ and $\sigma_{I,sp(0)}^2 = 0.4\sigma_R^2 = 0.492C_n^2 k^{7/6}L^{11/6}$ ($k=\frac{2\pi}{\lambda}$ is the oprical wave number; L is the length of optical link; D is the receiver's aperture diameter).

C. Link Loss

The link loss consist of atmospheric attenuation and geometric loss. The atmospheric attenuation is caused by absorption and scattering both molecular and aerosol in the air. While the geometric loss depends on the angle of divergence of the transmitter telescope and the area of the receiver's aperture. Total link loss can be expressed as,

$$a = \frac{D}{(\theta L)^2}\exp(-\beta_v L) \quad (10)$$

where D is diameter of the receiver's aperture, θdenotes divergence angle, L is link distance and β_v is extinction coefficient.

D. Turbulence Model

An inter-HAP channel, under the influence of IRT, can be modellede as a scintillation channel with fades and surges. In case of weak turbulence, generally the influence of turbulence is modelled as a random process with log normal distribution [9]. The probability density function (PDF) of the log normal model is given by,

$$p(I) = \frac{1}{I\sqrt{2\pi\sigma_I^2}}\exp\left\{-\frac{\left(\ln(I)+\frac{\sigma_I^2}{2}\right)}{2\sigma_I^2}\right\} \quad (11)$$

where is the scintillation index defined in (9).

IV. Inter HAPS Link Performance Analysis

The conditional BER of FSO system using IM/DD with OOK depends on the Q-parameter and can be expressed by,

$$BER = \frac{1}{2}erfc\left(\frac{Q(I)}{\sqrt{2}}\right) \quad (12)$$

where Q-parameter is given by,

$$Q(I) = \frac{gR2aP_tI-0}{\sigma_{n/on}+\sigma_{n/off}} = \frac{gRa2P_tI}{\sqrt{4k_B\frac{T}{R_L}F_n\Delta f}+\sqrt{2qg^2F_ARI2aP_t\Delta f+4k_B\frac{T}{R_L}F_n\Delta f}} \quad (13)$$

TThe average BER over lognormal fading channel is obtained by,

$$BER = \frac{1}{2}\int_0^\infty erfc\left(\frac{gRa2P_tI}{\sqrt{2}\left(\sqrt{4k_B\frac{T}{R_L}F_n\Delta f}+\sqrt{2qg^2F_ARaI2P_t\Delta f+4k_B\frac{T}{R_L}F_n\Delta f}\right)}\right) \times\frac{1}{\sqrt{2\pi\sigma_I^2}}\frac{1}{I}\exp\left\{-\frac{\left(\ln(I)+\frac{\sigma_I^2}{2}\right)^2}{2\sigma_I^2}\right\}dI \quad (14)$$

The equation (14) can be simplified by using the Gauss-Hermite approximation, so it can be stated,

$$\langle BER\rangle = \frac{1}{2\sqrt{\pi}}\sum_{i=1}^{N} w_i erfc\left[\frac{Q(x_i)}{\sqrt{2}}\right] \quad (15)$$

where w_i and x_i are the weights and zeros of the Hermite polynomial, respectively.

V. Numerical Results and Discussion

In this section BER analysis of the free space optical system link using APD receiver over atmospheric turbulence channel is presented. Log-normal distribution is used to model random process of atmospheric turbulence for inter-HAP link channel. In this paper, the optical link connects two HAPS that both located at same heightat 20 km from the earth's surface. Based on fig.3 in [13], for wavelength 1550 nm, we choosethe attenuation coefficient is $4.8\text{x}10^{-4}$ km^{-1}. The other values of system parameters and constants are given in Table 1.

TABLE I CONSTANTS AND SYSTEM PARAMETERS.

Parameter (symbol)	Value	Unit
Average transmitted power (P_t)	30	dBm
Optical Wavelength (λ)	1.55µ	m
Boltzmann Constant (kB)	1.38×10^–23	W/kHz
Electron charge (q)	1.6x10^019	C
APD load resistance (RL)	50	Ohm
Amplifier noise figure (Fn)	2	-
Ionization factor (k_A)	0.025	-
Responsivity (r)	0.8	A/W
Receiver Temperature (T)	300	k
Receiver's aperture diameter	0.2	m
Divergence angle	140µ	rad

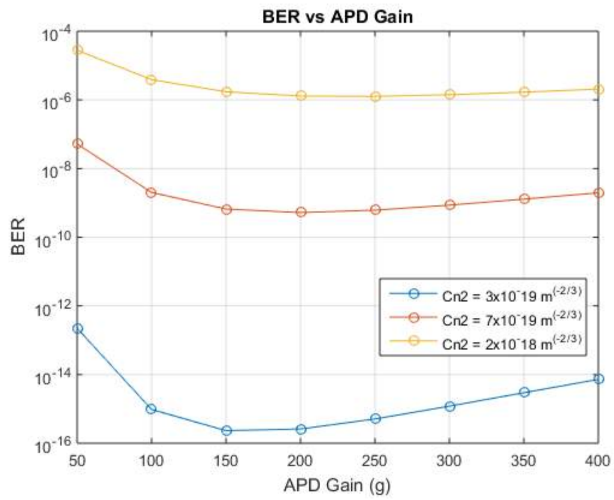

Fig. 1. BER vs APD gain with various turbulence strength, with average link distance L = 600km, data rate Rb = 5Gbps

BER dependence on average APD gain for different values of parameter C_n^2 is presented in fig.2 . There are three conditions of turbulence strength : $C_n^2 = 3x10^{-19}$ $m^{-2/3}$ for weak turbulence, $C_n^2 = 7x10^{-19}$ $m^{-2/3}$ for moderate, and $C_n^2 = 2x10^{-18}$ $m^{-2/3}$ for strong turbulence condition. As expected, the lower parameter C_n^2, the optical link has better performance. In the case of weak turbulence, the optimum gain is about 150, and for moderate and strong turbulence conditions, the optimum gain are 200 and 250, respectively.

In Fig. 2, the system BER versus the average gain APD with different values of data rate is ilustrated. For different value of data rate, the optimum gain varies between 150 and 250. When the data rate is larger, the optimum gain becomes smaller.

Fig. 3 ilustrates the average BER against the average APD gain for different link distances. The greater link distance, the worst average BER performance. With proper selection of APD gain, the BER performance can be optimized. So, we can conclude that the appropriate selection of average APD gain is one of important link parameter parameters in design of free space optical communication system for inter HAP link

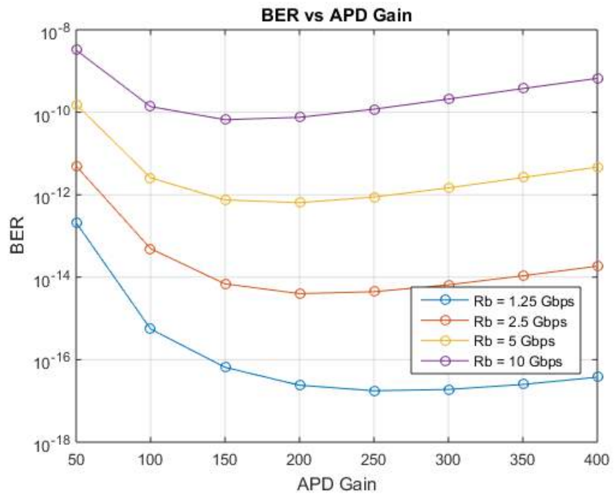

Fig. 2. BER vs APD gainwith various link distance L = 600km with turbulence strength C_n^2= 7.58x10^{-19} $m^{-2/3}$

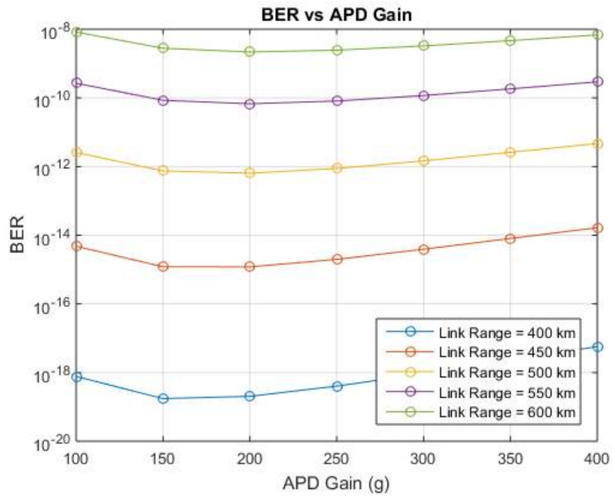

Fig. 3. BER vs APD gain with different link distances, R_b = 5Gbps, C_n^2= 7.58x10^{-19} $m^{-2/3}$

VI. CONCLUSION

We have theoretically analyzed the BER performance of free space optical interplatform link using APD receiver over turbulence atmospheric turbulence channels which is modelled by the log normal distribution. The average BER was theoretically derived with taking into account noise receiver, attenuation and geometric losss, atmospheric turbulence strength, link distance and data rates. The numerical result show that with different conditions of turbulence strength, link distance, and data rates, the optimal average APD gain varies between 150 and 250.

REFERENCES

[1] Iskandar and S. Shimamoto, "Channel characterization and performance evaluation of mobile communication employing stratospheric platform," IEICE Transactions on Communications, vol. E89-B, no. 3, pp. 937-944, March 2006.

[2] Iskandar, "Reverse link CDMA system capacity evaluation for stratospheric platform mobile communications," Jurnal Makara Seri Teknologi, Universitas Indonesia, vol. 11, No. 2, pp. 49-58, November 2007.

[3] Iskandar and S. Shimamoto, "On the downlink performance of stratospheric platform mobile communications channel," Proceedings IEEE Global Telecommunication Conference (GLOBECOM), ISSN: 1930-529X, DOI: 10.1109/GLOCOM.2006.922, pp. 1-5, 2006.

[4] Iskandar and S. Shimamoto, "Analysis of CDMA capacity for multiple stratospheric platform mobile communications under imperfect power control and fading," in Proceedings IEEE Wireless Communications and Networking Conference (WCNC), ISSN: 1525-3511, DOI: 10.1109/WCNC.2007.468, pp. 2512 - 2516 , 2007.

[5] Iskandar, A. Kurniawan, E. B. Sitanggang, and S. Shimamoto, "Step Size Optimization for Fixed Step Closed Loop Power Control on WCDMA High Altitude Platforms (HAPs) Channel," Proceedings IEEE Global Telecommunication Conference (GLOBECOM), vol. 1, pp. 1-5, Dec. 2008.

[6] Iskandar, A. Kurniawan, and H. Syaputra, "On the Evaluation of Fixed Step Closed Loop Power Control for CDMA High Altitude Platforms (HAPs) Communication Channel," Proceedings IEEE International Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC), vol. 1, pp. 1-5, Sept. 2008.

[7] Iskandar, A. Kurniawan, and M.E. Ernawan, "Performance of Variable Step Closed Loop Power Control in CDMA High Altitude Platforms Communication Channel," Proceedings of The 2010 IEEE 72nd Vehicular Technology Conference (VTC Fall 2010), DOI 10.1109/VETECF.2010.5594401, pp. 1-5, Sept. 2010.

[8] A. K. Majumdar, J. C. Ricklin, "Free Space Laser Communications Principles and Advances", Springer ISBN 978-0-387-28652-5, 2008

[9] J. Horwath, N. Perlot, D. Giggenbach, and R. Jüngling, "Numerical simulations of beam propagation through optical turbulence for high-altitude platform crosslinks", Proceedings of the SPIE 2004, vol. 5338B, 2004.

[10] Bach T. Vu, Ngoc T. Dang, Truong C. Thang, and Anh T. Pham, "Bit Error Rate Analysis of Rectangular QAM/FSO Systems Using an APD Receiver Over Atmospheric Turbulence Channels," J. Opt. Commun. Netw., vol. 5, no. 5, pp. 437–446, May 2013.

[11] Andrews, L. C., Philips, R. N. Laser Beam Propagation through Random Media. 2nd ed. Bellingham (USA): Spie Press, 2005.

[12] Ghassemlooy, Z., Popoola, W., Rajbhandari, S. Optical Wireless Communications: System and Channel Modelling with MATLAB. Boca Raton (USA): CRC Press, 2013.

[13] P. Gopal, V. K. Jain and S. Kar, "Performance Comparison of PIN and APD based FSO Satellite Systems for various Pulse Modulation Schemes in Atmospheric Turbulence," Commun. Netw., vol. 5, pp. 200–203, September 2013.

[14] Petković, M. I., Milić, D. N and Đorđević, G. T, "BER Performance of IM/DD FSO System with OOK using APD Receiver," Radioengineering., vol. 23, No.1 pp. 480–487, April 2014.

[15] VU, B. T., DANG, N. T., TRUONG, C.-T., PHAM, A. T. Bit error rate analysis of rectangular QAM/FSO systems using an APD receiver over atmospheric turbulence channels. IEEE/OSA Journal of Optical Communications and Networking, 2013, vol. 5, no. 5, p. 437 - 446.

[16] Parthasarathy, Swaminathan and Giggenbach, Dirk and Kirstädter, Andreas, "Simulative Performance Analysis of ARQ Schemes for Free-Space Optical Inter-HAP Channel Model ".Proceedings of Photonic Networks.Photonic Networks;16 ITG Symposium 2014, 7-8May 2015, Leipzig,German.ISBN:9783800739387.

[17] G. P. Agrawal, Fiber-optic Communication Systems, G. P. Agrawal, Ed. John Wiley and Sons, India, 2002.

[18] K. Kiasaleh, "Performance of apd-based, ppm free-space optical communication systems in atmospheric turbulence," IEEE Transactions on Communications,, vol. 53, no. 9, pp. 1455–1461, 2005.

[19] M. Abramowitz, Handbook of Mathematical Functions with Formulas, Graphs and Mathematical Tables, M. Abramowitz, Ed. Dover, New York,1965.

[20] M. Mohorcic, C. Fortuna, A. Vilhar, "Evaluation of WavelengthRequirements for Stratospheric Optical Transport Networks," *Journal ofCommunications* , vol. 4, no. 8, pp.588-596, September 2009.

Over The Top Chat Service Key Performance Indicator

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Abstract— **Evolution in telecommunication network and increasing use of smartphones in Indonesia has triggered the development of a wide range of service. One of them is Over the Top Service (OTT). But, this development is not accompanied by government regulation regarding the Key Performance Indicator (KPI) for OTT service. In this paper, we provide recommendations for OTT chat service Key Performance Indicator in Indonesia. The method used in this paper is analyzed collected data from measuring QoS OTT chat service through 4G Network and benchmarking.**

Keywords—Over The Top, Chat Service, Key Performance Indicator, 4G, LTE

I. INTRODUCTION

Services in the telecommunications sector increasingly diverse, including telephone services, SMS, internet service, and Over the Top. By increasing the number of services, then the burden will increase network traffic that can lead to decreased quality of service. These services require various indicators to measure and evaluate the performance by using the Key Performance Indicator.

Services over the Top (OTT) has been much used by the inhabitants of Indonesia along with the increasing use of smartphones in the community. These types of services include We Chat, Facebook, Line, and WhatsApp. Various studies of new technologies to support OTT has been prepared even implemented in Indonesia, but there is no regulation regarding the Key Performance Indicator of OTT services are provided by the government when the KPI is very important as an indicator of quality and evaluation tools of a service.

To overcome these problems, this paper provides recommendations to the Indonesian government regarding the Key Performance Indicator of OTT chat service by doing an analysis of the measurements of the Quality of Service of such services and benchmarking. Network used to perform the measurements is the 4G network in Bandung, which uses a sampling method, the direct measurement method, and passive methods.

II. OVER THE TOP CHAT SERVICE

A. *Over The Top Service*

OTT service is service provided by a third party. Server in OTT service not provided by network operator. Even though server not provided by network operator, but the communication in OTT service is still using network operator infrastructure. OTT service distributing voice, video, or data in their services [1]. OTT service has made a big change in telecommunication industry. There are a lot of developing OTT service such as Line Messenger, Kakao Talk, Bee Talk, Facebook, WhatsApp, etc.

B. *OTT WhatsApp Chat*

Here is an OTT service topology WhatsApp messenger:

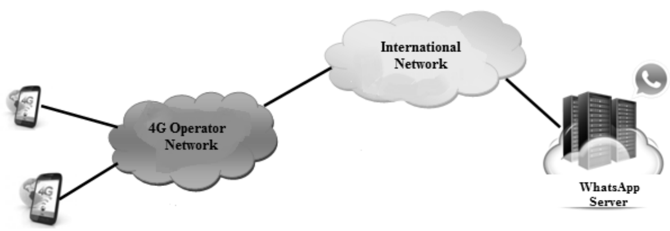

Fig. 1. WhatsApp Architechture

1. 4G operator provide network infrastructure for OTT service. There's two kind of 4G technology which is Long Term Evolution (LTE) and WiMax. In Indonesia, network operator just provide LTE. They are three provider have deployed 4G LTE technology, there are XL, Telkomsel, and Indosat.

2. International network should use IP Multimedia Subsystem (IMS) framework because IMS technology can bridge GSM/GPRS/EDGE, UMTS, HSPA/HSDPA, and traditional PSTN able to provide 4G services [2].

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3. WhatsApp Server runs on the FreeBSD operating system [3] and now has simultaneous transactions of 1 million TCP request.

C. *QoS and QoE Theory*

Quality and performance in the telecommunications field can be divided into several levels based on ITU-T Recommendation E.800 [4] like explanation below:

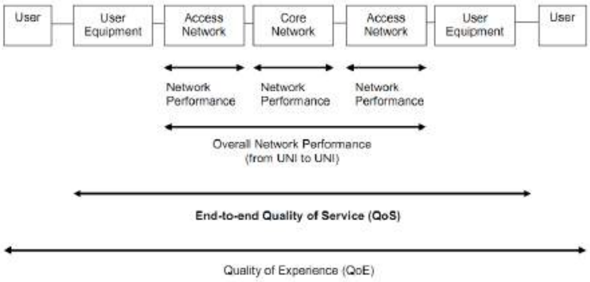

Fig. 2. QoE and QoS Theory [4]

- Network Performance
  Network performace is the performance of a network element like access network and core network.
- Overall Network Performance
  The performance of all elements is integrated to form a network so will measure performance.
- End-to-End QoS
  End-to-End QoS is overall performance of the elements and process from user equipment, access and core network, and to another user equipment. QoS measurements depend on some typical characteristics of a telecommunications service.
- QoE (Quality of Experience)
  QoE has similarity with QoS but it is overall of a telecommunications service by involving users in the process of measurement. Assessment for QoE more subjective than QoS because it is influenced by the experience and the level of expectations of users.

D. *4G LTE*

LTE network is based on architecture called Envolved Packet Switched System [5]. This is architecture of 4G LTE shown in figure 3.

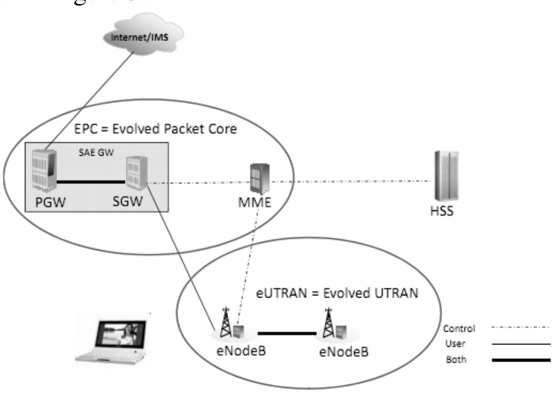

Fig. 3. 4G Architechture [6]

EPS is composed by two main subsystems [7], there is eUTRAN (involved Utran) as the radio access network and EPC (Envolved Packet Core) as the core network.

Main network elements of E-UTRAN are user equipment (UE) which is terminal of end user and envolved node B (eNB).

Main network elements of EPC are MME (Mobility Management Entity), HSS (Home Subscriber Systems), SGW (Serving PDN Gateway), PGW (Proxy PDN Gateway), and PCRF (Policy and Charging Rules Function).

MME is control plane element to manage end user and for connection management from eNodeB. HSS is database for user and subscriber and verifies end user subscription. SGW is have function as interconnection between radio sides (eUTRAN) with EPC. PGW function is as interconnection between EPC with external IP network.

PRCF is crucial for end to end QoS approach because this is network node able to define QoS policies of a particular service requested by user.

E. *Performance Parameter*

In this research, we will use SMS service parameter in ETSI TS 102 250-2 V2.3.1 as the approach for chat service performance parameter because it's still not regulated yet. This is parameter chat service which adopted from ETSI TS 102 250-2 V2.3.1 [8]:

1) *Endpoint Service Availability*

Endpoint service availability is measurement present age chat sent with time interval between sender and receiver not more than 3 minutes.

$$\text{Endpoint Service Availability } [\%] = \left(\frac{\text{chat sent in 3 minutes}}{\text{all chat service attempts in 1 period}}\right) x\ 100 \quad (1)$$

2) *Service Non-Accessibility*

Probability that the end-user cannot access the chat when requested while it is offered by display to the network indicator on UE.

$$\text{chat Service Non Accessibility } [\%] = \left(\frac{\text{unsuccessful chat service attempts}}{\text{all chat service attempts}}\right) x\ 100 \quad (2)$$

3) *Access Delay*

The time period between sending a chat message to the network and receiving a send confirmation from the network at the originating side.

$$\text{chat Access Delay } [s] = (t_{A\ receive} - t_{A\ send})\ [s] \quad (3)$$

4) *Completion Failure Ratio*

Ratio of unsuccessfully received and sent chat messages from one UE to another UE, excluding duplicate received and corrupted messages

$$\text{Completion Failure Ratio } [\%] = \frac{\text{unsuccessfully received chat}}{\text{all chat service attempts}}\ x\ 100 \quad (4)$$

5) *End-to-End Delivery Time*

The time period between sending a chat message to the network and receiving the very same chat message at another UE.

$$chat\ End-to-End\ delivery\ time\ [s] = (t_{B\ receive} - t_{A\ send})\ [s] \qquad (5)$$

6) Receive Confirmation Failure Ratio

The probability that the receive confirmation for a sent chat attempt is not received by the originating UE although requested.

$$Receive\ Confirmation\ Failure\ Ratio\ [\%] = \frac{non-confirmed\ chat\ reception}{all\ chat\ service\ attempts}\ x\ 100 \qquad (6)$$

7) Receive Confirmation Time

The time period between sending a chat message to the network and receiving the receive confirmation for this chat from the network.

$$chat\ Confirmation\ Time\ [s] = (t_{A\ receive\ confirmation} - t_{A\ send})\ [s] \qquad (7)$$

8) Consumed Confirmation Ratio

The probability that the consumed confirmation for a sent chat attempt is not received by the originating UE although requested.

$$chat\ Consumed\ Confirmation\ Failure\ Ratio\ [\%] = \frac{non-confirmed\ SMS\ consumptions}{all\ SMS\ service\ attempts}\ x\ 100 \qquad (8)$$

9) Consumed Confirmation Time

The time period between sending a chat message to the network and receiving the consumed confirmation from the network.

$$chat\ Consumed\ Confirmation\ Time\ [s] = (t_{A consumed confirmation} - t_{A\ send})\ [s] \qquad (9)$$

III. Research Scheme

A. *Research Stage*

This is research scheme for this paper :

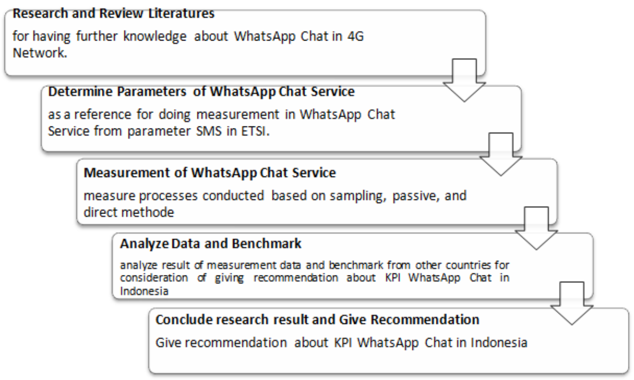

Fig. 4. Research Scheme

To give recommendation about KPI of chat service in Indonesia, first thing to do is doing research and review literature for having further knowledge about OTT, WhatsApp, and 4G technology. After that, determine parameter of WhatsApp service for doing measurement. Measurement was done with WhatsApp chat service on 4G network in Bandung using XL network operator.

B. *Measurement Methods*

In this research, author use all of this measurement method:

- Based on data source, this research use sampling method, which measuring QoS by taking a sampling of data from a service at a given time or over a range of specific work processes [9]
- Based on direct measurement whether or not do, this research use direct method which measure QoS with traffic directly observing conditions at some points in the network architecture.
- Based on involvement of the observer, this research use passive method, which measure QoS with the collection of information without any interference on the network.

C. *Performance Parameter Indicator*

After doing literature review and doing benchmarking, we select parameter from short message service parameters into the parameters for OTT chat service for doing measurement. We choose chat end-to-end delivery time and chat completion failure for being performance parameter indicator.

1. Chat End to End delivery time

The duration is calculated from the A-Party start sending chat : up to the B-party has received chat.

The end of a successful process of chat end to end delivery time is when the very same chat is received by B-Party.

The equation of chat end-to-end delivery time:

$$chat\ End-to-End\ delivery\ time\ [s] = (t_{B\ receive} - t_{A\ send})\ [s] \qquad (5)$$

2. SMS Completion failure ratio

Ratio of chat not received in MT and sent chat from MO : excluding duplicate received and corrupted chat. A corrupted chat is a chat which has at least one bit error.

Beginning of the chat is when sender push send button by MO to initiate sending a chat, until the receiver equipment receive the very same chat.

Equation of completion failure ratio is :

$$Completion\ Failure\ Ratio\ [\%] = \frac{unsuccessfully\ received\ chat}{all\ chat\ service\ attempts}\ x\ 100 \qquad (4)$$

IV. Benchmark and Literature Review

A. *Procedure of Key Performance Indicator (KPI) Chat Service Determination*

Key performance indicator is divided in two groups. There are performance indicator and target value which are related to the quality of the network and each service.

This is several consideration for arrange KPI [10].

- The international standards and recommendation of the ITU-T, ETSI, IEEE, ANSI, and others is the 1st consideration
- After that, the 2nd consideration is comparing standards of various telecom operators and National Regulatory Bodies in the world.
- Next consideration is find regulation National Regulatory Body on each country, and
- The last consideration is Service Level Agreement between providers and costumers.

B. Parameter of Key Performance Indicator refer to benchmark and literature Review

A lot of research have precede about Key Performance Indicator (KPI) and performance for services, for example : determining the KPI for the SMS service [11], the measurement of QoS for service File Transfer Protocol using aggregation method on 3G networks [12], measuring QoS of service Web Browsing using aggregation method [13], QoS measurement of telephony services [14], the measurement of service QoS Video Streaming [15], as well as measuring Quality of Experience (QoE) of OTT services [16].

Because of KPI of QoS OTT chat service still not exist in many telecommunication or governments in several country, we reused KPI from SMS like SMS end-to-end delivery time and SMS completion ratio from several international telecommunications organizations. General network quality not included in consideration, so we only focus to the chat service QoS parameters.

This is summary of the parameter in SMS service shown in table 1 from data collection in several country which used for being standard consideration in chat service.

Table 1. Benchmark OTT chat service

	Parameter	Value
	Chat End-to-End Delivery Time (s)	
a.	*Nigerian Telecommunication [17]*	5 s
b.	*World Class Operator Optimus (Portugal) [18]*	5.6 s
c.	*World Class Operator TMN (Portugal)*	5.7 s
d.	*World Class Operator Vodafone (UK)*	7 s
e.	*Pakistan Telecommunication [19]*	8 s
f.	*World Class Operator SFR & Boygues (France) [20]*	30 s
g.	*France Telecommunication Regulatory [20]*	120 s
h.	*Channel Island Competition and Regulatory [20]*	120 s
i.	*Peraturan Menteri Komunikasi dan Informatika nomor 12/PER/M.KOMINFO/04/2008 [21]*	180 s

C. Target value refer to benchmark and literature review

OTT services technology is still said to be quite new, therefore the data required for target setting value for OTT chat service does not exist. There is no presence of a standard estimate of the minimum value for each of the parameters network performance will be referred to as a target value because the hypothesis has not been too much research that discusses this technology.

V. Measurement and Analysis

The goal for this paper is conclude parameter and target value for telecommunication regulation body in Indonesia. There is many parameter in chat, but we choose two which has the most important role in sent chat message. That parameter is chat end-to-end delivery time and completion failure ratio.

End-to-end delivery time is important to used as parameter in chat service because this parameter indicating how good transmission process in chat service and directly related to customer satisfaction.

Completion failure ratio need to use as parameter because ideal chat message must have a good percentage of successful delivery.

Chat service divided into 3 part as described below:

1. Short chat which have maximal 160 byte long. Information in this type of chat is usually in the form of text or emoticons.
2. Medium chat which have maximal 100 kb long. Information in this type of chat is usually in the form of picture.
3. Long chat which have minimum 10 Mb long. Information in this type of chat is usually in the form of videos.

Based on Peraturan Menteri Komunikasi dan Informatika nomor 12/PER/M.KOMINFO/04/2008 [21], the sample have to 100. After measurement of over the top (OTT) chat service, the key performance indicators (KPI) are:

1. Chat end to end delivery time
   a) Short Chat (text)
   Chat attempts is 10 with average chat end to end delivery time :
   $$= \frac{all\ end\ to\ end\ delivery\ time}{all\ chat\ service\ attempts}$$
   = 0.6954 s
   b) Medium Chat (picture)
   $$= \frac{all\ end\ to\ end\ delivery\ time}{all\ chat\ service\ attempts}$$
   = 12.555 s
   c) Long Chat (video)
   $$= \frac{all\ end\ to\ end\ delivery\ time}{all\ chat\ service\ attempts}$$
   = 126.2361 s
   From the calculation of
2. Chat completion failure ratio
   a) Short Chat
   $$Completion\ Failure\ Ratio\ [\%] = \frac{0}{100}\ x\ 100 = 0\%$$
   b) Medium Chat
   $$Completion\ Failure\ Ratio\ [\%] = \frac{16}{100}\ x\ 100 = 16\%$$
   c) Long Chat
   $$Completion\ Failure\ Ratio\ [\%] = \frac{1}{100}\ x\ 100 = 1\%$$

Refered to the measurement of Over The Top Chat Service, Key Performance Indicator are end-to-end delivery time and completion failure ratio. Based on Peraturan Menteri Komunikasi dan Informatika nomor 12/PER/M.KOMINFO/04/2008 [21], end-to-end delivery time for short chat service which has similarity with short message service is enough because the value can less than 3 minutes. Even if compare with government policy in Nigeria, this value is make them satisfied too for this very quick end-to-end delivery time. But in medium and long chat, this take longer time for send chat, because the size which bigger than short chat.

VI. Conclusion

Depend on benchmarking and measurement result of What Sapp chat service performance in 4G network which took place in Bandung, Indonesia, we choose chat end-to-end delivery time and chat completion failure ratio for being the parameter. 4G network in Indonesia has very good performance in sent chat message. Measurement were performed by two 4G equipment and by using XL operator.

This paper is still in progress and still requires a lot of advice in the process.

References

[1] Huang Feng-hui, Du Yu, "QoE Issue of OTT Service Over 5G Network", Broadband and Wireless Computing, Communication and Applications (BWCCA), 2014 Ninth International Conference, 2014.

[2] Lava Al-Doski, Ayman Abbosh, Radu Babiceanu, Seshadri Mohan, "Optimizing Resource Utilization in IP Multimedia Subsystem", Systems Conference (SysCon), 2012 IEEE International, 2012.

[3] Himali Bajaj, Rajni Jindal, "Thinking Beyond WhatsApp", Computing for Sustainable Global Development (INDIACom), 2015 2nd International Conference, 2015.

[4] ITU-T Recommendation E.800 : Definitions of terms related to quality of service.

[5] Ian F Akyldiz, David M. Gutierrez-estevez, Elias Chavarria Reyes, "*The Evolution to 4G Cellular Systems: LTE-Advanced"*, Physical Communication 3 (2010) 217–244, 2010.

[6] Danijel Majacic, Kresimir Pecnic, "The Role of MSC Server in LTE based Network Architecture", MIPRO, 2010 Proceedings of the 33rd International Convention, 2010.

[7] Sonia Farconi and Alessandro Vizzarri, "Review of Studies on End-to-end QoS in LTE Networks" 7th International Conference Ubiquitous and Future Networks (ICUFN), in Elsevier United States, 2015.

[8] in ETSI TS 102 250-2 V2.3.1 (2013-04: Universal Mobile Telecommunication System (UMTS): LTE; Requirements for Evolved UTRA (E-UTRA) ad Evolved UTRAN (E-UTRAN).

[9] Strohmeier, Felix; Dorken, Heinz; Hechenleitner, Bernhard. (2001): AQUILA Distributed QoS Measurement. 2001.

[10] Sigit Haryadi , "Telecommunication Sevice and Experience Quality", ISBN: 978-602-18578-6-1, Jauary 2013.

[11] Liu Ya, Qi Feng, Zhang Jie, Xia Xiaofeng, "KPI-Oriented Analysis Model for SMS Operation Quality", Information Engineering and Computer Science, 2009. ICIECS, 2009.

[12] Sigit Haryadi, Raisha Andina, "QoS Measurement of File Transfer Protocol Services in 3G Networks Using Aggregation Method", Telecommunication Systems, Services, and Applications (TSSA), 2012 7th International Conference, 2012.

[13] Sigit Haryadi, Sandy Akbar Nusantara, "QoS Measurement Web Browsing Services in 3G Networks Using Aggregation Method", Telecommunication Systems, Services, and Applications (TSSA), 2012 7th International Conference, 2012.

[14] Sigit Haryadi, Ivantius Limampaw, "QoS Measurement Telepony Services in 3G Networks Using Aggregation Method", Telecommunication Systems, Services, and Applications (TSSA), 2012 7th International Conference, 2012.

[15] Sigit Haryadi, Video Streaming, "QoS Measurement Video Streaming Services in 3G Networks Using Aggregation Method", Telecommunication Systems, Services, and Applications (TSSA), 2012 7th International Conference, 2012.

[16] Huang Feng-hui, Du Yu, "QoE Issue of OTT Service Over 5G Network", Broadband and Wireless Computing, Communication and Applications (BWCCA), 2014 Ninth International Conference, 2014.

[17] Nigerian Communications Act (No.19 of 2003) Quality of Service Regulation, 2012.

[18] Autoridade National Telecommunications Regulatory Authority of Egypt, "NTTRA Egypt", [online] [accessed Oct, 18th 2015].

[19] Pakistan Telecommunication Authority Notification, Islamabad, Pakistan, 14 April 2011.

[20] Channel Islands Competition and Regulatory Authorities. "Measurements of Quality of Telecommunication Service in the Channel Islands". CICRA 13/14 Consultant Paper. Channel Islands, March 2013.

[21] Peraturan Menteri Komunikasi dan Informasi nomor 12/PER/M.KOMINFO/04/2008. "Standar Kualitas Pelayanan Jasa Teleponi Dasar pada Jaringan Bergerak Selular". Jakarta. 2008.

[22] Suci Ramadona, Sigit Haryadi. "Over The Top Chat Service Key Performance Indicator", 1st International Conference on Wireless and Telematics ITB, 2015.

Hubber Strategy and Regulation

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***Abstract*—Hubber system is proposed as a solution for Indonesian Telecommunications problem in the era of global trend, where the sustainability of Indonesian Telecommunications operators is threatened as the result of the evolvement of OTT communication service (IP-based) that substitutes the traditional operator's communication services (non IP-based), SMS and telephony. The definition of hubber as proposed is: a hub for IP-based communication between OTT provider or global CDN with Indonesian network provider that is embodied in Telecommunications network and distributed data center infrastructure. As a Point of Interconnection, hubber has administrative aspect, business aspect, and technical aspect. Revenue forecasting has been done to Telkomsel – the largest Indonesian operator, and the result shows that Telkomsel's revenue will go down for the first time in 2015. To achieve positive 7% revenue growth, Telkomsel has to set its revenue target from hubber of 4.83 trillion Rupiahs in 2015 with a steady increase in annual growth and reaching almost a tenfold increase in value, 46.14 trillion in 2020.**

Keywords*— *hubber, OTT, Telecommunications operators

I. INTRODUCTION

The use of Internet has been growing rapidly in recent decades. The development of technologies has led to a trend of new service: OTT (over-the-top) service.

While hubber concept has not been implemented in Indonesia, most Telco services through OTT can be consumed with low price or even free of charge because OTT providers rely their profits from advertisement, application sales, and in-app purchase. OTT services have been widely substituting the legacy services of telecom operators, especially telephone and SMS. Meanwhile, operators need to invest heavily to comply the traffic demand as the result of OTT trend, because the use of OTT is very intense and the delivered contents need very high bandwidth. In contrast, most Indonesian Internet services that are offered by Telco operators and popularly used are time based, for example, unlimited volume per month, so traffic from OTT does not contribute much to Telco operators' revenue.

If this trend continues, the national Telco operators will suffer losses, even though they are also needed by OTT providers to distribute their services to their end consumers. Meanwhile, in Indonesia, there has not been an effective regulation related to the relationship between the Telco operators and the OTT service providers.

II. STUDY OF THE OTT

Over-the-top (OTT) service refers to the form of services provided via the Internet, by utilizing the traditional service provider network canal [1]. In line with the above definition, reference [2] defines OTT providers as service providers offering only Telecommunications services without dealing with Telecommunications networks needed to distribute them, and only rely on the worldwide Internet network.

OTT services can be classified into communication services and content services [1]. OTT providing communication services is currently a problem for the Telco because those services substitute Telco's legacy communication services such as telephony and SMS.

With the presence of OTT, traffic on the network is no longer dominated by the voice traffic, but by the data [3]. These changes in characteristics of the traffic changes the revenue of Telco operators. Telco operators must continue to pay to increase data capacity in order to maintain service quality. Meanwhile, the gain by Telecommunications operators tend to have linear growth. Increased costs are not proportional to the increase in revenue. This makes Telco's existences are being threatened by OTT.

The emerging campaign of net neutrality principle became one factor that makes Telco operators do not have many options to react to the threat of OTT. The definition of net neutrality proposed by Tim Wu is "a principle behind an anti-discrimination in the networks that give users the right to use any network attachments or applications as long as they are not harmful." There is a contradiction between Telco's business benefit and neutrality of the network [4].

III. STRATEGY PLANNING PHILOSOPHY

Increasing use of the Internet service in Indonesia in recent years turned out to be contrary to the national Telco's revenue growth. According to data from Indonesian ICT ministry in 2013, the three major cellular operators in Indonesia, Telkom, Indosat and XL has been experiencing a dominant negative total revenue growth from 2007 to 2011 [5].

In particular, Telkomsel as the largest mobile Telco operator in Indonesia also experienced a decline in revenue growth of a few years back. Analysis of operational data from

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the annual report in the period of 2008 to 2014 revealed that the cause of the decline in revenue growth of Telkomsel is the same as the cause of a decrease in revenue of Telco operators in many countries. The growth of SMS and telephony service usage has been decreasing, on the other hand, data services usage has been experiencing rapid growth. As a consequence, a high investment for data service infrastructures should be performed, and that can be seen from the growing number of BTS 3G/4G from year to year, while revenue from data services, which generally are time based and not volume based, has been slowly growing, and those conditions cause a decrement in ARPU. Because the growth of users tends to decline, the growth of revenue is predicted to be declining as well.

OTT is a service that occupies the data pipe belonging to Telco operators. To access global OTTs, the access to OTT data centers abroad is needed. Communication with the OTT which mostly have a client-server protocol (with or without CDN) requires that a message first sent OTT server and after that is sent to the destination, although the location of the sender and recipient are much closer geographically compared to OTT server location. That is one factor that makes communication in Indonesia inefficient.

OTT-related problems are existed not only in Indonesia, but also in most countries in the world that have Internet access, both developed and developing countries. According to the reference [6], the countries in the world have been making and implementing national policies and regulations regarding OTT services according to their jurisdictions, which tries to answer the problems posed by the presence of OTT. Regulation that is applied in various countries has its respective context, motivation, and environment.

From the philosophy explained above, hubber system design has a main objective to maintain Telco operators' revenue growth by creating new sources of revenue and reducing cost.

In addition to the main objective, hubber is also created to meet additional objectives, namely:

- Enhancing information security and national sovereignty
- Enhancing the resilience of the national economy
- Improving the quality of national Telecommunications

IV. STRATEGY AND REGULATION PLANNING FOR HUBBER SYSTEM

A. *Strategy Planning in Implementing Hubber*

Hubber system is required to bridge the traditional Telecommunications era to the future Telecommunications era. In the era of traditional Telco, access and services are provided solely by Telco operators, and customers only can use services that is provided by the operator which they are subscribed to. In the future of Telecommunications era, customers can use various services from various service providers (SP) as well. This means that OTT services will be shifted upward and become the 'top service'. These changes are illustrated with figure IV.1 and IV.2.

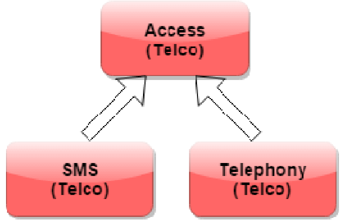

Figure IV.1 Traditional Telecommunications

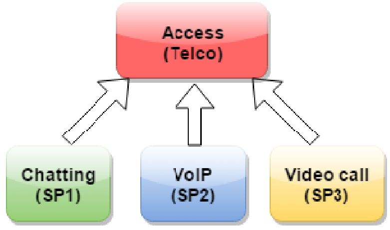

Figure IV.2 Future Telecommunications

B. *Regulation Planning of Hubber System*

Hubber network regulation proposed in this paper covers: the scope of hubber, hubber definition, hubber roles and responsibilities, and hubber business model.

- *Scope of Hubber*
  1. Hubber is an IP-based communications link
  2. Hubber connects Indonesian Telco operators with OTT or global CDN
  3. Hubber is a local CDN
- *Hubber Definition*

Definition of hubber proposed in this paper are: a hub for IP-based data communications between OTT or global CDN provider with Telecommunications providers that provide Internet access to users in Indonesia, which has a special infrastructure consisting of Telecommunications infrastructure and distributed data center. In hubber system, OTT service users in Indonesia subscribe to internet service from Telco operators to get access, and Telco operators connect with OTT services provider through the hubber. Basically, hubber does not manipulate or interpret the data packets communicated between users and OTT providers.

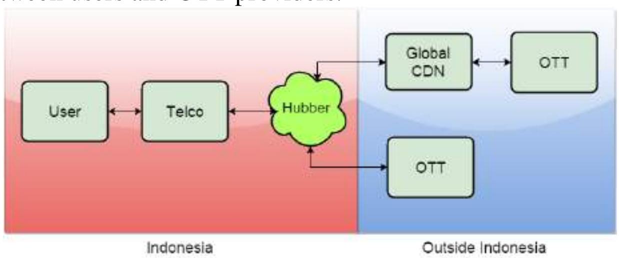

Figure IV.3 Hubber System

In this paper, hubber is also described with the term PoI (Point of Interconnection) between Telco Indonesia with OTT or global CDN. Hubber as a PoI mentioned in this paper has administrative, business, and technical aspect. Physical network connection which supports hubber system is just a part of the technical aspect of the PoI.

- *Roles and Functions of Hubber*

Roles and functions of the hubber in proposed regulation is as follows:

1. Managing the relation between OTT provider and global CDN with national Telco operators
2. Running the local CDN functionality
3. Managing network security management
4. Maintaining QoS and GoS to the customer together with Telco operators

- *Hubber Business Model*

In terms of business, the presence of hubber changes best effort service distributed via the Internet into a new form of service with the QoS. Some OTT providers that choose to connect to their users in Indonesia via hubber can guarantee the quality of their services to their customers in accordance to the selected class of service.

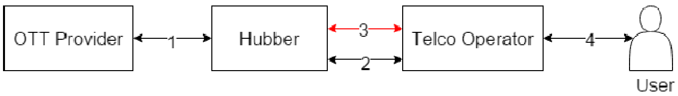

Figure IV.4 Traffic Flows in Hubber System

1. OTT providers can put the contents, data, and applications that are frequently accessed by Indonesian customers on the hubber server. In addition, the traffic from OTT providers to Indonesian users and vice versa are passed through hubber network.
2. Hubber network distributes the traffic from OTT to network operators in Indonesia and vice versa.
3. For communications between users in Indonesia, OTT traffic especially static content is served by hubber server.
4. Customers are connected to the Internet, especially OTT services only through Telco operators.

Furthermore, in the figure IV.5 and IV6, the service exchange and expected fee are illustrated and explained.

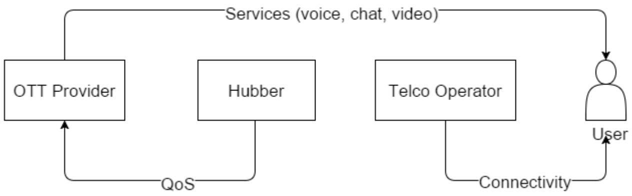

Figure IV.5 Service Interaction in Hubber System

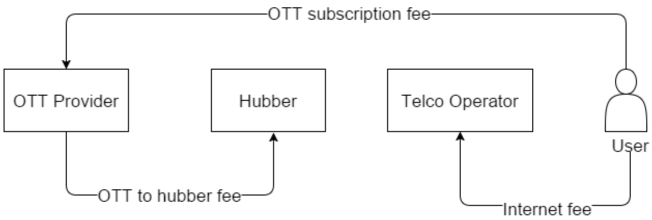

Figure IV.6 Billing in Hubber System

In hubber system, the concept of OTT service subscription fee is introduced. The subscription fee is paid by the customer and then divided to both OTT providers and Telco operators as an additional revenue. The proposed payment mechanism of the OTT subscription fee is using the prepaid or postpaid system that conforms to the Internet subscription mechanism for respective customers. This mechanism chosen in order to facilitate the implementation and supervision of the payment. The payment mechanism is illustrated in figure IV.7.

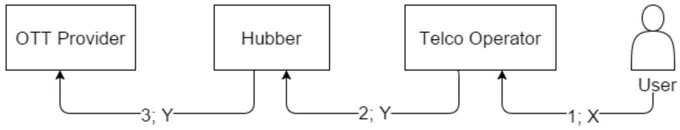

Figure IV.7 Payment mechanism in Hubber System

1. Users make payment with prepaid or postpaid mechanism for Internet services like they do in pre-hubber system. In addition, users also pay a subscription fee for OTT services that they want to use, symbolized as a number X. The amount of X depends on the service level (class of service) selected by the customer.
2. From the payment X, Telco operators received some portion of it as a consequence of the use of their resource in delivering the OTT services. The other portion, called Y, is given to hubber.
3. Y is received by OTT providers as a subscription fee for their services.

The existence of the class of service is also proposed in hubber system, allowing customers to choose the level of service from one OTT provider according to the customer's needs and capability. Examples of the class of service implementation in hubber system is as follows:

Table IV.1 The Example of Class of Service Proposed in Hubber System

CLASS	AVAILABLE SERVICES	CHARGE
3	Only reading	Free
2	Class 3 service + *texting*	Charged
1	Class 2 service +*sharing file*	Charged (higher price)

C. *Hubber Business Simulation*

In this business simulation, a forecast is conducted using data from Telkomsel as a representative Telco operators of Indonesia to determine the Telco operators' revenue target from hubber. The principle used in this forecast is that the loss of income from the Telco operators' legacy services – telephony and SMS – are replaced by income from a new source, in this case, hubber. Telkomsel's total revenue growth determined in advance in accordance with the national GDP growth target, and Telkomsel's revenue target from hubber is determined to cover the loss of revenue from the SMS and telephony, so that Telkomsel's revenue growth target can be achieved. Using national GDP as a reference, the value of 7% is used as the linear target value of the revenue growth.

Forecasting in this simulation is done with time-series extrapolation method of the existing annual growth values [7].

Business simulation calculations yield predictions and targets as shown in the figure IV.8.

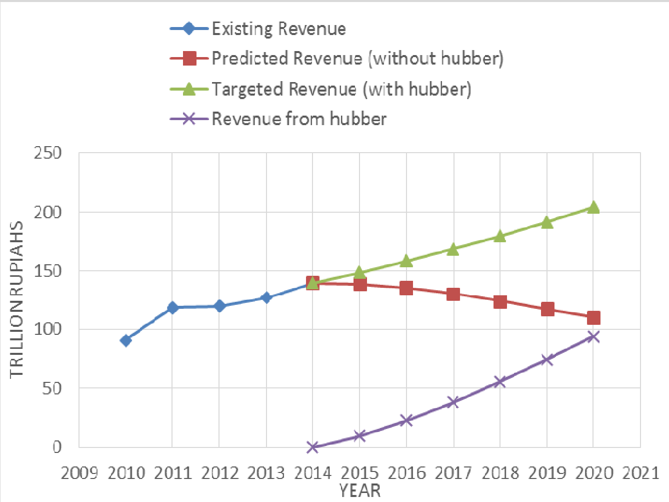

Figure IV.8 Result of Hubber System Business Simulation

From figure IV.8:

- Existing Revenue: revenue value in the annual report.
- Predicted revenue: revenue calculation as the result of predicting and summarizing the values of SMS, telephony, and Internet volume from year to year. This is calculated with assumption that hubber is not implemented.
- Targeted revenue: target values that is obtained by setting the annual growth rate of 7%.
- Revenue from hubber: additional revenue for Telkomsel with assumption that hubber system is applied, originated from the OTT subscription fees from the customers.

From the business simulation that has been done, the revenue target of hubber operator is calculated, and the result is indicated by the the cross-marked line in figure IV.8. It can be seen that the value of operator's revenue from hubber implementation is increasing every year. This is in line with the trend that the use OTT service (IP-based communications) as the main communication services will continue to increase while SMS and telephony will be gradually decreased until completely replaced by IP services.

Table IV.2 Determining Telkomsel's Revenue Target from Hubber

Year	Targeted Revenue (Trillion Rupiahs)
2015	4.83
2016	11.22
2017	18.81
2018	27.32
2019	36.48
2020	46.14

For the next calculation, proposed a 50:50 profit sharing between OTT and Telkomsel. This assumption is used because of the initial idea where OTT and Telco have the same significant role in operating communications services. Thus, total revenues from hubber implementation can be determined by summing Telkomsel's revenue target from hubber and OTT provider's target revenue from huber.

Table IV.3 Determining Hubber ARPU

Year	Total revenue from hubber implementation (Trillion Rupiahs)	Data enabled customer (thousands)	Hubber ARPU (Rupiahs per months)
2015	9.65	71791.30	11204.10
2016	22.43	74539.11	25080.57
2017	37.63	76518.46	40977.48
2018	54.63	77983.57	58379.77
2019	72.97	79094.35	76880.18
2020	92.27	79954.30	96173.13

Hubber ARPU as shown in the rightmost column of table IV.3 is the cost that a customer is expected to pay in addition to Internet cost each month. Hubber ARPU value is expected to increase every year. This ARPU hubber is the expected average of the total subscription cost of a customer for all OTTs that is used, because in the OTT era a customer can subscribe to more than one OTT providers. Therefore, the more OTT providers are connected via hubber, the more affordable is target subscription fee per OTT.

The increasing hubber ARPU target is in line with the increasing hubber deployment target, so that OTT services with guaranteed QoS can be enjoyed and the better quality of OTT communication services can be experienced by more customers in Indonesia.

V. Conclusion

Hubber system is needed in Indonesia to maintain the existence of Indonesian telecom operators facing the global trend where non IP-based communication services are gradually replaced by IP-based communication services provided by OTT providers.

Hubber system in Indonesia is proposed to comply these regulation points:

- Huber definition: a hub for IP-based data communications between OTT or global CDN provider with Telecommunications providers that provide Internet access to users in Indonesia, which has a special infrastructure consisting of Telecommunications infrastructure and distributed data center.
- Roles of hubber: managing the relationship between OTT or Global CDN with national Telco, performing the function of a local CDN, managing network security management, and maintaining QoS and GoS services to the customer.
- In the new business model, hubber system offers changes from today OTT services which is best effort via the Internet into a service that has a QoS. This is manifested by the OTT subscription fee which is then divided as revenue for both OTT providers and Telco operators. Meanwhile, traffic from OTT providers who do not cooperate with hubber still pass through the network as best effort traffic.

Based on the results of business simulation done on Telkomsel, it is predicted that Telkomsel's revenue will go down for the first time in 2015. To achieve positive 7% revenue growth, Telkomsel has to set its revenue target from hubber of 4.83 trillion Rupiahs in 2015 with a steady increase in annual growth and reaching almost a tenfold increase in value, 46.14 trillion in 2020.

REFERENCES

[1] M. K. John Ure, "Discussing the Grey Areas in Regulating OTT Services," in *Telecom Regulatory Affairs Asia*, Singapore, 2013.

[2] E. Bertin, "A Few Myths About Telco and OTT Models," in *Intelligence in Next Generation Networks (ICIN)*, Berlin, 2011.

[3] R. van den Dam and E. Nelson, "Telco 2015: Re-envision the future of mobile communications," in *Wireless Telecommunications Symposium (WTS)*, New York, 2011.

[4] F.-Y. Ling, "Research on the Net neutrality: the Case of Comcast Blocking," in *International Conference on Advanced Computer Theory and Engineering (lCACTE)*, 2013.

[5] M. B. Setiawan, "Pemanfaatan Spektrum Frekuensi Radio dan Orbit Satelit Dalam Mendorong Pembangunan Broadband," Kementrian Komunikasi dan Informatika, Jakarta, 2013.

[6] K. Jayakar and E.-A. Park, "Emerging frameworks for regulation of Over-The-Top services on mobile networks: An international comparison," 2014.

[7] J. S. Armstrong, "Extrapolation for Time-Series and Cross-Sectional Data," in *Principles of Forecasting: A Handbook for Researchers and Practitioners*, Kluwer, 2001.

Over The Top Call Service Key Performance Indicator

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***Abstract*—Over The Top (OTT) call service is one of part of OTT services. This technology is the rapid penetration of large number of internet service worldwide. With the rapid amount of number of user, the regulator and telecommunication provider must have key performance indicator referred to as Key Performance Indicator (KPI). However, the development of KPI still stays in the conceptual stage, does not define the specific parameter as the main indicator for the network performance. This research give a comprehensive and scientific measuring to proposes definition criteria of KPI for the OTT call service technology in order to give recommendation for the regulator and telecommunication provider.**

Keywords— OTT, call service, KPI; network performance

I. Introduction

There are many emerging technologies exist nowadays for mobile phone, either the services or the network. For the service, instead of using regular telephone call, user more prefers using internet packet data today. This technology called OTT (Over The Top) call service. This service is one of the examples of OTT service besides chat and video call to fulfill those users who prefer mobile telephony call service.

In addition of the existence of OTT service, the presence of the 4^{th} Generation (4^{th} G) for mobile network technology in Indonesia is creating new hope for the users that need high speed mobile network.

With the advent of new technologies and services that are applied in Indonesia, the regulator and provider have no rules yet about KPI (key performance indicator) standards for OTT call service using 4^{th} G technology. Whereas KPI is a key indicator in network performance to be followed and determines whether the 4^{th} G network and telephony service as OTT call service given by the provider is already good. KPI divided into two important things that have the parameter determinants in each of them, the performance indicators and target value. Both relate to the quality of the network and the quality of each service-provided.

This research provides KPI recommendation to the Indonesia's regulator and provider based on measuring the performance indicator and determining the target value using one of OTT services, WhatsApp Call, using 4^{th} G mobile network technology.

II. Over The Top Call Service

A. 4^{th} Generation

As seen on Fig. 1, three main components on 4G architecture, UE (User Equipment), E-UTRAN (Evolved UMTS Terrestrial Radio Access Network), and EPC (Evolved Packet Core).

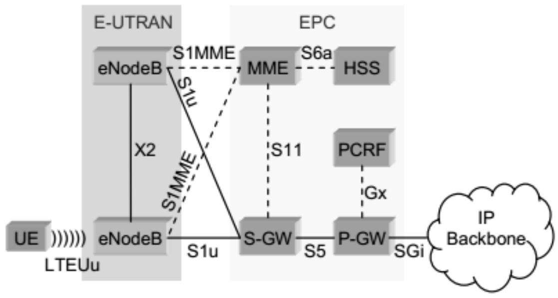

Fig. 1. 4^{th} G Architechture [1]

E-UTRAN has a function to handle radio access side from UE to core network. It has eNodeB (evolved node B) as a base station which is combining the function of Node B and RNC.

Inside EPC, there are MME (mobility Management Entity), SGW (Serving Gateway), HSS (Home Subscription Service), PCRF (Policy and Charging Rules Function), and PDN-GW (Packet Data Network Gateway). It's a new system on cellular communication which the core network is using all-IP.

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B. *Service Assurance*

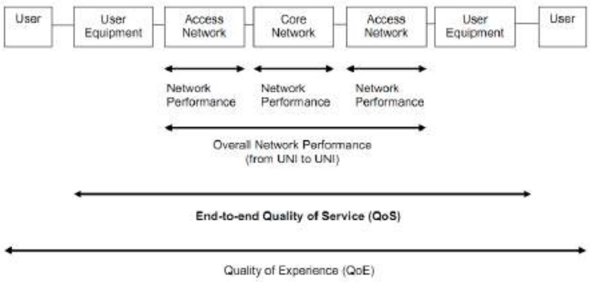

Fig. 2. Classification of telecommunication performance service [2]

Based on ITU-T recommendation E.800 [2], quality performance in telecommunications field can be divided into several levels as follows on Fig. 2. As you can see, the lowest level is Quality of Experience (QoE), because its measure the performance service of end-to-end Quality of Service (QoS).

QoS parameters used in this research will use which set by ETSI and ITU-T for telephony service. Those parameters are used is because no specific QoS parameter for OTT service yet.

C. *Over The Top (OTT) Call Service Architecture*

This research focused on OTT WhatsApp Call service on 4th G mobile network.

Fig. 3. WhatsApp Call topology on 4G mobile netwok.

- In Indonesia, only two operators that have used 4G mobile network technology. This part is has already explain before.
- International internet network called as IMS (IP Multimedia Subsystem) which is integrated on telecommunication network as a carrier. It provides a framework for IP-based mobile network, including Voice over IP (VoIP), instant Messaging (IM), etc.
- WhatsApp Call network. Currently, there are no real information the topology of OTT WhatsApp Call. The information obtained is only that WhatsApp Inc. located in Santa Clara, California.

D. *Performance Parameter*

OTT service parameter for telephony service not currently listed in ETSI and ITU-T, therefore this research is to propose it as a hypothesis. We use non-OTT telephony parameter according to ETSI TR 125 913 [3] and ITU-T, and will be examined which one will be selecting from non-telephony service parameters into the parameters for OTT WhatsApp Call service.

1) *Telephony Service Non- Accessibility* [%]

This parameter calculate the probability that end- user can't get service from the network when the indicator on the mobile telephone network showed in good condition. Measurements were made with the condition B-party is not in a busy state and A-party is not in error condition. Calls attempts can be said to be successful, if A-party hearing the dial-tone and the mobile telephone of B-party is ringing.

2) *Telephony Setup Time* [s] *=PDD*

This parameter is account the duration between the user get the dial tone after press the call button. Telephony setup time does not take into account the elapsed time by a signal failure.

3) *Telephony Speech Quality*

This parameter is an indicator that represents the quality of speech transmission end-to-end cell phone service. It have two kind of measurement method.

- *On Call and Sample Basis*
- *On GOS*

4) *Telephony Cut-off Call Ratio [s]=Drop Call*

Its calculation of the probability of a successful call which not terminated by the termination is done both A-party and B-party.

5) *Telephony CLIP Failure Ratio[%]*

Telephony CLIP Failure Ratio calculation of the percentage of call setup in which the parameters of CPN (Calling Party Number) valid has been send but not received in full [4].

III. RESEARCH SCHEME

This research measure the performance indicator and branch mark the target value of call service using the OTT WhatsApp Call 4G mobile network in Bandung-Indonesia by doing several stages as shown on Fig. 4.

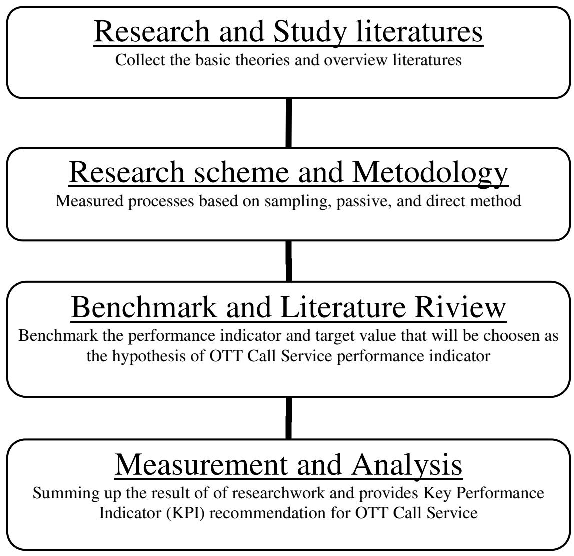

Fig. 4. Research scheme

A. Measurement methods

There are several methods to measure the QoS of a telecommunication service [5]. In this research, we are using three of them:

1) Based on the data source, Sampling method: measuring QoS by taking a sampling of data from a service at a given time or over a range of specific work processes.

2) Based on direct measurements whether or not do,Direct method: measurement of QoS with traffic directly observing conditions at some points in the network architecture.

3) Based on involvement of the observer, Passive method: is the measurement of QoS with the collection of information without any interference on the network.

B. Performance parameter indicator

We are selecting the parameters of OTT call service from telephony service parameters. We use these parameter indicators for OTT call service parameter as a recommendation.

1) Telephony Speech Quality

- *On GOS Basis*

This telephony voice quality calculation procedure is using R factor [7]. Factor R can be transformed into a quality index as Mean Opinion Score (MOS), percentage Good or Better (Gob), or percentage Poor or Worse (POW).

$$R = R_0 - I_s - I_d - I_{e.eff} + A \quad (1)$$

Where:

R_o is representing as the SNR (signal-to-noise-ratio), including noise sources such as noise circuit and room noise. I_s is a combination of all the distractions that occur simultaneously with the voice signal. I_d factor is a disorder caused by the delay. $I_{e.eff}$ (effective equipment impairment factor) represents the interference caused by the low bit rate codec, and the disruption caused by loss of packet at random. A factor is an advantage that allows the compensation of the factors that caused the disruption another advantage of the user access [6].

The sound quality telephony can be calculated as follows:

- Percentage Good or Better (Gob)

$$GoB(\%) = \frac{R - 60}{16} * 100 \quad (2)$$

- Percentage Poor or Worse (POW)

$$PoW(\%) = \frac{45 - R}{16} * 100 \quad (3)$$

- MOS_{CQE} (MOS Conversational Quality Estimate)

For R < 0, than $MOS_{CQE} = 1$

For 0 < R < 0, than:

$$MOS_{CQE} = 1+0.035R+R*(R-60) * 7x10^{-6} \quad (4)$$

2) Telephony Cut-off Call Ratio [s]=Drop Call

The measurement of telephony cut-off call ratio begins when alerting or busy tone heard by the A-party coming from B-party [6].

$$Telephony\ Cut-off\ Call\ Ratio\ [\%] = \frac{unintentionally\ terminated\ telephony\ calls}{all\ sucessful\ telephony\ call\ attempts}\ x\ 100 \quad (5)$$

IV. BENCHMARK AND LITERATURE RIVIEW

A. Procedure of Key Performance Indicator(KPI) Call Serice Determination

In general, KPI are divided in two groups, which are performance indicator and target value. Both are related to the quality of the network and the quality of each service.

KPI arranged on several considerations [6].

- The 1st consideration is the international standards and recommendation of the ITU-T, ETSI, IEEE, ANSI, and others.
- The 2nd comparing standards of various telecom operators and National Regulatory Bodies in the world.
- The 3rd consideration is the National Regulatory Body on each country, and
- The 4th consideration is the SLA (service level agreement) between providers and costumers.

B. Parameter of Key Performance Indicator (KPI) refer to Benchmark and Literature Riview

There are so many research [4], [7-10] efforts focusing on QoS measurement for end-to-end of the user side, but none of them describes how to measure the QoS and the data parameter (KPI) of the OTT call service.

Key Performance Indicator of OTT call service or particularly WhatsApp Call cannot be found in a variety of telecommunication institutions, telecom vendors, telecom providers or the International Telecommunication regulators. Therefore, we are making slices of the data parameters (KPI) QoS Telephony Speech Quality and Telephony Cut-off Call Ratio, which is obtained from several international telecommunications organizations, both operators, vendors, and international telecommunications regulator. The collection of data is emphasized to the QoS parameters of the call service related to the quality of OTT call service.

From the results of benchmark the parameters (KPI) QoS Telephony Speech Quality and Telephony Cut-off Call Ratio, we obtained a summary table as follows.

Table 1. BENCHMARK OVER THE TOP CALL SERVICE

	Parameter	Nilai
	Telephony Speech Quality on GOS Basis	
a.	*Pakistan Telecommunications Authority [11]*	> 3
b.	*National Telecommunications Regulatory Authority of Rwanda[12]*	≥ 3
c.	*World Class Operator Vodafone (UK) [13]*	3
d.	*World Class Operator T-Mobile (US) [14]*	3.54
e.	*Nigerian Telecommunication [15]*	≤ 3.6

f.	*Telecom Regulatory Commission of Sri Lanka [16]*	> 4
	Telephony Cut-off Call Ratio [%]	
a.	*National Telecommunications Regulatory Authority of Rwanda [12]*	≤ 1%
b.	*Telecom Regulatory Authority of India [17]*	≤ 2%
c.	*Nigeria Telecommunication [15]*	≤ 0.5%
d.	*Telecom Regulatory Authority of Indonesia [18]*	≤ 5%
e.	*Telecom Regulatory Commission of Sri Lanka [16]*	< 5%
f.	*National Class Operator Indosat [19]*	≤ 5%
g.	*Channel Island Competition and Regulatory Authorities[20]*	< 2%

C. Target Value refer to Benchmark and Literature Riview

Not much research discusses this technology, because the OTT call service technology is quite new. So the data required for setting target value for OTT call service does not exist yet. There is no presence of a standard estimate of the minimum value for each of the parameters of network performance which will be being referred as target value.

V. MEASUREMENT AND ANALYSIS

Based on the measurement of over the top (OTT) call service, the key performance indicators (KPI) are:
With R_0 factor (SNR) is 50dB, distraction 2.5 dB, disorder 1.5 dB, effective equipment impairment factor is 4,0, and compensation factor is 28 dB, we can get:

$$R = R_0 - I_s - I_d - I_{e.eff} + A$$

$$R = 70\text{dB}$$

$$\text{GoB} = 62{,}5\%,\ \text{PoW} = -156.25\%,\ \text{MOS}_{\text{CQE}} = 3.597$$

Based on ITU Forum [21] and Telecommunication regulatory commission of Sri Lanka [16], this performance indicator for speech transmission quality category (Telephony Speech Quality) that we get is on medium level, which mean some users in Bandung feel dissatisfied.

For the calculation of telephony cut-off call ratio that we measure in Bandung, we get around 2%. Its mean that this performance is still around standardization the range of good value limit set by various national telecom regulator and provider.

VI. CONCLUTION

Based on calculation of OTT call service that we measure in Bandung using 4th G OTT WhatsApp Call service on three methods (sampling, direct and passive), and referred to the hypothesis benchmark of the target value, we concludes that the quality of this service is adequate.

QoS from telephone services by the user is difficult to obtain the result scientifically. The measurement of MOS using questioner was distorted by costumer's subjectivity. MOS measurement using the technical parameters only an estimate numbers because some parameters are empirical figures. An accurate drop call measurement can only be done by way of aggregate that can only be done by operator.

References

[1] Sonia Farconi and Manuela Vaser, "4G LTE Architectural and Functional Model of Video streaming and VoLTE Services," 7th International Conference Ubiquitous and Future Networks (ICUFN), 2015.

[2] ITU-T Recommendation E.800 : Definitions of terms related to quality of service.

[3] ETSI TR 125 913 V7.3.0 (2013-04: Universal Mobile Telecommunication System (UMTS): LTE; Requirements for Evolved UTRA (E-UTRA) ad Evolved UTRAN (E-UTRAN).

[4] Sigit Haryadi, Ivantius Limampauw, "QoS Measurement of Telephony Services in 3G Networks Using aggregation Method", The International Conference on Telecommunication System Service and Applications (TSSA), 2012.

[5] Strohmeier, Felix; Dorken, Heinz; Hechenleitner, Bernhard. (2001): AQUILA Distributed QoS Measurement. 2001.

[6] Sigit Haryadi , "Telecommunication Sevice and Experience Quality", ISBN: 978-602-18578-6-1, Jauary 2013

[7] Sigit Haryadi, Sandy Akbar Nusantara, "QoS Measurement of Web Browsing Services in 3G Networks Using aggregation Method", The International Conference on Telecommunication System Service and Applications (TSSA), 2012.

[8] Sigit Haryadi, Raisha Andina, "QoS Measurement of File transfer Protocol Services in 3G Networks Using aggregation Method", The International Conference on Telecommunication System Service and Applications (TSSA), 2012.

[9] Sigit Haryadi, Arnold Pramudita, "QoS Measurement of Video Streaming Services in 3G Networks Using aggregation Method", The International Conference on Telecommunication System Service and Applications (TSSA), 2012.

[10] Maya Rahayu, Sigit Haryadi, "Over The Top Chat Sevice Key Performance Indicator", International Conference on Wireless and Telematics ITB. 2015.

[11] Pakistan Telecommunication Authority Notification, Islamabad, Pakistan, 14 April 2011.

[12] Rawanda Utilities Regulatory Agency, 'Regulation for Quality of Service of celular mobile and fixed network services. February 2013.

[13] P3 Mobile Benchmark, "The P3 Mobile Benchmark UK", 2014.

[14] P3 Mobile Benchmark, "The P3 Mobile Benchmark US", 2014.

[15] Nigerian Communication ACT (no 19 of 2003) Quality of Service Regulations, 2012.

[16] Telecommunication regulatory commission of Sri Lanka, "Counsultation Paper on Esstablishing Quality of Service Regualtory Framework for Fixed and Mobile Voice Services", December 2013.

[17] Telecom Regualtory of India, "The Indian Telecom Service Performance Indicators", New Delhi, 2013.

[18] Peraturan Menteri Komunikasi dan Informatika Republik Indonesia no 16 tahun 2013, "Standar Kualitas Pelayanan Jasa Teleponi Dasar Pada Jaringan Bergerak Seluler", 2013.

[19] National Class Operator Indosat "Pencapaian Standar Kualitas Pelayanan Jasa Teleponi Dasar Pada Jaringan Bergerak Selular", Juli 2014.

[20] Channel Island Copetition and Regulatory Authorities, "Measure of Quality of Teleommunication Services in the Channel Island", CICRA 13/14 Consulation Paper. Channel Island, March 2013.

[21] Dr. Ing. H. W. Gierlich, " The importance of models and procedures for planning, monitoring and control in the provision of communications services", CITEL (PCC.I)/ ITU Forum on Information and Communication Technology Service: Quality, Control, and Surveillance, Colombia, September 2013.

Broadband User Demand Forecasting in Indonesia based on Fourier Analysis

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Abstract— **In this paper, a broadband user demand forecasting in Indonesia based on Fourier analysis is presented. In this approach, the historical data of broadband user demand is considered as time-series data. To obtain trend of the time-series data, polynomial expression is incorporated in the Fourier series representation instead of constant. The presented forecasting model is then confirmed its validity by using hypotethical daily price fluctuation data. It can be shown that the presented forecasting model gives good agreement between forecasted data and actual data. The same forecasting model then is applied to forecast broadband user demand in Indonesia. Several cases of fixed and mobile broadband are presented. It can be shown that the forecasting model based on Fourier analysis gives good result with the error of the estimated value is less than 10%.**

Keywords—forecasting, fourier, broadband, Indonesia

I. Introduction

Successful market penetration of mobile-gadget, for example smart-phone and tablet devices, has driven enormous growth of data-traffic internet. Several facts confirm this situation: global mobile data grew 69% in a single year from 1.5 exabytes per month in 2013 to 2.5 exabytes per month in 2014; average smart-phone monthly mobile data traffic grew 45% from 536 Megabytes (MB) in 2013 to 819 MB in 2014; average monthly mobile data traffic for laptop was 2600 MB, in comparison, average monthly mobile data traffic for mobile-connected tablet was 2076 MB[1]. To put into another perspective, mobile data traffic in 2014 only was nearly 30 times of the size of entire internet network back in year 2000, moreover, the number of mobile-connected devices exceeded the the world's population in 2014[1].

In Indonesia, total number subscriber of top three operator has reached approximately 250 million subscriber in cumulative. This number is almost similar with the number of population in Indonesia. Looking more into details, the density for fixed-line and mobile connection are 15.9% and 97.7%, respectively. While the density of fixed and mobile broadband are 1.1% and 22.2%, respectively[2]. Such big number of market also has significant number of mobile data broadband user. Therefore, The Government of Indonesia has put special attention to the development of broadband in Indonesia[2][3]. The targets to be achieved by 2019 are, for cities, broadband access infrastructure reaches the penetration level of 30% from the total population, 71% from the total households with speed of 20 Megabytes per second (Mbps), 10% from total office buildings with speed of 1 Gbps. While in the rural areas, broadband access infrastructure reaches the penetration level of 6% from the total population, 49% from the total households with speed of 10 Mbps[3]. Another target to be achieved in 2019 is that the service fee should be maximum 5% of the montly average income per capita[3]. These data and targets shows that broadband growth has become main interest from the regulator and provider point of view due to massive prospective demand today.

In order to fulfil the broadband demand, the provider should prepare some infrastructures which able to accomodate the broadband growth in Indonesia. However, some additional infrastructures mean additional costly investment. Over-planning will burden the provider financially, while lack of planning will reduce the opportunities and revenues. Therefore, accurate broadband user forecasting is needed to obtain accurate estimation of the growth of user, which will have consequence in the cost of investment that has to be prepared by the provided.

Broadband forecasting method for predicting growth has been presented by several authors. Bellcore Data Market Demand Model (BDMDM) was used to do forecast aggregation while Broadband Internode Forecasting Tools (BIFT) was used to do forecast disaggregation in [4]. Multi-faceted approach was used to forecast by considering statistical and historical analysis in [5]. Analytical method based on quantitative series forecasting was developed in [6] to forecast with considering quantitative factor of telecommunication service life-cycle. Gompertz function-based S-curve was used to forecast the mobile broadband traffic in [7]. Compound annual growth rate (CAGR) based forecasting is presented in [8]. Although those methods provide good accuracy, however, some of them are using relatively many parameters which may be difficult to obtain for forecasting broadband demand user.

In this paper, broadband demand forecasting for Indonesia is presented in simpler way by using fourier analysis. By using this approach, the historical data of broadband demand is considered as time-series data. Some common methods for time-series forecasting techniques are the autoregressive (AR), the moving average (MA), and its derivative combinations such as Auto

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Regressive Moving Average (ARMA), and Auto Regressive Integrated Moving Average (ARIMA)[9]. These techniques analyses time-series data under assumptions that the underlying stochastic process is stationary[10]. Other time series method is Generalized Auto Regressive Conditional Heteroskedasticity (GARCH) which is the generalized form of Exponentially Weighted Moving Average (EWMA) model[9]. Fourier analysis is considered as spectral analysis, alongside with wavelet analysis. These techniques is powerful to asses the anomalies in time of a time-series data[10]. For example, to predict regular repetitive characteristic of a time-series data. Accurate modeling of a certain time-series waveform function by Fourier analysis approach can be extended to predict the future outcome. Fourier-based forecasting has been applied for several time-series condition, for example in market demands [11][12], electrical peak load [13], stock market [14][15], and many more. As far as the authors concern, broadband demand forecasting based on fourier analysis in Indonesia has not been found in any literature.

This paper is organized as follows, in Section II the basic knowledge of Fourier analysis is presented, then the forecasting method based on Fourier analysis is discussed. In Section III, the presented forecasting model based on Fourier analysis is confirmed its validity by using hypotethical data of daily price fluctuation for time interval of one and quarter year. Section IV presents the broadband user demand forecasting in Indonesia by using presented forecasting model. Several cases of fixed and mobile broadband forecasting is presented in this section. Section V will conclude this paper.

II. FOURIER ANALYSIS-BASED FOR FORECASTING

Fourier analysis is based on the underlying concept that any arbitrary signal can be represented as linear combination of sinusoids over finite range of time. To represent such arbitrary signal, it is possible to sum constant and sines and cosines at integer multiples of the fundamental frequency with right amplitudes in that finite time interval[16]. The Fourier series in trigonometric form is shown in Eq. (1) – (5).

$$y(t) = a_0 + \sum\nolimits_n (a_n \cos(n\omega t) + b_n \sin(n\omega t)) \quad (1)$$

$$a_0 = \frac{1}{T}\int_{t_0}^{t_0+T} y(t)\, dt \quad (2)$$

$$a_n = \frac{2}{T}\int_{t_0}^{t_0+T} y(t)\cos(n\omega \mathrm{t}) dt \quad (3)$$

$$b_n = \frac{2}{T}\int_{t_0}^{t_0+T} y(t)\sin(n\omega \mathrm{t}) dt \quad (4)$$

$$\omega = \frac{2\pi}{T} \quad (5)$$

Where a_0, a_n, and b_n are the Fourier coefficients of signal $y(t)$. a_0 is any offset of the time-series data, usually as direct-current (DC) or constant form. n is the integer multiples of the fundamental frequency ($n = 1$), or it is also usually called as order of harmonics. ω is the frequency of the signal.

In order to create model for forecasting based on Fourier analysis, slightly different approach is used in term of a_0. The offset here represents the trends of the observation signal, therefore term a_0 may not in the form of constant or DC any longer, instead it can be in time-function signal. To approach this, a polynomial expression in a finite time interval is chosen for term a_0 to obtain more general term. The new term $a_0(t)$ is shown in Eq. (6).

$$a_0(t) = m_k t^k + m_{k-1} t^{k-1} + \cdots + m_2 t^2 + m_1 t + m_0 \quad (6)$$

Where m is the coefficient of the polynomial, and k is the order of the polynomial. To obtain this trends, polynomial curve fitting can be applied to the observed time-series data.

To forecast the observed time-series data, firstly the trend obtained from polynomial curve fitting is substracted from the observed time-series data. Suppose that $a_0(t)$ is obtained from polynomial curve fitting analysis, therefore the fluctuation of the observed time-series data $y_f(t)$ is shown in Eq. 7.

$$y_f(t) = y(t) - a_0(t) \quad (7)$$

From the obtained fluctuation time-series data, it can be resolved into trigonometrical form of Fourier series as shown in Eq. (8).

$$y_f(t) = \sum\nolimits_n (a_n \cos(n\omega t) + b_n \sin(n\omega t)) = \hat{y}_{fn}(t) \quad (8)$$

The predicted model $f_n(t)$ is obtained by the trends and the Fourier coefficients as shown in Eq. (9).

$$f_n(t) = a_0(t) + \hat{y}_{fn}(t) \quad (9)$$

By extending time t more than the finite interval time used in the observed time-series data, thus we able to forecast the observed data to the future.

III. PERFORMANCE VALIDATION BY USING DATA TESTING

To confirm the presented Fourier analysis for forecasting, a hypotethical data of price fluctuation during time interval one and quarter year is selected. Figure 1 shows the time-series graph of price fluctuation during selected time interval. It can be seen that the time-series data of price has natural characteristic of periodical fluctuation. To verify the presented method, the data in Fig. 1 is divided into two parts, the first 208 days data are used as the basis for forecasting, while the rest of data will be used for validation purposes.

The forecasting begin with trend approximation and finding the fluctuation only, as in Eq. (6) and (7). Clearly from Fig. 1, the trend of the data is linear, therefore the polynomial in Eq. (6) can be approximated by first order polynomial trend. Figure 2 shows the trend and the fluctuation of the price for the first 208 days data. Then, the fluctuation part will be resolved into Fourier series according to Eq. (8). The result of Fourier resolving is shown in Fig. 3. It can be seen that the data of price has regular periodical fluctuation, the biggest is the low-days fluctuation, as can be seen directly in Fig. 1. Also significant regular periodical fluctuation at 25 and 50 cycle of days.

The Fourier representation obtained in the previous step then re-combined with the trends to obtain the data signal model

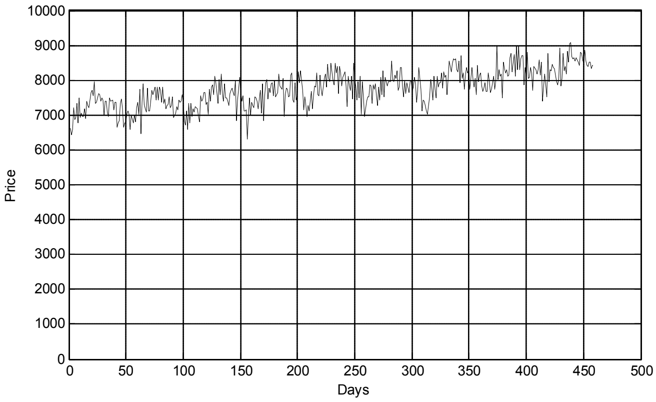

Figure 1. Price fluctuation during one and quarter year

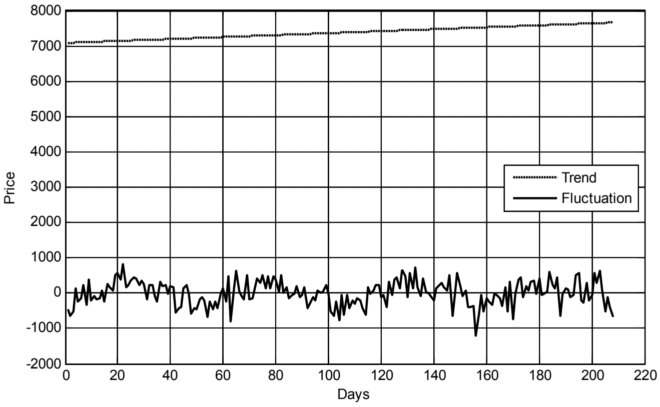

Figure 2. Trend and fluctuation of data price

Table 1. Fixed-broadband user (Case: Speedy)

Year	Broadband user
2004	11.000
2005	31.000
2006	93.000
2007	241.000
2008	645.000
2009	1.145.000
2010	1.649.000
2011	1.789.000
2012	2.341.000
2013	3.013.000
2014	3.400.000

Table 2. Mobile-broadband user (Case 1: Telkomsel)

Year	Broadband user
2008	6.200.000
2009	12.600.000
2010	21.685.000
2011	39.822.000
2012	54.611.000
2013	60.531.000
2014	67.860.000

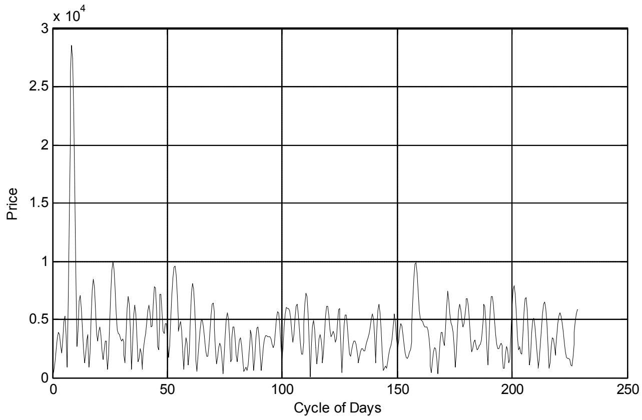

Figure 3. Spectral of cycle of data price

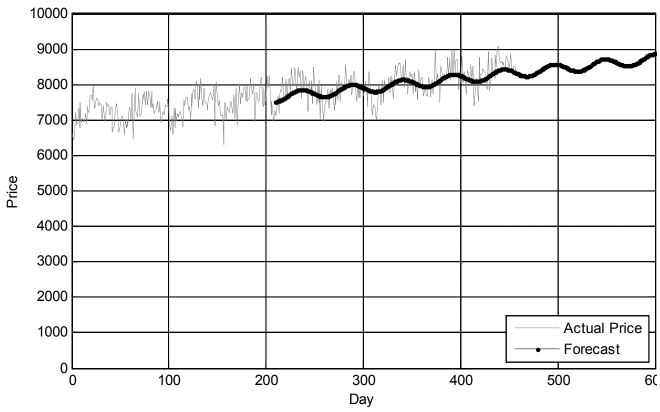

Figure 4. Forecasting based on Fourier analysis for data price

according to Eq. (9). The forecasting will begin for the 209th day, based on the information obtained from 1st – 208th days. The available data for days 209 to end will be used as the validator for the forecasting by Fourier analysis. Figure 4 shows the forecasting obtained from Fourier model imposed with the actual data price. The forecasting model based on Fourier analysis gives good forecasting result, as confirmed by the validation using 209th – end of the actual data price.

IV. BROADBAND DEMAND FORECASTING

The presented forecasting method based on Fourier analysis will be applied to the broadband demand forecasting in Indonesia. Several cases will be estimated by using the presented method. Fixed-broadband demand will be represented by using the data of Speedy by Telkom[17], while mobile-broadband demand will be represented by using the data of Telkomsel[18], XL[19], and Indosat[20]. Table 1 – 4 show the available data of broadband user for each provider. The forecasting will use the data up to 2013 only, while data 2014 is reserved for validation of the presented forecasting model.

By using the same approach as has been discussed in Section II and III, the broadband user data as shown in Table 1 – 4 are forecasted based on Fourier analysis. The results are shown in Table 5, where k is the polynomial order as in Eq. (6) and n is the harmonic order of Fourier series as in Eq. 8. Using the data at year 2014 as the basis validation point, it can be seen that the

Table 3. Mobile-broadband user (Case 2: XL)

Year	Broadband user
2007	15.469.000
2008	26.016.000
2009	31.438.000
2010	40.351.000
2011	46.359.000
2012	45.750.000
2013	60.549.000
2014	59.643.000

Table 4. Mobile-broadband user (Case 3: Indosat)

Year	Broadband user
2010	24.091.000
2011	25.282.000
2012	26.674.000
2013	29.056.000
2014	30.046.000

Table 5. Fourier-based forecasting result

Provider	k	n	Error
Speedy	2	2	7.08%
Telkomsel	1	2	9.39%
XL	2	5	2.85%
Indosat	1	3	1.47%

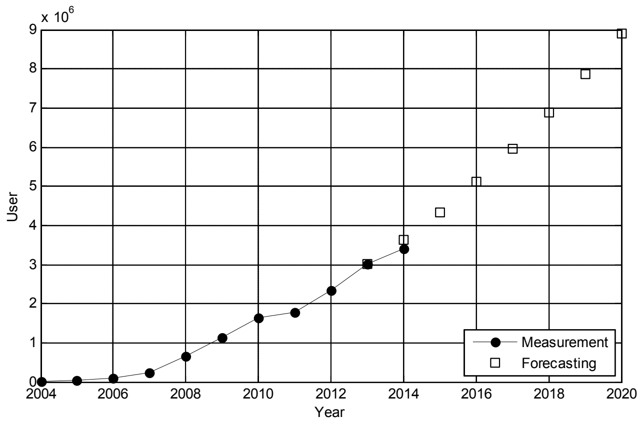

Figure 4. Fixed-broadband user forecasting (Case: Speedy)

presented forecasting method gives good estimation results that has error less than 10%. Expansion of the forecast for 2015 to 2020 by using the parameters as shown in Table 5 is shown in illustrative figures in Fig. 5 – 8. Figure 5 shows the fixed-broadband user forecasting in Indonesia up to year 2020. Figure 6 – 8 show mobile-broadband user forecasting in Indonesia up to year 2020 of Telkomsel, XL, and Indosat, respectively.

V. Conclusions

A forecasting method based on Fourier analysis for broadband user in Indonesia has been presented. The presented forecasting model has been verified by using hypotethical data of price fluctuation for time interval one and half year. The presented forecasting model shows good agreement to forecast the price fluctuation. The forecasting model then applied to broadband user demand in Indonesia for several cases, which are fixed and mobile broadband. The results show that by using the presented forecasting model, the error is less than 10% for the basis of year 2014 data. Forecasting for year 2015 – 2020 then has been conducted.

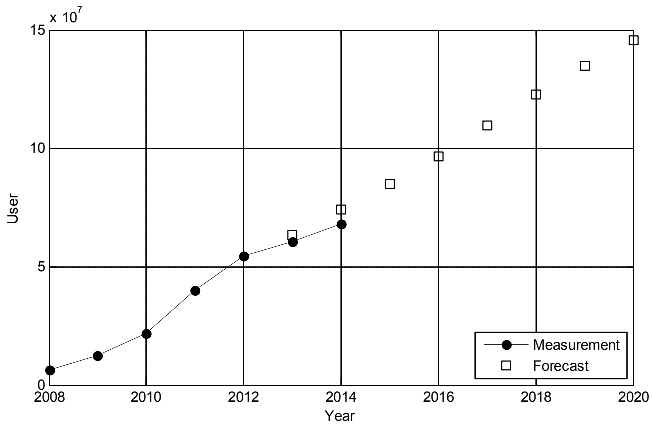

Figure 1. Mobile-broadband user forecasting (Case 1: Telkomsel)

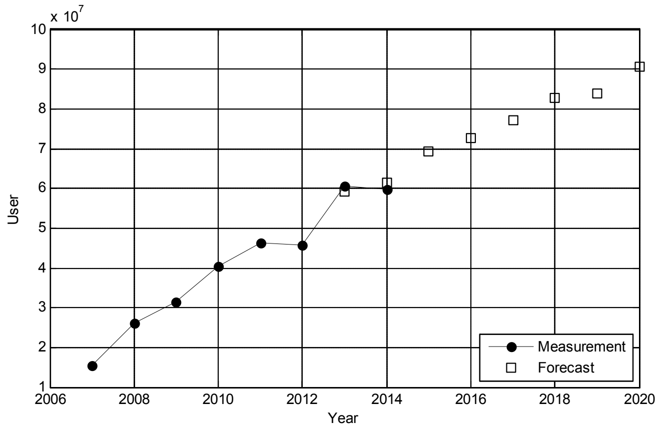

Figure 2. Mobile-broadband user forecasting (Case 2: XL)

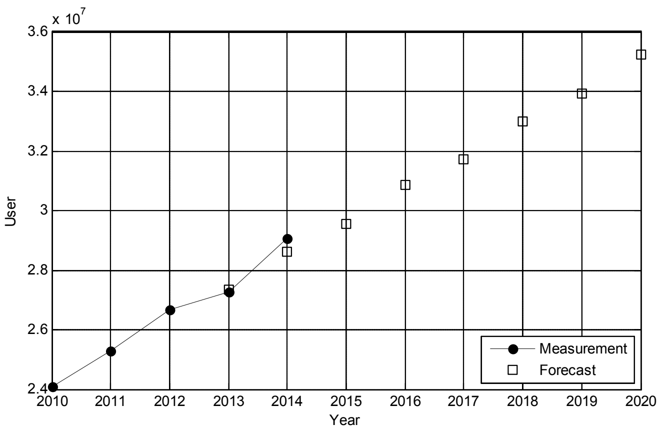

Figure 3. Mobile-broadband user forecasting (Case 3: Indosat)

REFERENCES

[1] --, “Cisco Visual Networking Index: Global Mobile Data Traffic Forecast Update, 2014 – 2019”, *White Paper Cisco Public*, 3 February 2015.

[2] --, “Indonesia Broadband Plan 2014 – 2019, Connetivity, Innovation, Transformation”, *Ministry of National Development Planning Republic of Indonesia*, 2014.

[3] --, “Government Regulation No. 96 Year 2014 about Indonesia Broadband Plan 2014 – 2019”, *Office of Cabinet Secretary of Republic Indonesia*, 2014.

[4] O. J. Wasem, A. M. Gross, and G. A. Tlapa, “Forecasting Broadband Demand Between Geographic Areas”, *IEEE Communication Magazine*, Vol. 33, Iss. 2, 1995, pp. 50 -57.

[5] M. Hopkins, G. Louth, H. Bayley, R. Yellon, A. Ajibulu, and M. Niva, “A Multi-Faceted Approach to Forecasting Broadband Demand and Traffic”, *IEEE Communications Magazine*, Vol. 33, Iss. 2, 1995, pp. 36 – 42.

[6] M. Sokole, “Analytical Method for Forecasting of Telecommunications Service Life-Cycle Quantitative Factors”, Ph.D. dissertation, Faculty of Electrical Engineering and Computing, University of Zagreb, Zagreb, Croatia, 2009.

[7] I. Z. Kovacs, P. Morgensen, B. Christensen, and R. Jarvela, “Mobile Broadband Traffic Forecast Modeling for Network Evolution Studies”, *IEEE Vehicular Technology Conference*, 2011, pp. 1 – 5.

[8] M. N. Shuhailie, M. S. Azura, and A. R. Fahmi, “Malaysia’s Communication Content and Infrastructure: The Broadband Demand’s Forecast, Strategies, and Policy Implications”, *16th International Conference on Advanced Communication Technology*, 2014, pp. 244 – 248.

[9] Z. Deljac, M. Kunstic, and B. Spahija, “A Comparison Of Traditional Forecasting Methods For Short-Term And Long-Term Prediction Of Faults In The Broadband Networks”, *The 34th International Convention MIPRO*, 2011, pp. 517 – 522.

[10] Y. Zhong-chang, “Fourier Analysis-Based Air Temperature Movement Analysis And Forecast”, *IET Signal Processing*, Vol. 7, Iss. 1, 2013, pp. 14 – 24.

[11] M. Yohda, M. Saiko-Arita, A. Okada, R. Suzuki, and Y. Kakemoto, “Demand Forecasting by the Neural Network with Discrete Fourier Transform”, *IEEE International Joint Conference on Neural Networks*, Vol. 4, 2004, pp. 2759 – 2763.

[12] A. Fumi, A. Pepe, L. Scarabotti, and M. Schiraldi, “Fourier Analysis for Demand Forecasting in a Fashion Company”, *International Journal of Engineering Business Management: Special Issue on Innovations in Fashion Industry*, Vol. 5, 2013, pp. 1 – 6.

[13] M. Beiraghi, and A. M. Ranjbar, “Discrete Fourier Transform Based Approach to Forecast Monthly Peak Load”, *Asia-Pacific Power and Energy Engineering Conference*, 2011, pp. 1 – 5.

[14] D. Serksnas, “Time Series Model for Predicting Intraday Volatilities”, M.S. Thesis, St. Anne’s College, University of Oxford, Oxfor, UK, 2013.

[15] H. Huang, W. Zhang, G. Deng, and D. Chen, “Predicting Stock Trend Using Fourier Transform and Support Vector Regression*”, IEEE 17th International Conference on Computational and Science Engineering*, 2014, 213 – 216.

[16] M. J. Roberts, “*Signal and Systems: Analysis Using Transform Methods and MATLAB*”, International Ed., Boston: McGraw-Hill, 2004, pp. 208 – 209.

[17] --, “2014 Annual Report: Sustainable Competitive Growth Through Digital Business”, *PT. Telekomunikasi Indonesia (TELKOM*), 2014.

[18] --, “2014 Annual Report: Continuing to Win in the New Digital Era”, *PT. Telekomunikasi Selular (TELKOMSEL)*, 2014.

[19] --, “2014 Annual Report: Unleashing Indonesia’s Potential Through Mobile Internet Leadership”, *PT. XL Axiata (XL)*, 2014.

[20] --, “2014 Annual Report: Enabling A Digital Indonesia”, *PT. Indosat (INDOSAT)*, 2014.

Application and Data Level Interoperability on Virtual Machine in Cloud Computing Environment

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***Abstract*—The rapid development of internet nowadays impact on increasing many applications that utilizing cloud computing technology for the sake of organization. Increasing requirements in applications caused by inevitable business process growth in organization enabling a tendency to switch from old cloud provider to more reliable one. However, in practice, application functionality failures often occur in case of migrating process to new cloud system due to several circumstances e.g. vendor lock-in problem. This paper introduces a new method for system migration testing between two cloud providers. The goal is to determine the interoperability level of application and data in virtual machine within hypervisor system that moves from one cloud provider to another cloud provider. The contribution of this paper is to provide a reference test method for measuring the interoperability between the two cloud systems were migrated.**

Keywords—interoperability; virtual machine; cloud migration

I. INTRODUCTION

The rapid development of internet technology today impact on many organizations that began to shift from using physical infrastructure to cloud computing services to support the business process needs. The emergence of several cloud computing service providers that offer a range of benefits are taking part in the increasing usage of cloud computing technology, especially in the industrial field.

Increasing business processes and operational needs also better additional features offered by other cloud providers such as cheaper price, more reliable system, higher system performance, more secure system and other benefits make organization has tendency to move their service or can be called a migration from one cloud provider to another.

Yashpalsinh and Kirit [9] specified three type services of cloud computing i.e. IaaS (Infrastructure as a Service), SaaS (Software as a Service), and PaaS (Platform as a service). In the term of IaaS environment, the tiniest component that can be moved to another environment is virtual machine [1]. In virtual machine migration process there is a step called virtual machine image conversion that makes the image can be used in other system. However, the success of the virtual machine conversion process from one system to another does not guarantee the success of the migration process. Parameters such as network configuration, CPU, memory, security, application and original data structure may have changed in the migration process. In migrating system process from one cloud provider to other cloud provider in which there are some applications in previous cloud that have been running, needs a testing method to measure the level of interoperability between two cloud systems, so that all components in the previous system such as operating system, security system, application and the data stored, can be run appropriately with its functionality in the new system.

TIOSA [1] is a virtual machine interoperability testing method based on Open Data Center Alliance (ODCA). The drawbacks of TIOSA's method is only test the functionality of the operating system and hypervisor also have not tested on a specific level of functionality such application and data on the new system, so TIOSA does not suit the evaluation of interoperability in actual system, especially in the industrial field.

The purpose of this study is to introduce a new method that test interoperability at the level of application and data running on virtual machine system, between two different cloud providers. In this study will also be conducted simulation interoperability testing using the method proposed. The benefits of this research is to give a reference about interoperability testing method between cloud systems that more specific to the application and data, so it can be implemented in the field of industry according to business process.

II. INTEROPERABILITY AND CLOUD COMPUTING TEORY

Internet has become a thing that cannot be separated from all aspects of human life. Utilization of various Internet technologies e.g. [7] that used Internet approach as new marketing strategy mix for business development with E-business, also [8] that designed Home Health Internet of Things as a novel health monitoring for the elderly people that stay at home due to several circumstances.

The latest trend in the field of information technology is cloud computing that migrates physical computing resource e.g. server and data storage to huge data centers [9] through internet.

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There are several characteristics of cloud computing [9] such us data, applications or other services that provided by cloud computing can be accessed easily by using internet with a simple browser also high level IT abilities are not required in case of implementation or maintenance. Due to the term of high availability, multiple sites usage can ensure the reliability and scalability of cloud computing service. The other characteristics are easier maintenance because the application does not need to be installed on each user's computer and the payment is based on the facilities were used, also the system performance can be monitored easily. The last but not least is security. In cloud computing environment, security can be either equal with traditional system or better.

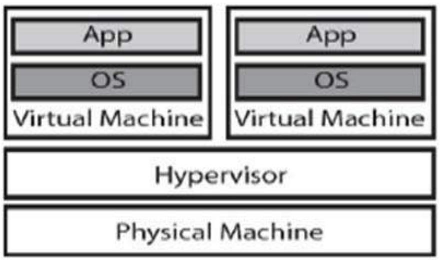

Fig. 1. Virtualization Principle[4]

Virtualization technology is the main key that enabling the emergence of cloud computing paradigm. Fig. 1 illustrates several virtual machines can be run simultaneously on one physical machine. Virtual machine migration is one of the virtualization capabilities allowing system to be transferred transparently from one physical host to another without losing system features [4]. In general, migration processes were done by moving or transfer the application along with the entire virtual machine system including CPU, memory, disk from the source system to the destination system.

Since virtual machine migration utilized for managing applications also its resources in the data center and cloud virtualized systems, it also enables system environment to relocate dynamically to other systems in a faster execution and more reliable way [2].

DMTF defines three types of virtual machine migration operation [2]. First, level 1 migration. Virtual machine system that will migrate must be ensured in the same product or architecture along with the destination system because this type of migration only runs on a particular virtualization product. The cycle is 'suspend' state in source system followed by 'reboot' state in the destination system. Live migration mechanism is possible to run on level 1.

Second, level 2 migration that support different hypervisors migration. The cycle is 'shut-down' state in source system followed by 'reboot' state in the target system.

The most flexible level for the virtual machine migration is level 3 that can be run on multiple families of virtual hardware but this type of migration still impossible due to several reasons.

Interoperability becomes one of critical factors. Cloud interoperability makes easily m*ig*ration and integration process of applications and data between cloud service providers[5]. There are several attributes of interoperability between systems classified by LISI (Levels of Information System Interoperability) [5] :

1) Procedure

Several standard such us application development regulation, product development guidance, hardware and software standards could impact the interoperability level.

2) Applications

The critical aspect for application attribute is how well application running on different systems and platforms in accordance with its functionality.

3) Infrastructure

Infrastructure attribute shows the level of connectivity between systems and applications as well as how systems interact with other applications.

4) Data

This attribute means ability and flexibility of data format that run across systems environment became the main concern of interoperability.

TIOSA (Virtual Machine Interoperability Testing at an OS and Application Level) [1] is a method to measure system interoperability between two different hypervisors. TIOSA method using greybox-based testing method focusing on applications and operating systems verification. TIOSA testing method [1] included:

1) Structured and Repeated Process Models

a) Interoperability Check

Determine the possibility of migration between two hypervisors by collecting metadata required for the migration process.

b) Virtual Machine copy process of two hypervisor in a private cloud

This process ensures the virtual machine converted successfully and can be run on hypervisor target in accordance. The results of all tests that have been carried out is still within the limits of tolerable state

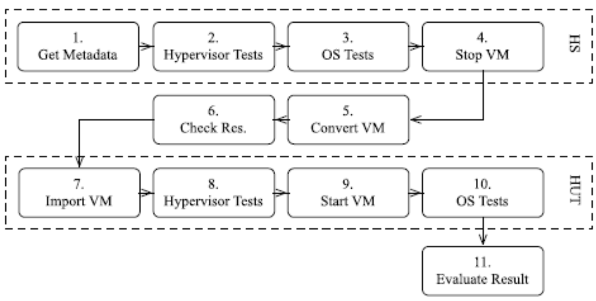

Fig. 2. TIOSA Method[1]

2) *Test result measurement*

The test results are grouped in several categories:

a) *SUCCESS*

Success status can be given if after migration process there is no changes in the term of several hypervisors and operating system parameters in target systems. All of the hypervisors, virtual machines, and operating systems run perfectly in new systems.

b) *WARNING*

The condition if after migration process there is a change but not too intrusive compared with the main functionality.

c) *FAILURE*

There are significant changes and not appropriate with the functionality of the system prior to the migration process.

3) *Metrics comparative evaluation*

There is a comparative evaluation metrics to determine indicators of the overall results of testing.

Implementation of TIOSA testing method uses combination of several cloud infrastructure products for virtualization e.g. VMWare, Citrix Xen, KVM and Microsoft Hyper-V. The total combined migration is 12 combinations for each operating system. The operating systems used in the implementation are CentOS 6.2, Ubuntu 12.0.4, and Microsoft Windows 2008 R2 64-bit. Figure 3 depicts the final result of the migration process with 36 combinations.

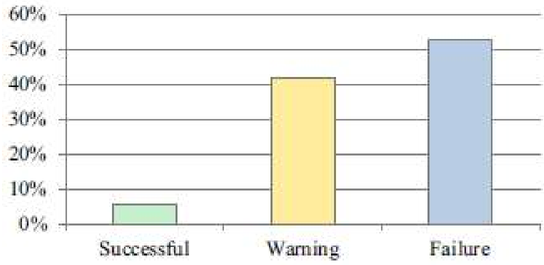

Fig. 3. Final Result of TIOSA Testing

The drawbacks of TIOSA testing method are only testing interoperability at the level of the hypervisor and operating system, and have not been tested at the level of certain applications and data running on the system, so that TIOSA method cannot be implemented optimally for interoperability testing on the system that are in the industry.

III. Proposed Method

The method proposed in this paper is the further development method of TIOSA that will add additional stages on the process of virtual machine migration in the cloud system. Application and data testing in both system also migration impact evaluation stage are being added in proposed method indicated by the numbers 2, 12, and 13.

Figure 4 depicts the proposed new interoperability method testing of virtual machine migration.

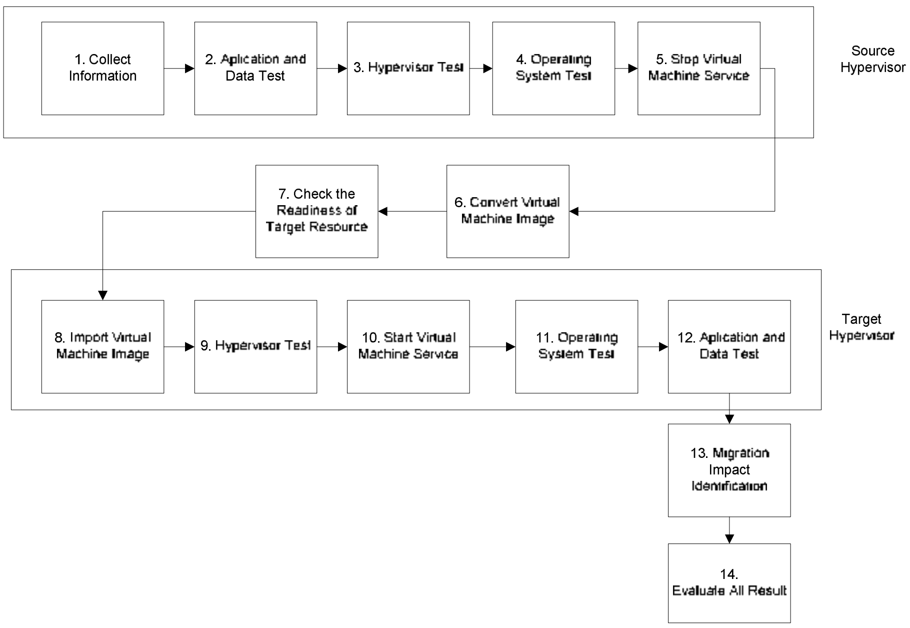

Fig. 4. Proposed New Method

The process of collecting information relating to the possibility of migration between the hypervisor source and target such as the availability of virtual machine conversion application that supports the two systems to migrate and conversion application checking process that will be used whether support OVS (Open Virtualization Format) or not will become the early stage.

After collecting information between two hypervisors, the next stage is testing the application functionality and data consistency in accordance with the business processes running in the system. Hypervisor parameter test e.g. network management and storage in source hypervisor will be the next stage. Operating system test will perform by testing several attributes like the version of the kernel, network connectivity, user management as well as other parameters including other information resources.

Once all the above stages done, virtual machine service will be stopped and start to converting image using tool that available depend on hypervisor product used. Citrix Xen tool converter is one of the tools provided by Citrix that aims to convert the image in case of migration. Checking resource stage aims is to ensure readiness in the target hypervisor environment to start migration process. Virtual machine image import process can be done in an automated way using tools that are connected to the network between two hypervisor or by manually copying the image directly to target hypervisor. Hypervisor test followed by operating system test will performed after the convert process done. The last stage in target hypervisor is application functionality and data consistency test.

Migration impact identification process will measure several parameters in case of decreasing or increasing system performance after migration process. Final evaluation will be done by measuring and determining the test scores results of all the test data that have been obtained. Furthermore, the entire test results will be classified according to the level of interoperobility level.

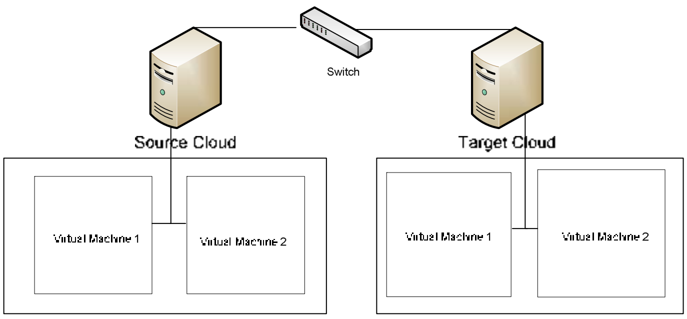

Fig. 5. Proposed Testing Environment

Figure 5 depicts the testing system design of the proposed method. Source cloud and target cloud environment are simulated using separate physical server that connected to network by switch.

Hypervisor in each cloud environment manages two virtual machines that contain operating systems, applications, and databases. In the migration process virtual machine image from source cloud will be converted using appropriate convert tool depend on their hypervisor type. The images will be transferred to target cloud through local network.

IV. Discussion and Conclusion

Based on the studies that have been done, there are several comparisons between TIOSA and the proposed method specified on the Table 1.

TABLE I. COMPARISON OF TIOSA METHOD AND PROPOSED METHOD

No	TIOSA Method	Proposed Method
1	Does not tested at the level of application functionality	Will test at the level of application functionality
2	Does not tested at the level of data consistency in database	Will test at the level of data consistency in database
3	Does not be able to identify the impact of the migration process on the system	Will able to identify the impact of the migration process on the system
4	Does not approach the interoperability of migration system in the industry	Approach the interoperability of migration system in the industry

Data collection process will be conducted based on several test set for each testing components i.e. hypervisor test set, operating system test set, application functionality set test, data consistency test set also performance impact test set. Those tests are performed repeatedly before and after migration process to obtain data accuracy. Overall result will be achieved by classifying all of aforementioned test above to determine the level of interoperability.

In this paper we proposed further development method for interoperability testing between the two systems that migrate on different hypervisors. The method we propose has some benefits from the TIOSA's method because it test at the level of functional applications and data consistency that running on the system. Moreover, the proposed method can identify the performance impact caused by the migration process. This method can be used by organizations that want to test the interoperability between cloud systems in their system.

References

[1] L. Alexander, K.Gregory, M. Michael, S. Ryan, R. Jannis, C. Enrique, and V P. Gopan , "TIOSA: Testing VM Interoperability at an OS and Application Level," *IEEE International Conference on Cloud Engineering.*, pp. 245-252, December 2014.

[2] Open Data Center Alliance(ODCA), "Implementing the Open Data Center Alliance Virtual Machine Interoperability Usage Model". Available : http://www.opendatacenteralliance.org

[3] Lewis, Grace A, "Role of Standards in Cloud-Computing Interoperability," *IEEE 46th Hawaii International Conference on System Sciences.*, pp. 1652-1661, January 2013.

[4] S. Anja, and D. Waltenegus "Does Live Migration of Virtual Machines cost Energy?", *2013 IEEE 27th International Conference on Advanced Information Networking and Applications,* pp. 514-521, March 2013.

[5] D. Scoot, B. Albert, B.M. James, and S. Man-Talk "Cloud to Cloud Interoperability,".*2011 6th International Conference on System of Systems Engineering,* pp. 258-263, June 2011

[6] vSphere 5 Documentation Center, "Introduction to VMware vSphere Virtual Machines,". Available : https://www. pubs.vmware.co

[7] D. Jovevski, Tijan. E, and P. Karanikić, "Internet Marketing Strategies and ICT as a common ground for business development," *MIPRO, 2010 Proceedings of the 33rd International Convention,* pp. 1120-1125, May 2010.

[8] P.P.Ray, "Home Health Hub Internet of Things (H3IoT): An architectural framework for monitoring health of elderly people," *2014 International Conference on Science Engineering and Management Research (ICSEMR),* pp. 1-3, November 2014.

[9] J.Yashpalsinh, and M. Kirit, "Cloud Computing - Concepts, Architecture and Challenges," *2012 International Conference on Computing, Electronics and Electrical Technologies [ICCEET],* pp. 877-880, March 2012.

p-Value Based Cooperative Multiband Spectrum Sensing for Cognitive Radio

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***Abstract*—In this paper, we study an implementation of the Benjamini-Hochberg's procedure for cooperative multiband spectrum sensing in cognitive radio. We propose to use equal gain combining on forwarded p-values from cooperative cognitive radio (CR) users as the test statistic at the fusion center. By doing so, the distribution of the test statistic under the null hypothesis at the fusion center does not vary with different number of samples and types of local detectors. Simulation results show that the familywise miss detection rate (FWM) significantly decreases as the number of cooperative CR users increases. This means that the interference to a primary user due to the decision errors in spectrum sensing significantly decreases. Thus, p-value based cooperative multiband spectrum sensing using the Benjamini-Hochberg's procedure is well motivated.**

***Index Terms*—Cognitive radio, energy detector, cooperative multiband spectrum sensing, multiple testing procedure, Benjamini-Hochberg's procedure.**

I. Introduction

Measurement campaigns conducted all over the world have shown that the spectrum bands for wireless communications are mostly underutilized [1]. This is due to inefficiency and inflexibility of the command and control as the current model to manage spectrum usages. Therefore, it is necessary to find an alternative spectrum management system which can provide more efficient and flexible spectrum usage and hence to maximize the utilization. This step is important since the future wireless communication systems need to support variety of data which demands very high spectrum bandwidths. In this case, the use of dynamic spectrum access (DSA) in the future is well motivated to replace the command and control model used today [2].

DSA supported by cognitive radio technology is a promising technique to maximize spectrum utilization. It is known to provide an efficient and flexible spectrum usage since it allows unlicensed users (CR users) to opportunistically use available spectrum bands (spectrum holes) which are temporarily unused by incumbent users who hold the licenses (primary users). In DSA, spectrum sensing needs to be done by a CR user to detect the existence of spectrum holes. Spectrum sensing can be conducted on a single band, namely single band spectrum sensing, or several bands at once, namely multiband spectrum sensing. Literature on single band spectrum sensing is abundant. However, multiband spectrum sensing is relatively less studied than single band spectrum sensing [3]. To mention a few, multiband spectrum sensing can be found in [4]–[6].

In most literature above, multiband spectrum sensing has not used multiple testing procedure (MTP) as a method to jointly detect the multiple spectrum bands. The use of MTPs in multiband spectrum sensing is to provide control over decision errors not only per band but also for the overall system. In [7], we propose to use Benjamini-Hochberg's procedure in which energy detection is used as the test statistic. In this paper, we extend the work in [7] to cooperative spectrum sensing scheme [3], where several CR users working cooperatively to find unused spectrum bands in a primary network. The aim is to improve the performance of multiband spectrum sensing in terms of the familywise miss detection (FWM) in low SNRs which cannot be achieved by local spectrum sensing (with a single CR user).

II. System Model

Suppose that the primary network using a multicarrier-based system which operates over a wide frequency bandwidth. The bandwidth is divided into K nonoverlapping subbands and a primary user can be assigned to use K_p subbands simultaneously, where $1 \leq K_p \leq K$. A cognitive radio (CR) user observes signal $\mathbf{x}_k[n]$ of the subband k, $k = 1, 2, \dots, K$ which are complex Gaussian distributed $\mathcal{CN}(0, \sigma_{ik}^2)$. Here, we assume that the observations are identically independent distributed (i.i.d.) within subbands and accross subbands. Thus, the spectrum sensing problem of the subband k can be written as

$$\begin{aligned} &\mathcal{H}_{0,k} : \mathbf{x}_k[n] \sim \mathcal{CN}(0, \sigma_{0k}^2) \\ &\mathcal{H}_{1,k} : \mathbf{x}_k[n] \sim \mathcal{CN}(0, \sigma_{1k}^2), \quad k = 1, \dots, K, \end{aligned} \tag{1}$$

where $\mathcal{H}_{0,k}$ and $\mathcal{H}_{1,k}$ denote the null hypothesis (subband unoccupied) and the alternative (subband occupied) for subband k, respectively.

Within a particular time interval, K_0 out of K subbands might not be used by the primary users. We assume that the CR network supports some CR users to simultaneously use several unoccupied subbands. The number of subbands K_c, $1 \leq K_c \leq K_0$, assigned to a specific CR user is, say, based on priority and currently active CR users. The aim is to accomodate simultaneous use of multiple subbands by a primary user or a CR user. Thus, an overall view of spectrum sensing performance is necessary. In this case, false alarm and miss detection are calculated not only per subband, but for $1 \leq K_c \leq K_0$ and $1 \leq K_p \leq (K - K_0)$. Note that

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the existing unoccupied subbands K_0 could span from 0 (all subbands occupied) to K (all subbands unoccupied).

Here, we give an illustration. Suppose that each subband is individually tested with nominal probabilities of false alarm $\alpha_k = \alpha$ and miss detection $\beta_k = \beta,\ k = 1, \ldots, K$. Let us assume that there are $K_0 = 5$ unoccupied subbands and two active CR users, one with high priority (can use $1 \leq K_c \leq 4$ subbands) and the other with low priority ($K_c = 1$). The low priority user could have a subband to use, while the high priority user might defer to simultaneously use all four unoccupied subbands and thus uses lower K_c. This is due to the higher false alarm rate caused by the multiplicity effect [9]. With the same argument, any primary user that uses a higher number of subbands K_p simultaneously will experience a higher aggregate interference level due to a higher probability of miss detection. In this case, a procedure that can solve the multiplicity effect should be implemented to jointly detect the subbands and hence to provide control on the decision errors at the system level. This will be elaborated in the next section.

III. MULTIPLE HYPOTHESES TESTING

If one does not take the multiplicity into account, then the probability that some of the true null hypotheses are rejected by chance alone may be large. For example, suppose that $K = 16$ subbands in (1). Thus, we have 16 hypotheses being tested at the same time, with the size of each test (false alarm) exactly equal to $\alpha = 0.1$. If all tests are mutually independent and K_0 out of 16 being null hypotheses (unoccupied subbands), then the probability that at least one true null hypothesis will be rejected is given by $1-(1-\alpha)^{K_0}$. Therefore, when unoccupied subbands span from $K_0 = 1$ to $K_0 = 16$, the actual probability of false alarm for the overall K_0 subbands spans from 0.1 to $1 - (1 - 0.1)^{16} \approx 0.81$.

If we want to provide control on the decision errors at the system level for spectrum sensing problem of multicarrier-based primary and secondary networks such as (1), then we have to take the multiplicity effect into account by using multiple hypotheses testing[1]. Multiple hypotheses testing refers to the testing of more than one hypothesis at a time [10]. The aim is to solve the multiplicity effect by making the individual tests more conservative to arrive at rejecting $\mathcal{H}_{k,0}$ hypothesis. It is often called *multiple comparison procedure* (MCP) or *multiple test procedure* (MTP) [11]. We prefer to use MTP in sequel. We begin with the definition of some measures and procedures in MTP, then we detail the implementation of MTPs for cooperative multiband spectrum sensing in the following section.

The descriptions of performance measures and some procedures in MTP can be found in [7]. However, we repeat here for convenience. The so-called *familywise error rate* (FWE)

[1]Controlling the decision errors at the system level here has a meaning of having the probability of at least rejecting one true null hypothesis is below α regardless the number of true null hypotheses K_0, and of having the probability of at least rejecting one true alternative hypothesis is below β regardless the number of true alternative hypotheses. They later will be called as familywise error rate (FWE) and familywise miss detection rate, respectively.

TABLE I: Number of correct and false decisions for testing K subbands

	Declared $\mathcal{H}_0$	**Declared $\mathcal{H}_1$**	**Total**
True $\mathcal{H}_0$	U	V	K_0
True $\mathcal{H}_1$	T	S	$K - K_0$
	$K - R$	R	K

is defined as the probability of committing any type I error or false alarm in families of comparisons. Referring to Table I, it is formally written as

$$\text{FWE} = P_0(V \geq 1), \tag{2}$$

where $P_0(\cdot)$ represents the probability of an event under $\mathcal{H}_0$, which could be complete null hypotheses (all $H_{0,k}$ are true) or partial null hypotheses (some subsets of nulls, say $\mathcal{H}_{0,j1}, \ldots, \mathcal{H}_{0,jk}$, are true). An MTP is said to control the FWE in the weak sense if FWE $\leq \alpha$ only under complete null hypotheses and in the strong sense if FWE $\leq \alpha$ under partial null hypotheses, regardless of which subsets of null hypotheses is true [9], including complete null hypotheses.

Here, we define *familywise miss detection* (FWM). It refers to the probability of committing any type II error or miss detection in families of comparisons, formally

$$\text{FWM} = P_1(T \geq 1), \tag{3}$$

where $P_1(\cdot)$ represents the probability of an event under $\mathcal{H}_1$ which could be under complete or partial alternative hypotheses. Note that a false discovery rate, defined as

$$\text{FDR} = E\left[V/R\right], \tag{4}$$

is an alternative controlling procedure which now is commonly used in the MTP [8]. This is because FDR controlling tests are more powerful than those controlling the FWE (for a nominal value of false alarm, it has a better probability of detection).

The simplest way of conducting an MTP is to follow the simple Bonferroni procedure [9]. More precisely, we test each subband individually at the level $\alpha_k = \alpha/K,\ \forall k \in 1, \ldots, K$. Thus, it will guarantee to fulfill FWE $\leq \alpha$. However, this approach is too conservative in protecting decision errors and hence it results in a small power of test (probability of detection). It means that the secondary network makes higher interferences to the primary network. To increase the power of test, a stepwise procedure can be used. For example is the Holm's sequentially rejective procedure [12].

The Holm's sequentially rejective algorithm is based on the ordered p-values, $p_{(1)} \leq p_{(2)} \leq \cdots \leq p_{(k)}$, corresponding to hypotheses $\mathcal{H}_{(1),0}, \mathcal{H}_{(2),0}, \ldots, \mathcal{H}_{(K),0}$. The intuitive reasoning is as follows: once $\mathcal{H}_{(1),0}$ has been rejected using the Benferroni critical value α/k, we should believe that $\mathcal{H}_{(1),0}$ is false. Thus, there are only $k-1$ hypotheses which might still be true, implying the critical value $\alpha/(k-1)$ for $\mathcal{H}_{(2),0}$. If $\mathcal{H}_{(2),0}$ is rejected, use $\alpha/(k-2)$ for $\mathcal{H}_{(3),0}$ [9].

The Bonferroni and the Holm's procedures are based on cotrolling the FWE which is usually required in a strong sense. Thus, it tends to have a small power of test. The work of

Benjamini and Hochberg in [8] suggests a new point of view on the problem of multiplicity. The Benjamini-Hochberg's procedure is based on controlling the FDR which implies controlling the FWER in the weak sense[2]. Note that any procedure that controls the FWER also controls the FDR. Hence, if a procedure controls the FDR only, it can be less stringent, and a gain in power of test may be expected [8].

IV. Cooperative Multiband Spectrum Sensing

A local multiband spectrum sensing based on energy detector using MTP has been proposed in [7]. However, the resulting FWMs of Benjamini- Hochberg and Holm's procedures are still considerably large. When the SNR is smaller than -4 dB, the FWMs are larger than 0.1. Thus, we aim to improve the performance by using a cooperative spectrum sensing (CSS) scheme. In particular, it is to reduce the FWM while the resulting FWE is still in an acceptable level.

In this section, a CSS scheme using MTP is introduced. Each CR user is equipped with a local sensor using energy detector and forwards the measured K number of p-values from K subbands to the fusion center (FC). At the FC, the global decisions regarding the occupancies of K subbands are made.

A. Local Detector

Consider a CR network with L cooperative users. To model CSS, the equation (1) can be rewritten to include the subscript $l, l = 1, 2, \dots, L$ which represents the l-th CR user that joints the CSS,

$$\begin{aligned}\mathcal{H}_{0,l,k} &: \mathbf{x}_{l,k}[n] \sim \mathcal{CN}(0, \sigma_{0,l,k}^2)\\ \mathcal{H}_{1,l,k} &: \mathbf{x}_{l,k}[n] \sim \mathcal{CN}(0, \sigma_{1,l,k}^2),\\ & l = 1, \dots, L,\ k = 1, \dots, K.\end{aligned} \tag{5}$$

For the signal model (5), the likelihood ratio test at the CR user leads to energy detector as the optimum test statistic [13], i.e.,

$$E_{l,k} = \sum_{n=1}^{N} |\mathbf{x}_{l,k}[n]|^2,\ l = 1, \dots, L,\ k = 1, \dots, K \tag{6}$$

where $E_{l,k}$ denotes the received signal energy at the l-th CR user for the subband k, and N is the number of sample. The received signal energy $E_{l,k}$ follows Gamma distributions under the null hypothesis $\mathcal{H}_{0,l,k}$ and the alternative $\mathcal{H}_{l,k,1}$. However, for the large number of sample, $N \to \infty$, the distributions are approximately Gaussian under the two hypotheses (please refer to [7] for the details). In this case, the p-value for the energy detector at the l-th CR user for the subband k can be found from

$$\begin{aligned}p_{l,k} &= P_0(E_{l,k} > E')\\ &\approx \mathcal{Q}\left(\frac{E' - N\sigma_{0,l,k}^2}{\sqrt{N}\sigma_{0,l,k}^2}\right),\ l = 1, \dots, L,\ k = 1, \dots, K\end{aligned} \tag{7}$$

where $\mathcal{Q}(\cdot)$ is the Q-function of the standard normal distribution.

B. Cooperative Spectrum Sensing

Here, we assume that the CSS using a parallel configuration where the CR users do not communicate with each other and that there is no feedback from the FC to any CR user [14]. The reporting channel between each CR user and the FC is assumed to be perfect (there is no error due to the channel for each data transmitted by the CR user).

Suppose that the l-th CR user receives the signal and calculates the p-values $p_{l,k}\ k = 1, \dots, K$ according to (7) based on the measured energy (6). The results are then forwarded to the FC. The test statistic for the subband k at the FC $p_{FC,k}$ is then calculated based on the weighed summation of the forwarded p-values

$$T_{FC,k} = \sum_{l=1}^{L} w_{l,k} p_{l,k},\ k = 1, \dots, K, \tag{8}$$

where $w_{l,k}$ is the weight corresponding to the l-th CR user for the subband k. To simplify the analysis, we assume to use equal weigths $w_{l,k} = 1/L, \forall l \in 1, \dots, L, \forall k \in 1, \dots, K$ throughout this paper.

Note that under the null hypothesis $\mathcal{H}_{0,l,k}$, $p_{l,k}$ has a uniform distribution[3] over the interval (0,1) [15]

$$\mathcal{H}_{0,l,k} : p_{l,k} \sim \mathcal{U}(\mu_{l,k}, \sigma_{l,k}^2),\ l = 1, \dots, L,\ k = 1, \dots, K, \tag{9}$$

with the mean $\mu_{l,k} = 1/2$ and the variance $\sigma_{l,k}^2 = 1/12$. Thus, the test statistic $T_{FC,k}$ is a weighted summation of L uniform distributed random variables with the same means and variances. Using the central limit theorem, as $L \to \infty$,

$$T_{k,FC} = \frac{1}{L}\sum_{l=1}^{L} p_{l,k} \xrightarrow{\mathrm{D}} T_{FC} \sim \mathcal{N}(1/2, 1/(12L)), \tag{10}$$

where $\xrightarrow{\mathrm{D}}$ denotes convergence in distribution To summarize, when the number of CR users that joint the CSS is considerably large the distribution of the test statistics $T_{FC,k},\ k = 1, \dots, K$ are approximately Gaussian with the mean $\mu_{FC} = 1/2$ and the variance $\sigma_{FC}^2 = 1/(12L)$[4].

Suppose that a measured test statistic (10) for the subband k is denoted as T_k. The p-value of the subband k at the FC can be calculated from

$$\begin{aligned}p_{k,FC} &= P_0(T_{k,FC} > T_k)\\ &\approx \mathcal{Q}\left(\frac{T_k - 1/2}{\sqrt{12L}}\right),\ k = 1, \dots, K.\end{aligned} \tag{11}$$

In this paper, cooperative multiband spectrum sensing is done using an MTP where the joint detection for K subbands are implemented based on the p-values $p_{k,FC},\ k = 1, \dots, K$. Considering the advantage of using an MTP that based on

[2]The FDR is equivalent to the FWER when all null hypotheses are true.

[3]In this case, we leave out discrete distributions.

[4]We have confirmed that the real distribution can be sufficiently approximated by the asymptotic distribution for $L \geq 4$

controlling the FDR in terms of the power of test, here we propose to use the Benjamini-Hochberg's procedure [8]. The detail implementation of p-value based cooperative multiband spectrum sensing using the Benjamini-Hochberg's procedure can be found in Algorithm 1, where it guarantees FDR $\leq \alpha$.

Some remarks. In this paper, we choose to combine the p-values of (7) to calculate the test statistic at the FC in (10), instead of combining the resulting measurements of energy such as in [16]. By doing so, for the same number of cooperative users, the distribution of the test statistic $T_{k,FC}$ under the null hypothesis $\mathcal{H}_{0,k}$ will not change though we change the number of samples N in (6) for some or all cooperative CR users. This is because the distribution of the p-value (7) is always uniform regardless the number of samples N[5]. For the same reason, as long as the resulting distributions of the test statistics under the null are all continue, the distribution of $T_{k,FC}$ under $\mathcal{H}_{0,k}$ will not change as well when the CR users use different types of test statistics. Thus, as we can find p-values for local detectors, we can always use (11) to calculate the p-value at the FC regardless the types and the number of samples of the local detectors.

To find weights $w_{l,k},\ \forall l \in \{1,\dots,L\},\ \forall k \in \{1,\dots,K\}$, in (10) that minimize the FWM is an interesting problem. However, due to the complexity, it might be treated in our future works for cooperative multiband spectrum sensing.

V. Numerical Results

In this section, we evaluate the performance of cooperative multiband spectrum sensing by simulation. For all simulations, we set the nominal values of FDR to $\alpha = 0.1$ as a controlling value for the Benjamini-Hochberg's procedure. In Fig. 1, we use the Holm's procedure for comparison where we set the controlling value FWE to 0.1. In all simulations, primay user signals in occupied subbands received by all CR users are assumed to have the same SNRs. The sample number is fixed to $N = 100$ for Fig. 1 and 2, while for Fig. 3 and 4 it varies depending on the number of cooperative CR users. The number of subbands K varies from 2 to 16 and the number of CR users L ranges from 1 to 64. Here, we assume that the probability of each subband occupied by a primary user is equal, i.e., $P(\mathcal{H}_{1,k}) = P_1, k = 1, 2, \dots, K$. We study the cases where the subbands are considered to be busy $P_1 = 0.8$ and to be sparesely occupied $P_1 = 0.2$.

Here, the aim of using cooperative spectrum sensing is to reduce the FWM of Benjamini-Hochberg's procedure. We have indicated in [7] that if we only use local spectrum sensing the FWM is still considerably large, particularly for a low SNR region. Fig. 1 shows that the increase of cooperative CR users from 1 to 16 reduces the FWM as large as 85% for the Benjamini-Hochberg's procedure and as large as 60% for the Holm's procedure when the number of subbands $K = 16$ at SNR$= -10$ dB. In addition, the figure indicates that the use of Benjamini-Hochberg's procedure is still more attractive than that of Holm's procedure for cooperative multiband spectrum sensing, since the former has smaller FWMs than the later, particularly for large number of subbands K and CR users L. For detail comparisons between the two procedures implemented for multiband spectrum sensing, please refer to [7].

Fig. 2 shows that the minimum SNR to have FWM$\leq \beta$ shifts to a smaller value, for a larger number of cooperative CR users. Suppose that we aim to have $\beta = 0.1$[6] It can be achieved at SNR > -14 dB, -12, -10 dB, -8 dB and -4 dB for the number of cooperative CR users 64, 32, 16, 4 and 1, respectively. In general, protection against interference to the primary network due to the detection errors in spectrum sensing increases as we increase the number of cooperative CR users at the CR network.

To know the effect of the number of CR users and the number of samples to the performance of cooperative multiband spectrum sensing, we proceed with an experiment that evaluates the FWM and the FWE as functions of SNRs where

Algorithm 1 Cooperative Multiband Spectrum Sensing using Benjamini-Hochberg's Procedure

Step 0) INIT: N, L, K, α

AT THE l-th CR USER, $l = 1, \dots, L$:

Step 1) observes signals at subbands $k = 1, 2, \dots, K$,

Step 2) calculates the energy of the signal for each subband $k,\ k = 1, \dots, K$, using (6)

Step 3) computes p-values of all K subbands using (7)

Step 4) forwards the resulting p-values to the fusion center (FC)

AT THE FC:

Step 5) receives the K number of p-values from each CR user.

Step 6) for each subband $k,\ k = 1, \dots, K$, combines all the p-values from the L CR users to get the test statistic $T_{k,FC}$ of (10) and then calculates its p-value $p_{k,FC}$ of (11)

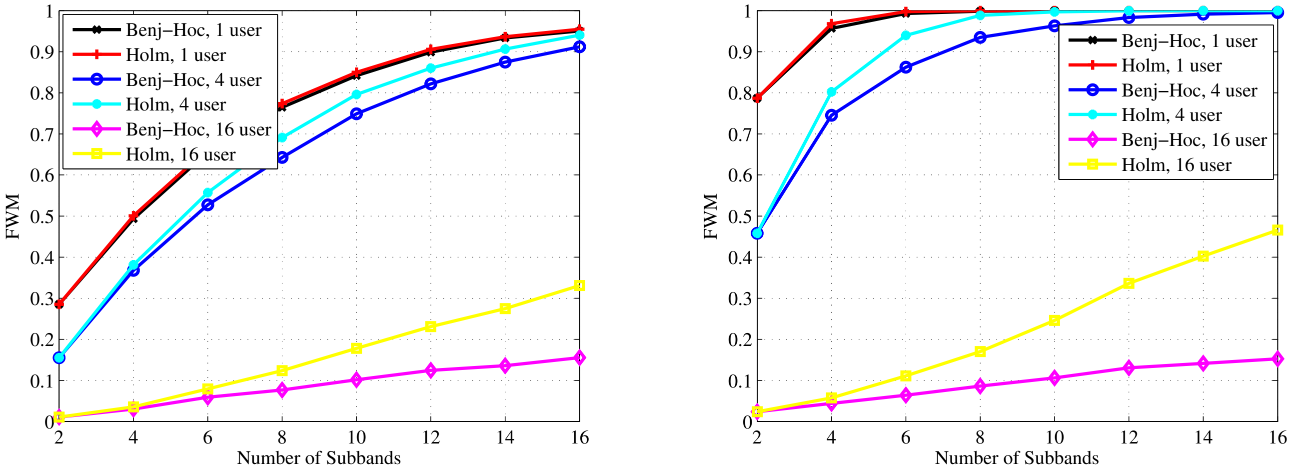
Step 7) Rank the p-values in ascending order, $p_{(1),FC} \leq p_{(2),FC} \leq p_{(3),FC} \leq \cdots \leq p_{(K),FC}$ for the corresponding hypotheses $\mathcal{H}_{0,(1)}, \mathcal{H}_{0,(2)}, \dots, \mathcal{H}_{0,(K)}$.

Step 5) Calculate $k_{max} = \max\{1 \leq k \leq K :\ p_{(k),FC} \leq \dfrac{(k)\alpha}{K}\}$.

Step 6) If such k_{max} exists, reject the null hypotheses $\mathcal{H}_{0,(1)}, \mathcal{H}_{0,(2)}, \dots, \mathcal{H}_{0,(k_{max})}$ (the corresponding subbands are declared occupied). Otherwise, reject nothing (all subbands are unoccupied).

[5]For the same measured energy E', the values of the p-value might be different for different number of samples N, but the distributions are the same.

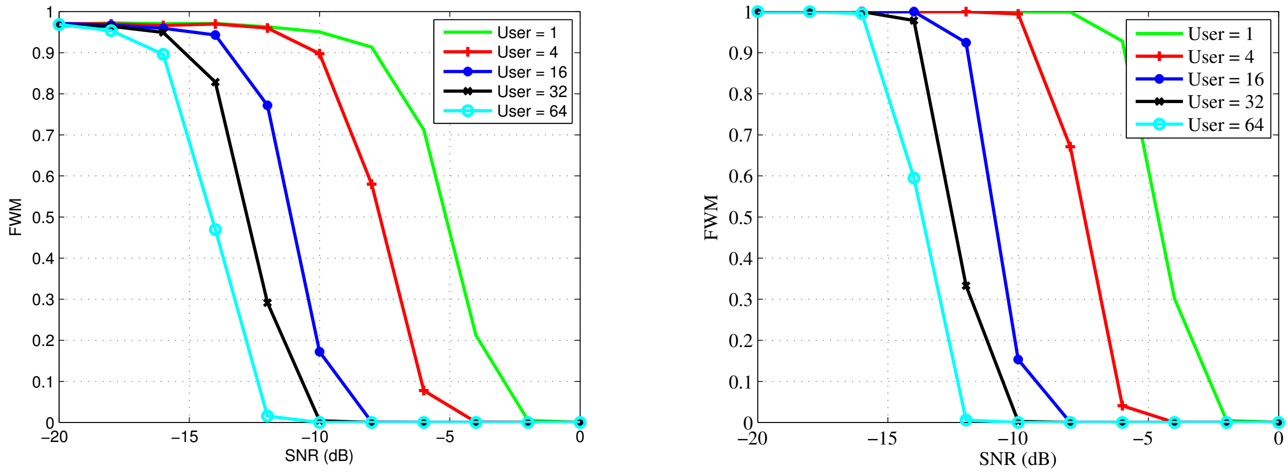
[6]It means that the probability of making interference to a primary user is not larger than 0.1 regardless the number of subbands being occupied by the primary user.



(a) Sparse ($P_1 = 0.2$)

(b) Busy ($P_1 = 0.8$)

Fig. 1: Familywise miss detection (FWM) vs. Number of subbands, using the Holm's procedure (Holm) and the Benjamini-Hochberg's procedure (Benj-Hoch) for the number of CR users varies from 1 to 16 and SNR= -10 dB.



(a) Sparse ($P_1 = 0.2$)

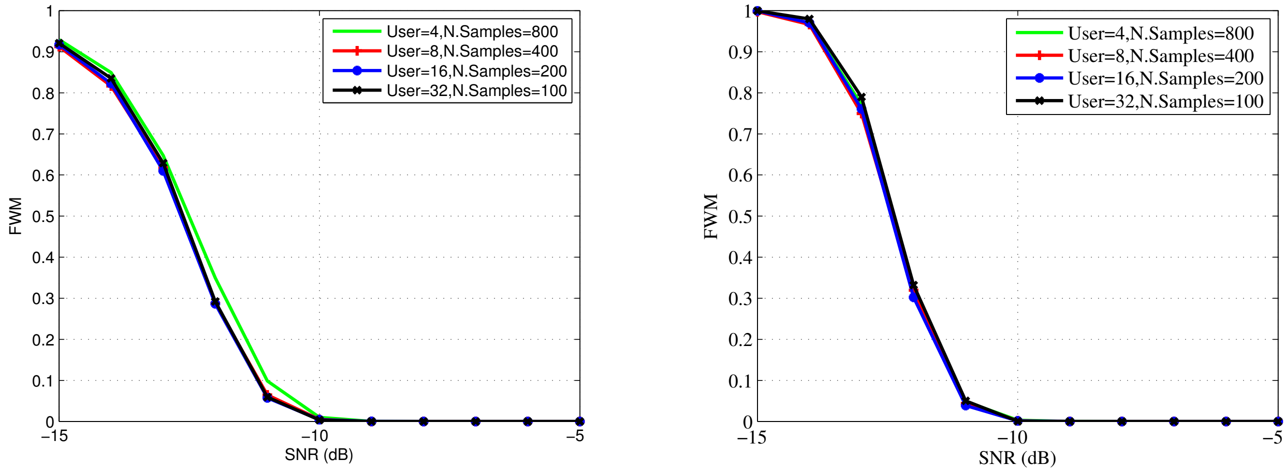
(b) Busy ($P_1 = 0.8$)

Fig. 2: Familywise miss detection (FWM) vs. SNR, using the Benjamini-Hochberg's procedure, for different number of cooperative CR users.

the numbers of CR users L and samples N vary. However, to be fair, the total number of samples, i.e. $N_T = L \times N$, is fixed. Here, we consider four cases: (Number of CR users, Number of samples)= $(4, 800), (8, 100), (16, 800)$ and $(32, 100)$. Thus, we fix $N_T = 3200$. Fig. 3 shows that the differences in FWM for the four cases in busy subbands are unnoticable. In this case, the FWMs are approximately the same as long as the total number of samples N_T is fixed. However, for sparse subbands at a low SNR region, the figure shows that there is a small advantage if we increase the number of cooperative CR users instead of the number of samples. For the FWE, the advantage of using a larger number of cooperative users instead of a larger number of samples is more pronounced. The reduction ranges from 1% to 5% as shown in Fig. 4. Note that the FWE cannot be made smaller than 0.1, since the Benjamini-Hochberg's procedure is based on controlling the FDR to get a larger power of test (consequently a smaller FWM).

VI. Conclusion

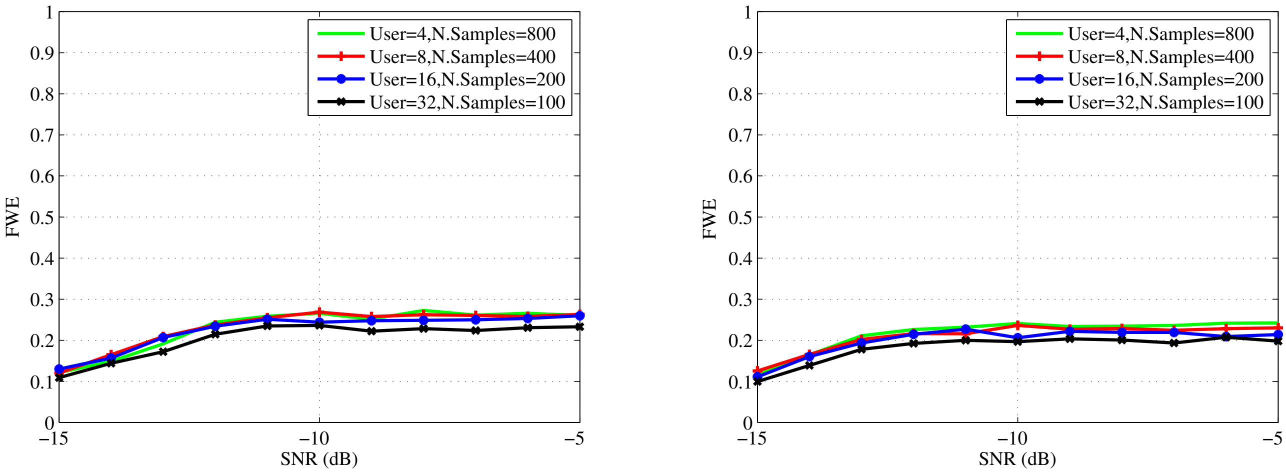
A p-value based cooperative multiband spectrum sensing using the Benjamini-Hochberg procedure is presented in this paper. The use of a number of cooperative CR users for spectrum sensing has improved the performance in terms of the FWM, particularly for a low SNR region. Thus, the



(a) Sparse ($P_1 = 0.2$)

(b) Busy ($P_1 = 0.8$)

Fig. 3: Familywise miss detection (FWM) vs. SNR, in four cases: (Number of CR users, Number of samples)= $(4, 800), (8, 100), (16, 800)$ and $(32, 100)$, and the number of subbands $K = 16$.



(a) Sparse ($P_1 = 0.2$)

(b) Busy ($P_1 = 0.8$)

Fig. 4: Familywise error Rate (FWE) vs. SNR, in four cases: (Number of CR users, Number of samples)= $(4, 800), (8, 100), (16, 800)$ and $(32, 100)$, and the number of subbands $K = 16$.

interferences to the primary network due to the decision errors at the CR network decreases, regardless the number of occupied subband of a primary user. In this paper, we remark the advantage of using p-value based cooperative multiband spectrum sensing, which is having an independence of using different types of local test statistics and different number of samples without having to recalculate the distribution of the test statistic under null hypothesis at the fusion center.

Acknowledgment

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References

[1] M.A. McHenry and D. McClosky, *Spectrum occupancy measurements, Chicago, Illinois*, Shared Spectrum Company, 2005.

[2] E. Hossain, D. Niyato, and Z. Han, *Dynamic Spectrum Access and Management in Cognitive Radio Networks*, Cambridge University Press, 2009.

[3] E. Axell, G. Leus, E.G. Larsson, and H.V. Poor, "Spectrum sensing for cognitive radio: State-of-the-art and recent advances," *IEEE Signal Processing Magazine*, vol. 29, no. 3, pp. 101–116, 2012.

[4] W.Y. Lee and I.F. Akyildiz, "Optimal spectrum sensing framework for cognitive radio networks," *IEEE Transactions on Wireless Communications*, vol. 7, no. 10, pp. 3845–3857, 2008.

[5] Zhi Quan, Shuguang Cui, Ali H Sayed, and H Vincent Poor, "Optimal multiband joint detection for spectrum sensing in cognitive radio networks," *Signal Processing, IEEE Transactions on*, vol. 57, no. 3, pp. 1128–1140, 2009.

[6] S.H. Wu, C.Y. Yang, and D.H.T. Huang, "Cooperative sensing of wideband cognitive radio: a multiple-hypothesis-testing approach," *IEEE Transactions on Vehicular Technology*, vol. 59, no. 4, pp. 1835–1846, 2010.

[7] F.Y. Suratman, G. Budiman, and I. Hidayat, "Multiple testing procedure based on energy detector for multiband spectrum sensing in cognitive radio," *IEEE Asia Pasific Conference on Wireless and Mobile*, Aug. 2015 (Accepted).

[8] Y. Benjamini and Y. Hochberg, "Controlling the false discovery rate: a practical and powerful approach to multiple testing," *Journal of the Royal Statistical Society. Series B (Methodological)*, pp. 289–300, 1995.

[9] P.H. Westfall and S.S. Young, *Resampling-based multiple testing: Examples and methods for p-value adjustment*, vol. 279, Wiley. com, 1993.

[10] J.P. Shaffer, "Multiple hypothesis testing," *Annual review of psychology*, vol. 46, no. 1, pp. 561–584, 1995.

[11] S. Dudoit and M.J.V.D. Laan, *Multiple testing procedures with applications to genomics*, Springer, 2008.

[12] S. Holm, "A simple sequentially rejective multiple test procedure," *Scandinavian journal of statistics*, pp. 65–70, 1979.

[13] S.M. Kay, *Fundamentals of statistical signal processing, Volume II: Detection theory*, vol. 7, Upper Saddle River (New Jersey), 1998.

[14] Ramanarayanan Viswanathan and Pramod K Varshney, "Distributed detection with multiple sensors i. fundamentals," *Proceedings of the IEEE*, vol. 85, no. 1, pp. 54–63, 1997.

[15] Bradley Efron, *Large-scale inference: empirical Bayes methods for estimation, testing, and prediction*, vol. 1, Cambridge University Press, 2010.

[16] J. Ma, G. Zhao, and Y. Li, "Soft combination and detection for cooperative spectrum sensing in cognitive radio networks," *Wireless Communications, IEEE Transactions on*, vol. 7, no. 11, pp. 4502–4507, 2008.

[17] G. Atia, E.B. Ermis, and V. Saligrama, "Robust energy efficient cooperative spectrum sensing in cognitive radios," in *14th Workshop on Statistical Signal Processing (SSP)*. IEEE, 2007, pp. 502–506.

[18] A.C. Davison and D.V. Hinkley, *Bootstrap methods and their application*, vol. 1, Cambridge university press, 1997.

[19] S.J. Shellhammer, "Spectrum sensing in IEEE 802.22," *IAPR Wksp. Cognitive Info. Processing*, pp. 9–10, 2008.

DESIGN OF GEOGRAPHIC INFORMATION SYSTEM FOR TRACKING AND ROUTING USING DIJKSTRA ALGORITHM FOR PUBLIC TRANSPORTATION

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***Abstract*— The ability to perform vehicle tracking is very useful in daily life - today, such as security on personal vehicles, public transportation systems, fleet management and other mass transit. In this study, we designed an application of Geographic Information Systems (GIS) that will be applied to public transport services. This system provides real-time position tracking of each vehicle (taxi) and capable of providing a taxi with the best category for passengers using Dijkstra's algorithm. This system can be helpful, especially for passengers who are often too long waiting for a taxi that had been booked and also helps the driver to save fuel consumption because the taxi drivers do not have to drive across town to get the passengers and are just waiting for confirmation sent by this system.**

Keywords—Dijkstra, GPS, GIS, Tracking, Routing.

I. INTRODUCTION

Global growth in the number of vehicles is expected to increase along with economic growth and the number of middle class society is increasing, such as in China and India. Although the growth in the number of vehicles in addition to the positive impact that the increasing number of middle and upper class residents, but the development of a tracking system (tracking) of a vehicle which is also a very important part in a transport system is still lacking. The use of tracking systems on the mode of transport will be very useful for a variety of aspects including safety and economy in private vehicles, mass transport systems, commercial vehicle, and so forth.

On public transport services, especially taxis, often encountered a problem where the passengers have to wait very long to get a taxi that had been booked through the service provider. This generally occurs at rush hour where traffic jams often occur so taxis were booked sometimes be stuck in traffic so that the lower the level of service a taxi operator. Another problem that often occurs in the driver taxi in which a taxi driver had to drive around far enough to get the passengers, causing increased fuel costs.

The main objective of our research is to build a system capable of providing the service automatically and instantly to the passenger taxi. By using this system the position and velocity of each taxi will be monitored through the application of GIS (Geographic Information System) web-based. To be able to serve a taxi booking automatically, then the system is implemented Dijkstra's algorithm that serves to find a taxi that is most suitable for passengers based on the parameters within the prospective passengers of the taxi and the traffic levels nearby occurs so that passengers will get a taxi that had been ordered in the not so long. Another benefit to be gained is a taxi driver does not have to drive around to different places to get the passengers because the system will immediately send location information so that passengers actually increase fuel efficiency and improve services for passengers.

II. STATE OF THE ART

A. Automatic Vehicle Location (AVL)

Automatic vehicle location system is a computer-based vehicle tracking vehicles [6]. The current position of each vehicle will be monitored and will be forwarded to the control center.

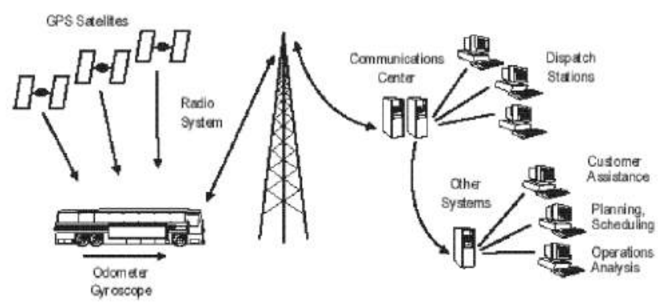

Fig 1. Working principle AVL.

Generally, the vehicle position information will be stored on the vehicle for which can last for several seconds. Position information can be forwarded to the control center in the form of raw data or after processing by the device on the vehicle before it is transmitted.

Agent or transportation company has implemented AVL systems to help them in many ways. By implementing this system, the many benefits that can be taken as from the operational side, the system will help meet the schedule accuracy, improve the efficiency of the services, facilities integrated system and reduce the number of supervisors (supervisors) in the field. The benefits that can be taken in terms of communication and security is a radio system that has replaced the old, reducing voice communication via mobile data terminals and improve response time to incidents and emergencies.

B. Global Positioning System

GPS is a system that serves to determine the location on the earth's surface with the aid of alignment by satellites [10]. GPS uses satellite data to calculate the accuracy of the position of

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the earth. All GPS works the same way but they look very different and have different software. Very significant differences between the various GPS receivers is the number of satellites that can simultaneously communicate with the receiver. Most described the 12-channel receiver, which means the device is able to communicate with 12 satellites. GPS older models only able 8 or 5 channels, but with the latest models of receivers capable of communicating with 14-20 satellites.

Number, position and signal strength of satellites allows the GPS to calculate the error rate. This error rate can be a good guide to know how accurate the readings done by GPS and an error rate that can be tolerated from a GPS device is under 10 m (ideally below 5 m).

GPS data recording is generally the same in all units. GPS receivers automatically record the data into its memory based on time elapsed or the distance of the movement that occurs. It is generally termed as trackpoints. GPS receivers can often be used as a complete navigation tool, not only offers driving directions and the location in detail but also offers a navigation tool that can help users when it will move from one place to another. Most receivers have been equipped with digital compass. Digital compass work based on data from satellites and this is not a compass made of magnet and will only work when the user starts to move.

C. Geographic Information System

Geographic information systems (GIS) originally evolved from two independent disciplines, namely: digital mapping and database. The development of digital cartography as a result of growing world of design, especially CAD (Computer Aided Design) since the 1960s. Similarly, the development of the use of data base management systems, especially database or Database Management Systems (DBMS) that enables the integration of spatial data and non-spatial contribute to accelerate the development of GIS. In a further development of GIS involves various disciplines such as remote sensing, photogrammetry and surveying.

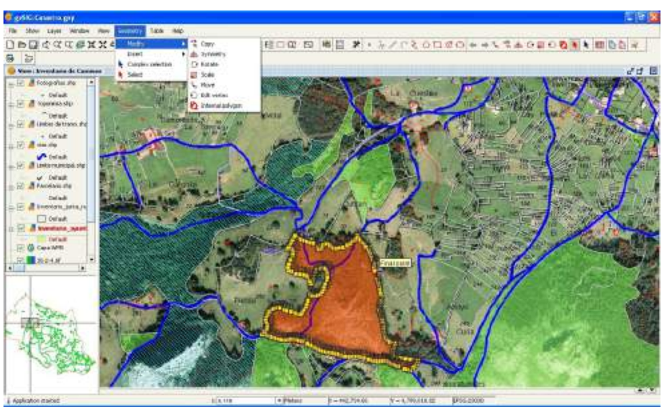

Fig 2. Example of GIS.

Today the use of GIS has been highly integrated with daily life. Geospatial technologies that work behind the scenes has helped either directly or indirectly projects and government programs. The map feature is currently available in almost every information system. Ranging from information services (service) to address the search needs, something very difficult thing happened in the era of the 1980s in Indonesia.

D. Dijkstra Algorithm

Dijkstra's algorithm invented by Edsger Dijkstra in 1959, is a graph search algorithm that solves the shortest path problem that originates from one node to a graph with node weights can not be negative [7]. The analysis was performed by examining the node with the smallest weight and put it into the set of solutions to the initial search origin node requires knowledge of all the paths and weight, so it is necessary exchange of information with all the vertices. Dijkstra algorithm has a simple nature and plates (straightforward), according to the working principle greedy. Constituent elements of the greedy algorithm is:

1. The set of candidates, C

   This set contains elements that have a chance to form a solution. On the issue of the shortest path in the graph, this candidate set is the set of vertices in the graph.

2. The set of solutions, S

   This set contains the solution of the problems were resolved and the elements consist of elements in the set of candidates but not all of them, or in other words, the solution set is upabagian from the set of candidates.

3. The selection function

   The selection function is a function that will elect any candidate which allows to produce the optimum solution for every step.

4. The feasibility function

   The feasibility function will check whether a candidate has been elected violate the constraint or not. If a candidate violates the constraint then the candidate will not be incorporated into the solution set.

5. The objective function

   The objective function will maximize or minimize the value of the solution. The goal is to choose one course the best solution from each member of the set of solutions.

Figure 3.10. below is an example of an undirected graph consisting of 5 dots and 7 pieces of track connecting between two points. Djikstra algorithm is used to find the shortest distance from a point to another point on the undirected graph.

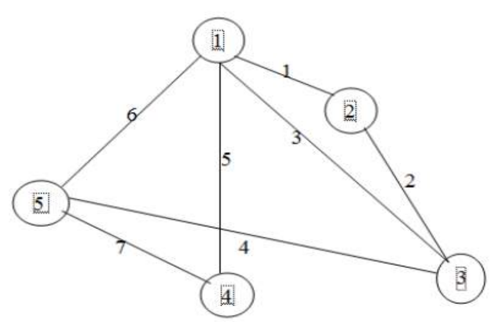

Fig 3. The example of undirectional graph

Based on the example of undirected graph above specified search starting point is the point 1 with the goal of 4 points and will look for the shortest distance that can be taken from point 1 to get to the point 4. The following is an explanation table graph using the Dijkstra's algorithm:

Path	Initial Path					Point	l(i,j)				
	1	2	3	4	5						
	0	0	0	0	0		1	2	3	4	5
1	1	0	0	0	0	**1**	∞	∞	∞	∞	∞
1-2	1	1	0	0	0	**2**	1	∞	∞	∞	∞
2-3	0	1	1	0	0	**3**	3	2	∞	∞	∞
3-5	0	0	1	0	1	**4**	5	∞	∞	∞	7
5-4	0	0	0	1	1	**5**	6	∞	4	∞	∞

TABLE I. Explanation of the graph using Dijsktra algorithm.

Completion Dijkstra path algorithm point 1 to point 4 has been completed as described above and the table according to the calculation graph search algorithm dijkstra appropriate procedural steps. In the first line of all successor in the set of 0 means to give value to a point source route to be used for all routes and ketidakterbatasaan another point, stating the fact that it is not known any trajectory.

For further because of point 1 as the source track then it is definitely elected. So it turned into a status set 0. 1. Point 1 will be check points directly neighboring ie points 2, 3, 4 and 5. From there dijkstra will choose who has the lowest weighting towards the next point. Elected point 2 with a weight of 1, sets the status of 0 is changed to 1 and so on. Then search the shortest distance of the above, obtained shortest path based search dijkstra from point 1 to 4 is via direct point 1 point 4 with the weight trajectory 5.

III. SYSTEM DESIGN

The system to be built consists of the android smartphone-based GPS receivers that are placed on each vehicle called on-board unit (OBU) and a server unit that serve receive position information from each GPS receiver and display it through a web-based service. The system works by monitoring in real time of the position of the vehicle and the level of traffic density.

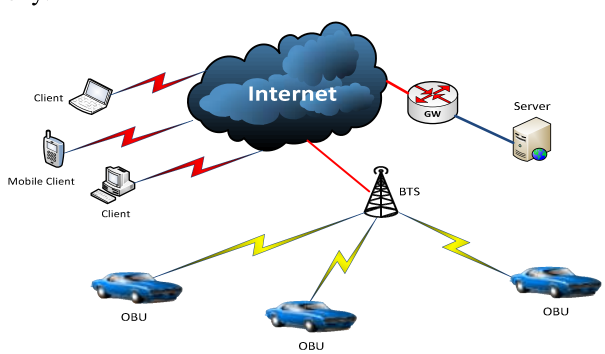

Fig 4. The design of system architecture

GPS receivers used in this system is a smartphone with Android operating system. Data from GPS feature on a smartphone will be sent to the server, using the application B'trace. Besides being able to track the position of a smartphone, this application will send the data of latitude and longitude in real time to a server. Data transmitted by the GPS receiver will be stored in the database of the server used. When passengers access the application pages, the system will show the position of each taxi is available on a web page based on latitude and longitude of the data collected from around the GPS receiver of each taxi.

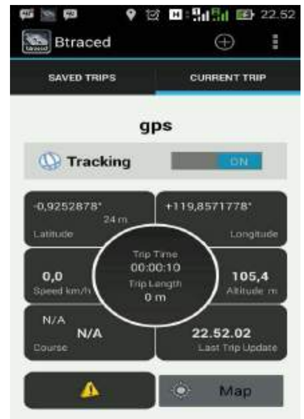

Fig 5. The tracking application

To be able to find the vehicle that best suits the passengers, then the system will use dijkstra's algorithm. Dijkstra's algorithm to be used in this system using the position of the vehicle as well as the density of traffic on urban streets as the parameters to be taken into account to find the most appropriate vehicle for passengers. If the taxi with best categories have been found, then the system will automatically send information about the position of passengers to taxi drivers through a service-based push e-mail.

IV. DISSCUSION AND CONCLUSION

The data will be sent to the server is the data that has been processed first by the GPS receiver before being sent to the server via communication celular services (GSM). Transmitted data is the data latitude and longitude to be used by the server to map the position of each taxi on a google map based map application. Latitude and longitude data that has been sent by the GPS receiver and stored into a database which will be processed by the server to determine the position of the nearest taxi with passengers.

This system will be tested in urban areas within the period that adjusted to the level of traffic density occurs. This system will be tested in two conditions. First in the event of high traffic density is at 07:00 AM - 03:00 PM. Second, when the current traffic conditions back smoothly ie at 05:00 PM - 09:00 PM. In this test, the designed system will be compared with the existing system services to know the service improvement that occurs based on the waiting time of each passenger on a taxi that has been booked.

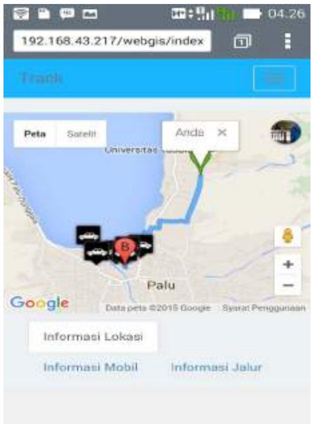

Fig 6. Application user interface

Dijkstra's algorithm is applied to the system will be able to solve the problem of finding the best with the taxi to the criteria based on the parameters of the taxi with the shortest distance to the passenger and traffic levels that occur in an area. These applications provide benefits not only to passengers where passengers are not too long waiting for a taxi he had a message but also provide benefits for taxi drivers where the close proximity of the passengers will save time and fuel consumption.

REFERENCES

[1] Aloquili O., Elbanna A., Al-Azizi, A. (2009), “Automatic Vehicle Location Tracking System Based on GIS Environment”, *The Institution of Engineering and Technology*, Vol. 3, No. 4, pp. 255-263.

[2] Cho H.J., & Cho M., (2013), “Effective Position Tracking Using B-Spline Surface Equation Based on Wireless Sensor Networks and Passive UHF-RFID”, *IEEE Transactions on Instrumentation and Measurement*, Vol. 62, No. 9.

[3] Dat H. P., Drieberg M., Cuong C.N. (2013), “Development of Vehicle Tracking System using GPS and GSM Modem”. *IEEE Conference on Open Systems*, Sarawak.

[4] Davis S. (2007), “GIS for Web Developers”, The Pragmatic Programmers.

[5] Jerath K., & Brennan, N. (2012), “GPS Free Train-Based Vehicle Tracking on Road Networks”, *American Control Confrence*, Faimont Queen Elizabeth, Montreal.

[6] Johnson M.C., & Thomas, L.E. (2000), “Automatic Vehicle Location Successful Transit Aplications”, U.S. Department of Transportations, Washington, DC.

[7] Joyner D., Nguyen, V.H., Cohen, N. (2011), “Algorithmic Graph Theory”.

[8] Lammle T., (2007), “Cisco Certified Network Associate Study Guide”, Wiley Publishing, Inc., Indianapolis.

[9] Lee S., Tewolde G., Kwon J., (2014), “Design and Implementation of Vehicle Tracking System Using GPS/GSM/GPRS Technology and Smartphone Application”, *IEEE World Forum on Internet of Things*.

[10] McWilliam N. (2005), “GIS, GPS and Remote Sensing”, The Expedition Advisory Centre Royal Geographical Society 1 Kensington Gore, London.

[11] Odom W., (2007), “CCENT/CCNA ICND1 Official Exam Certification Guide”, Cisco Press, Indianapolis.

[12] Sathiamoorthy M. (2009), “On GPS Tracking of Mobile Devices”, *International Conference on Networking and Services*”.

[13] Yi Z., Ding W., Yanfei Y. (2009), “On Dynamic Scheduling of Vehicles Based on GPS / GIS / RFID”, *Department of Computer Science & Technology*, Harbin.

[14] Yuan Gannan, Zhang Zhi and Guan Shang Wei. (2008), “Research and Design of GIS in Vehicle Monitoring System”, *Harbin Engineering University*, Harbin.

Designing Gamified-Service Towards User Engagement and Service Quality Improvement

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Abstract--- Organizations find new challenge which is related to employees' engagement along service support implementation. Gamified-service is defined as service support which integrated Information Technology Infrastructure Library (ITIL) and gamification model. This integration enables fun, challenge, and reward to boost employees' motivation. Pressure in the workplace and employees' personal issues become main triggrer to decrease productivity. According to this reason, we need to add persuasive approach along implementation of ITIL best practices. This research focused on designing gamified-service prototype in order to help organization build game-like workplace environment. We believe that gamified-service will help employees do their job with fun way.
Keyword---gamification;ITIL;service support;user engagement

I. INTRODUCTION

Most of organizations in this era implement information technology (IT) to help their activities run more effective and efficient. But, there are always unexpected incidents and problems during the use of IT. It can slow the activities down, so that there will be delay time which will cause disappointment of end users as stakeholder. It is why we need service support to fix the IT incident and problem.

The implementation of service support needs best practice that is able to ensure standard service support is well designed. In this research, we use Information Technology Library (ITIL). But, we think that best practice is not enough because it just handles the mechanism of how to deliver good service support, it can not handle service support employees' motivation. Moreover, it can not ensure the engagement between employees and the system they use. Previous research shows there is a positive correlation between full user involvement and participation with system success [1]. According to this matter, we need other persuasive approach.

Gamification model is the use of game design in non-game context[2]. This model is used as persuasive approach to motivate users of system in order to reach the system owner's goal. By using gamification, we can increase productivity of service support employees.

Naturally, human like challenges, games appeal to the players because there are challenges and uncertainties. Scientists who research about human's brain around the world agree that challenge- achievement- reward loop promotes production of dophamine. It can create satisfaction and reinforce desire to play [3].

This research focuses on the integration of ITIL and gamification model. The representation of the integration is a gamified-service. We believe that full engagement of employees will increase the productivity and service quality.

II. LITERATURE REVIEW

A. ITIL

ITIL is document that presents best practices to manage the implementation of IT service. ITIL explains the detail of management and IT operations such as incident management, problem management, change management, configuration management, and availability management. The latest version of ITIL is ITIL 3.0 which has been published in 2007 by Office of Government (OGC). It contains 5 phases of service lifecycle.

- Service Strategy
- Service Design
- Service Transition
- Service Operation;
- Continual Service Improvement

B. Service Support

The service support first presents in previous version of ITIL, version 2.0. It covers main cores of ITIL processes which are responsible to support the IT services. In the latest ITIL, it will be represent as service operation. Service support covers the concepts below :

- Service desk
  To run ITIL processes, we need a function. Service desk is a function to provide communication between users and IT employees. Service desk roles and responsibilities are senior service desk manager, service

desk manager, service desk supervisor, and service desk analyst [4]. Service desk primary tasks are providing solution of IT infrastructure incident, request, problem,etc. Moreover, service desk is a single and first line which communicate with users in order to information about status of incident, request, problem, etc.

- Incident management
  Process defines as set of structured activities and tasks that produce a specific goal [5]. ITIL processes have been specified. Such as incident management, it has been specialized to manage incident. Incident Management is often the first process when introducing the ITIL framework to a Service Desk, and offers the most immediate and highly cost reduction and quality gains [6].
  - Problem management
    This process contains activities to diagnose the root cause of incident. It also responsible to maintain problem information and resolution. So that organization is able to minimize incident.
  - Configuration management
    Process configuration management is a system that integrate all tools and provides critical data on improvement opportunities [7]. It eliminates a lot of manual works and produce better stability, predictability, and maintainability.
  - Change management
    Process change management is defined as set activities to ensure the standard procedure are used to control changes.
  - Release management
    The release management contains planning, designing, building, configuration, and testing hardware and software releases to create a set of release components.

C. *Gamification*

Many organizations have implemented gamification model in order to motivate their employees while doing their tasks. Last four years, gamification has been discussed. Many applications have been using gamification, like Nike+ Running, GoJek, and Foursquare. Gamification helps the organization to gain costumer ‘s satisfaction. It is not a user interface design, it is about how to reach the goal with fun way. Gamification enables win – win solution between organization and employee [8]. Organization can reach their goal, and employees are able to get the internal satisfaction, like full engaged system.

Based on research, job satisfaction, high loyalty, and productivity of employees are stimulated by good internal service and organization policy [9]. It means issues which are related to employees satisfaction is critical in helping organizations reach their goals and business such as satisfying their costumers.

D. *User Engagement Measurement*

User engagement is defined as the quality of the user experience that emphasises the positive aspects of the interaction. It is not just about usability but it also contains how users invest time, attention, and emotion when they are connected to the system [10]. We can identify each attributes of user engagement degree by using user engagement measurement. According to the previous research, user engagement attributes are focused attention, aesthetic, novelty, percieved usability, endurability, felt involvement[11].

III. DESIGNING GAMIFIED-SERVICE

Gamified- service is web-based service system that gamifying the processes which are mentioned in ITIL. In this paper we will introduce the prototype of gamified – service which runs function service desk and process incident management. Based on ITIL, there are 3 main roles of service desk, service desk analyst (SDA), service desk supervisor (SDS), service desk manager (SDM). First, SDA acts as the first level of service desk. It responsible to provide quick solution for reported incidents, entry all details about incident, do basic analyze, and deliver solution from upper level to costumer if the incident report is escalated. Second, SDS as second level. Beside ensuring the skill levels of staffs are mantained and representing SDAs at meetings, SDS becomes the escalation point when SDA finds difficulties or controversial calls. Third, SDM as escalation point for SDS. SDM provides solution for incidents which can not be handled by first and second level. Moreover, he responsibles to do advance analyze of business impact which caused by incidents and report it to the board. Fig 1 shows the use case diagram for service desk roles.

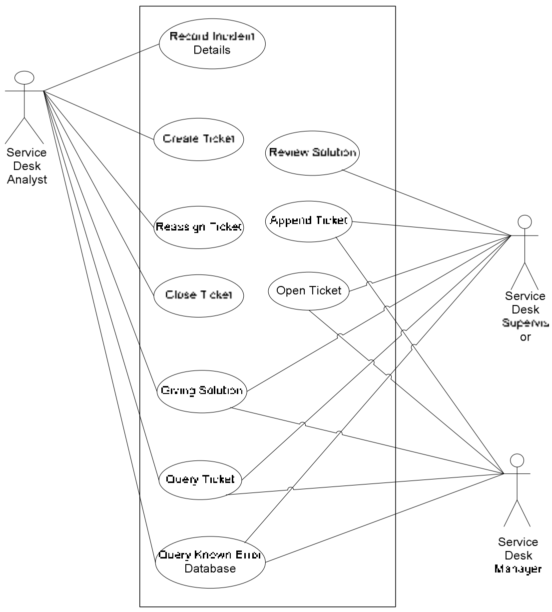

Fig.1 Use Case Diagram of Service Desk Roles

There are seven main features of gamified- service prototype, namely, (1) record costumer, (2) record incident, (3) solution database (4) known error database (KEDB), (5) escalation, (6) close incident, (7) leaderboard.

Fig.2. Gamified- Service Home Page

First, costumer will report the incident to SDA and he will create a new data costumer if costumer does not exist in database.

Second, when the record of costumer is already saved, SDA records details of incident such as, category of incident, description of incident, time and date of incident, level of urgency, etc.

Third, SDA checks solution database to find solution if the solution can not be found in solution database, he will provide solution based on his skills and knowledge. This activity is also the mechanism of input solution. When solution of incident is first time recorded to database, it is not directly inserted to solution database. It needs supervisor to review the solution. Once it is accepted, it will be recorded as solution

Fourth, incidents which are reported more than once will be recorded on KEDB. This features will help organization to find the weakness of IT Service and enable the opportunity to evaluation and improvement. In this feature, we will see incident records, their solutions, and how many times they reported.

Fifth, SDA can use solution database to deliver the solution and his own skills and knowledge, if costumer is satisfied, SDA is able to close the incident. In contrary, if SDA finds the costumer doubts, it means incident solution is failed, SDA will create incident ticket and forward it to upper level, SDS and SDM. SDS and SDM are responsible to open the escalated ticket. SDS escalated incident ticket to SDM, and if SDM can not fix incident, SDM will forward the incident ticket to problem management, which means the incident ticket is failed. Moreover, incident ticket will fail if the incident ticket is not open, and it causes SDA loses RP. But, the gamified-service does not include the punishment for irresponsible SDS and SDM. This matters will be charged to upper level of organization.

Sixth, when solution for incident is delivered, SDA close the incident management process. If the incident management is closed before time ends, SDA will get points. and in contrary if it is closed after time has ended, SDA will lose points. In this system, points are categorized by two types, redeemable points, and experience point (XP). When service has failed, SDA will lose the RP. Service will fail if the incident ticket is not opened and solution is not delivered to costumer until until the time ends. In contrary, when SDA delivers the solution to costumer on scheduled time, RP and XP will gain. RP can be exchanged to desirable contents, such as bonus, shopping voucher, etc. SDA can lose RP, but his XP will keep gaining. Every actions in gamified- service is considered and the experience keep gaining. It becomes the reason for giving the XP even the incident ticket has failed. Leaderboard shows the top 10 SDA with highest XP and RP.

TABLE 1 URGENCY LEVEL AND TIME TO SOLVE

Urgency level	**Estimated time to solve incident**
1 = Low	12– 23 hours 59 minutes 59 seconds
2 = Middle	6 – 11 hours 59 minutes 59 seconds
3 = High	0 – 5 hours 59 minutes 59 seconds

Seventh, leaderboard shows SDAs with highest XP and RP. It helps organization to create competition environment in workplace. There are many game elements that can be used, in this prototype we use points as core element. Points are able to be converted to other game elements, when SDA has points, then organization can add other elements, such as avatar, badges, exclusive contents, virtual contents, etc. These elements depend on points, system needs points to upgrade badges, so does when system need to give exclusive contents to SDA.

Along the implementation of gamified-service, we need to identify what are indicators of success. It also helps organizations get information about how far they have gone. Tabel 2 shows the basic evaluation indicators for service support [12]. We suggest a gap analysis between pre-implementation of gamified-service and post implementation. After do a gap analysis, organizations are able to find what should they do to improve their services.

TABLE 2 BASIC EVALUATION INDICATORS FOR SERVICE SUPPORT

Indicators	**Explanation**
Response time	Response user's calls as soon as possible
Call rate	Rate of request for help (RFH) that comes from service desk

Records of request call	Numbers of the events and insidents recorded by service desk
Rate of complains	Assessing the service desk whether handling incident
Accuracy of incident classification	Assessing the service desk whether handling accidents by the defined categories
Growth rate of desk insident	The rate that incidents could not be processed
Rate of incident handling	Remote processing rate when service desk recieves the RFH
First call completion rate	Connected rate when user first calls

IV CONCLUSION

Service support employees must be fully engaged to the system because they responsible to communicate with users in order to help users fix incidents, problem, etc. If they can not engaged to the system, then it will cause slow response to the request, event-handling, incident, etc. Engagement deals with employees' motivation, involvement, and participation. So, we propose design of gamified- service which integrate ITIL best practices and gamification model. By using gamification in service support, the engagement will be increased and may return benefits to organization such as internal satisfaction, costumer trust, etc. This design can be used to do some future research in gamified- service, like implementing mobile gamified-service, gamification system for costumer, and finding what are the most influence user engagement attributes for gamified service.

REFERENCES

[1] M.I.Hwang and R.G.Thorn, "The effect of user engagement on system success: a meta-analytical integration of research findings." *Information and Management Journal*, vol.35, pp.229-336, Apr.1999.

[2] S.Deterding, D. Dixon, R. Khaled, L. Nacke, "From Game Design Elements to Gamefulness: Defining "Gamification" ," , 2011.

[3] G. Zichermann and C. Cunningham, *Gamification by design: implementing game mechanics in web and mobile apps*. Sebastopol, Calif.: O'Reilly Media, 2011.

[4] D. Knapp, *A guide to service desk concepts*, 3rd ed. Boston, MA: Course Technology, Cengage Learning, 2010.

[5] T. H. Davenport, *Process innovation: reengineering work through information technology*. Boston, Mass.: Harvard Business School Press, 1993.

[6] Q. Wang, J. Song, L. Liu, X. Luo, and E. Xinhua, "Building IT-based incident management platform," *5th International Conference on Pervasive Computing and Applications*, 2010.

[7] J. Bon, *Foundations of IT service management based on ITIL*, 2nd ed. Holland: Van Haren Publishing, 2005.

[8] A.Narayanan, *Gamification for Employee Engagement*, UK : Impackt Publishing, 2014.

[9] J.K.Heskett, J.O.Jones, et.al., *Putting the Service-Profit Chain to Work,* USA :Harvard Business Review, 1994.

[10] J.Lehmann, M.Lalmas, E. Yom-Tov, G.Dupret,"Model of User Engagement," in *UMAP'12 Proceedings of the 20th international conference on User Modeling, Adaptation, and Personalization*, Montreal, CA, 2012, pp.164-175.

[11] H.L.O'Brien and E.G.Toms,"The Development and Evaluation of a Survey to Measure User Engagement," *Journal of the American Society for Information Science and Technology*, vol.61, pp. 50-69, Jan 2010.

[12] Z. Yao and X. Wang, "An ITIL based ITSM practice: A case study of steel manufacturing enterprise," *2010 7th International Conference on Service Systems and Service Management*, 2010.

A Conceptual Framework for Implementing Gamified-Service to Improve User Engagement by Using ITIL

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Abstract--- Gamified- Service is defined as a service support which used ITIL and gamification model to boost the motivation of service support employees. Along the implementation of service support, organizations find the challenge which is related to human resource who deliver the service. Pressure at workplace and personal issues become trigger to decrease their productivity. According to this fact, standard service support is not enough. We need persuasive approach to guarantee that employees are totally engaged. In other side, to ensure that good service support is implemented, we need best practices of IT Service Management, it is Information Technology Infrastructure Libarary (ITIL). This research is focused on creating a framework to build, implement, and maintain good service support with full engaged employee.

Keyword---gamification;ITIL;service support;user engagement

I. INTRODUCTION

Most of organizations in this era implement information technology (IT) to help their activities run more effective and efficient. But, there are always unexpected incidents and problems during the use of IT. Those unexpected situation can slow the activities down, so that there will be delay time which will cause disappointment of end users as stakeholder. It is why we need service support to fix the IT incident and problem.

The implementation of service support needs best practice that is able to ensure standard service support is well designed. In this research, we use Information Technology Library (ITIL). But, we think that best practice is not enough because it just handles the mechanism of how to deliver good service support, it can not handle the employees of service support's motivation. Moreover, it can not ensure the engagement between employees and the system they use. Base on this finding, we need other persuasive approach.

Gamification model is the use of game design in non-game context[1]. This model is used as persuasive approach to motivate users of system in order to reach the system owner's goal. By using gamification, we can increase productivity of service support employees.

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This research focuses on the integration of ITIL and gamification model. The representation of the integration is a conceptual framework to build, implement, and monitor gamified- service. We believe that full engagement of employees will increase the productivity.

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A. ITIL

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Based on research, job satisfaction, high loyalty, and productivity of employees are stimulated by good internal service and organization policy [8]. It means issues which are related to employees satisfaction is critical in helping organizations reach their goals and business such as satisfying their costumers.

D. User Engagement Measurement

User engagement is defined as the quality of the user experience that emphasises the positive aspects of the interaction. It is not just about usability but it also contains how users invest time, attention, and emotion when they are connected to the system [9].

According to the previous research, user engagement attributes are focused attention, aesthetic, novelty, percieved usability, endurability, felt involvement[10]. By using user engagement scale (UES), we can get the score of each attributes. While, the accumulation of every score is the degree of user engagement. But, it is also available for us to present the result as the score of each attributes in order to simply identify which attribute is being a focus to evaluation.

E. Service Life Cycle Development

In Application Management Document which is one of ITIL product shows SDLC as an approach to manage service application[11]. The steps of SDLC are :

- Feasibility study
  In this first step, we will do a set of practices to ensure that service is feasible to be implemented.
- Analysis
  Implementation of service support needs the analysis about what is the current and desired state of selected process or function, and how do we reach the desired state. Moreover, we have to do some research and observe about stakeholder's needs and system requirements.
- Design
  After we gather the stakeholder's needs and system requirements, we will create desired process business scheme. In this step, it is possible for us to build prototype in order to give better understanding to stakeholders about the system which will be deployed.
- Testing
  The prototype or design that have been created in previous step will be tested. In this step, we will ensure that design is well designed and meet the stakeholder's needs and system requirements.
- Implementation
  In this step , the design which has been created will be transformed to application. After the application is coded, we need to test and if the application has been agreed by stakeholder, we will set up the infrastructures to support the application.
- Evaluate
  After the application has been set up, we need to evaluate the system periodically in order to find bugs and errors.
- Maintenance
  When the bugs and errors have been listed, we need to fix the system. In this step, we must ensure that system run properly. Moreover, along the maintenance, we have opportunities to gather information and monitor the system in order to find a challenge for improvement.

III. FRAMEWORK DESIGN

A. Gamification Framework Design

From literature study [12] [13], we design gamification framework that will be used to develope game mechanincs on service system. Figure 1 shows the presentation of model.

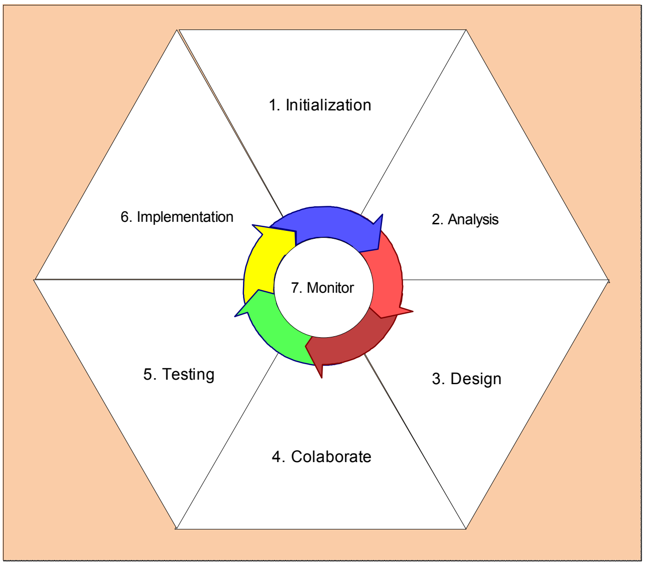

Fig. 1. Gamification framework design

The steps of gamification framework are :

- Initialization
  In this step, we need to know which part of process that must be gamified. Moreover, we need to identify who will join the project and form teams.
- Analysis
  After we identify what process that will be improved or built, we need more detail and deeper analysis about current state of process and what is the desired state, after that we need to identify who are the users of system. Identifying users is needed because we need to choose game elements. Moreover, in this steps we have to discover what kind of technology that will be used.
- Design
  In the analysis step, we have identified who are the users. In this design step we will design game element that will be used. Such as designing badges, virtual contents, etc.
- Colaborate
  After the elements are ready, we will integrate the element to the system and create game mechanics. Game mechanics is about how a game operate, what are applied rules, and how the players interact to the game[14]. In gamified- service perspective, this component defines tasks and actions that user can do to system.
- Testing
  In this step, we will test the the design or prototype. We will present the design and let the system owner to fully check the correctness.
- Implementation
  If the design is accepted and the tests are success. We will continue to implementation step. In this step, the full gamification model will be deployed. After the deployment, the second tests will be held. It includes functional and non-functional tests. Additionally, this step will engage the end users to try the system.
- Monitor
  In this step, we need to ensure that gamification elements and mechanics work properly and meet the agreements.

B. Proposed Conceptual Framework for Building Gamified-Service

The conceptual framework gamified- service is deal with two framework which have been explained and created at previous section. Fig 2. We can see the combination of SDLC and gamification model design

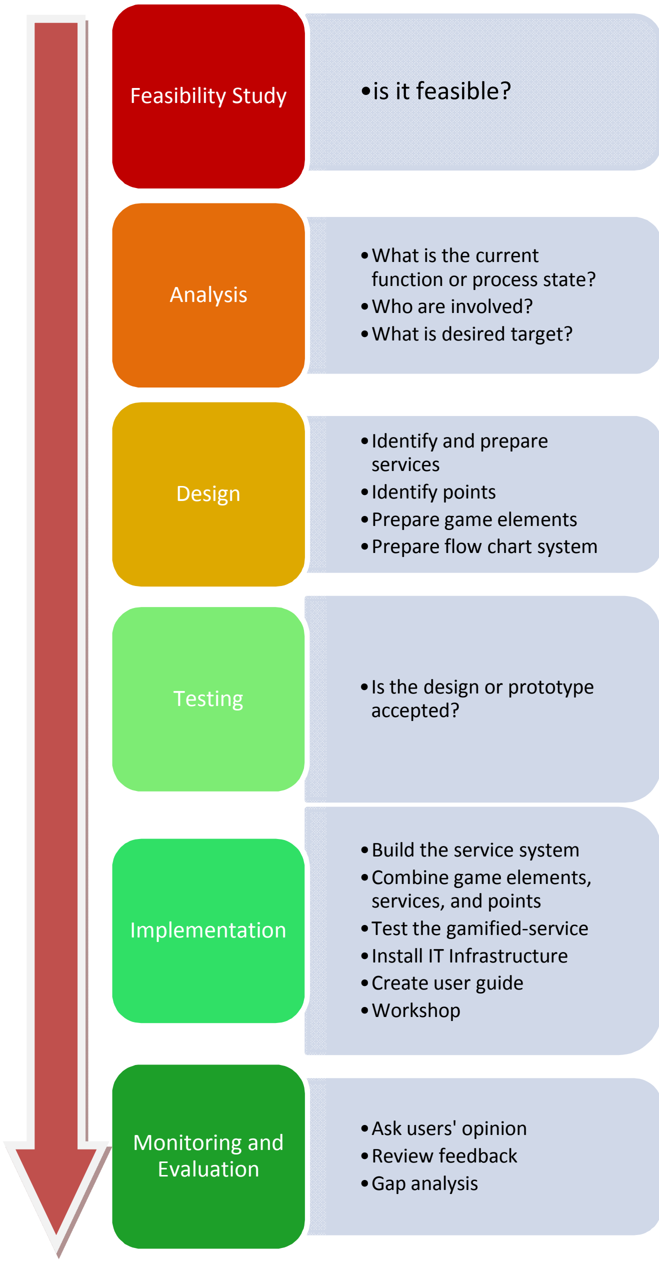

Fig. 2. Conceptual framework for Implementing Gamified-Service

The steps of conceptual framework for gamified-service are:

- Feasibility Study
  This step deals with the study to identify feasibility status of implementation gamified service. We need to identify the benefits of implementing gamified-service.
- Analysis
  If we find that gamified- service is feasible to be implemented. We can continue to analysis step. This step is focused on what is the current state of process or function that will be integrated with gamification model. We need to identify indicators in Table 1to simply assess current state of service support[15] :

TABLE 1 BASIC EVALUATION INDICATORS FOR SERVICE SUPPORT

Indicators	Explanation
Response time	Response user's calls as soon as possible
Call rate	Rate of request for help (RFH) that comes from service desk
Records of request call	Numbers of the events and insidents recorded by service desk
Rate of complains	Assessing the service desk whether handling incident
Accuracy of incident classification	Assessing the service desk whether handling accidents by the defined categories
Growth rate of desk insident	The rate that incidents could not be processed
Rate of incident handling	Remote processing rate when service desk recieves the RFH
First call completion rate	Connected rate when user first calls

  Moreover, we have to identify who are the users and then we identify user engagements, motivation, opinions and stakeholder's needs which are related to their view about the system. After gathering the stakeholder's needs and user's opinions, we need to identify their desired target.
- Design
  In this step, we prepare service and gamification contents. We also need to design rules for every actions. Below are activities of design:
  - Identify and prepare service
    This part deals with identification service features which are based on ITIL concept
  - Identify Points
    In this activity, points are identified. Points are used to get contents and stimulate elements
  - Prepare game elements
    After identify the service and points we will prepare what are game elements that will be integrated to the service system.
  - Create flow chart system
    After preparing all requirements, we need to create a process business of gamified- service. If needed, we can create prototype to deliver the design and gamified- service mechanism to system owner.
- Testing
  The design has been created and we can continue to the testing step. In this step, we need to present our design to system owner to ensure the design meets their requirements.
- Implementation
  The design that has been created and accepted will be transformed to service application. In this step, we will also do activities below :
  - Combine game elements, service, and points
    In this activity, every elements which has been choosen will be combined with rules.
  - Testing gamified- services
    In this activity, we have to complete functional test, non-functional test, user acceptance test (UAT). The goal of functional test is ensuring the gamified- service runs properly without errors. Based on ITIL, we may use diagnostic hooks which provide informations to fix errors. While, non-functional testing is more advance. It will test the degree of security, performance, usability, etc. When functional and non-functional test are done, we may engage the users to run UAT. This final test will direct to approval or rejection.
  - Set up IT infrastructure
    To run gamified- service we need other support technologies such as hardware and network. In this activity, we will engage IT Technical Expert.
  - Create user guide
    User guide is needed to provide guidance to users. It helps users to understand and operate the gamified-service.
  - Giving workshop and training
    This activity is important because it is the best way to help user to operate and understand system.
- Monitor and Evaluate
  We need to monitor system activity to ensure that system run properly. The other activities that we should do in this step are :
  - Asking user's opinion
    In order to gather information about user engagement, we need to interview system users. Their opinions are needed to identify what should be fixed and improved.
  - Review feedback
    After gather the information from users, we can set up solution to fix or improve the system.

- Gap analysis
  First, we need to review the degree of user engagement. We can use user engagement measurement to ensure that we reach one of gamified-service goals. In order to assess the quality of service support, we can do assessment based on table 1. In the analysis step, we have the information about the state before the implementation. To ensure that we meet the goals, we can compare the state before and after implementation.

IV CONCLUSION

Service support employees must be fully engaged to the system because they responsible to communicate with users in order to help users fix incidents, problem, etc. If they can not engaged to the system, then it will cause slow response to the request, event-handling, incident, etc. Engagement deals with employees' motivation, involvement, and participation. So, we propose conceptual framework which integrate ITIL and gamification model. By using gamification in service support, the engagement will be increased and may return benefits to organization such as internal satisfaction, costumer trust, etc. This framework can be used to do some future research in gamified- service, like implementing mobile gamified-service or gamification system for costumers.

REFERENCES

[1] S.Deterding, D. Dixon, R. Khaled, L. Nacke, "From Game Design Elements to Gamefulness: Defining "Gamification" ," , 2011.

[2] G. Zichermann and C. Cunningham, *Gamification by design: implementing game mechanics in web and mobile apps*. Sebastopol, Calif.: O'Reilly Media, 2011.

[3] D. Knapp, *A guide to service desk concepts*, 3rd ed. Boston, MA: Course Technology, Cengage Learning, 2010.

[4] T. H. Davenport, *Process innovation: reengineering work through information technology*. Boston, Mass.: Harvard Business School Press, 1993.

[5] Q. Wang, J. Song, L. Liu, X. Luo, and E. Xinhua, "Building IT-based incident management platform," *5th International Conference on Pervasive Computing and Applications*, 2010.

[6] J. Bon, *Foundations of IT service management based on ITIL*, 2nd ed. Holland: Van Haren Publishing, 2005.

[7] A.Narayanan, *Gamification for Employee Engagement*, UK : Impackt Publishing, 2014.

[8] J.K.Heskett, J.O.Jones, et.al., *Putting the Service-Profit Chain to Work,* USA :Harvard Business Review, 1994.

[9] J.Lehmann, M.Lalmas, E. Yom-Tov, G.Dupret,"Model of User Engagement," in *UMAP'12 Proceedings of the 20th international conference on User Modeling, Adaptation, and Personalization*, Montreal, CA, 2012, pp.164-175.

[10] H.L.O'Brien and E.G.Toms,"The Development and Evaluation of a Survey to Measure User Engagement," *Journal of the American Society for Information Science and Technology*, vol.61, pp. 50-69, Jan 2010.

[11] Office of Government Commerce OGC, Application Management, UK : The Stationary, 2002.

[12] [12] P.Herzigg, M.Ameling, et.al. " *Gamification in Education and Business*. Springer International Publishing, 2015.

[13] S. Harris and K. O'Gorman, *Mastering Gamification : Customer Engagement in 30 days : the Revolutionary way to Attract Customers, Keep Them Coming Back for More, and take Your Business to the Next Level*, UK : Impackt Publishing, 2014.

[14] A.Ernest and A.Rollings, *Fundamentals of Game* Design, USA : Prentice Hall,2006..

[15] Z. Yao and X. Wang, "An ITIL based ITSM practice: A case study of steel manufacturing enterprise," *2010 7th International Conference on Service Systems and Service Management*, 2010.

A Conceptual Framework of Engaged Digital Workplace Diffusion

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***Abstract*---Digital workplace technology creates challenges to improve the way people do their business. By using it, organization is able to collect, process, and provide data faster, share knowledge, enforce their employee to communicate and collaborate with any devices. These can be done by using technologies such as big data, cloud computing, search-based application, and internet connection. But, we argue that technology is not enough to help organization to do its business and reach its goal. We propose the conceptual framework of digital workplace diffusion which are integrated diffusion theory, user's engagement, and controls which are contained IT governance, risk management, compliance. We believe that user engagement and control are one of the keys for successful digital workplace diffusion. User engagement measurement is able to show the degree of employees' understanding and acceptance to the digital workplace. While control is used to help organization in managing, monitoring, ensuring the digital workplace is aligned to the requirements and regulations.**

Keywords--- digital workplace; digital workplace diffusion; user engagement; technologies.

I. INTRODUCTION

In information age, technology changes rapidly. New technologies can change the way people work and it becomes a new challenge to begin improvement.The digital workplace is meant to be a virtual equivalent to the physical workplace[1] where employees can work anywhere by using any devices, share knowledge, and browse data faster. This idea is based on the used of several recent trends, Bring Your Own Devices (BYOD), Internet of Things (IoT), people-centric work methods, and analytics [2] which help employees to collaborate and do tasks and activities effectively.

In this proposed model, we propose a conceptual framework to digital workplace diffusion. Beside adding the technologies and diffusion theory which can help organization to implement digital workplace, we propose user engagement-centric and control. Employees must understand and accept digital workplace to bring a successful digital workplace diffusion.

According to previous research, there is a positive correlation between full user involvement and participation with system success [3]. It becomes the reason for us to consider the focus on user engagement, we believe that the more users engage, the more chance for digital workplace diffusion to be success.

II. LITERATURE REVIEW

In this section, we will describe and discuss about theories which are related to digital workplace diffusion.

A. Digital Workplace

Digital Workplace is a coordination between technology, process and people [1]. Digital workplace enables employees to work effectively from anywhere, at any time, on any device and provides an internet-like participative mode and user experience no matter where their location [4].

Digital Workplaces create employees' ability to do their job by collaborating and communicating and connecting with others [5]. To achieve this goal, we need technologies as support. Digital Workplace is focused on developing an application for mobile environment. Integration of four technologies - mobile, big data, cloud computing and search-based application enables us to achieve the desirable feature of Digital Workplace [6].

B. Diffusion

Technological diffusion is a multi-stage process comprised of adoption, use and widespread incorporation or society [7]. Roger describes diffusion as the process by which an innovation is communicated through certain channel over time among the members of a social system. There are four elements of diffusion. Those are innovation, communication, time, and social system. Innovation is defined as an idea, practice, or object that is perceived as new by an individual or another unit of adoption [8]. Innovation is about figuring out how to add value for an organization. Innovation is the act with idea and successfully bringing them to life to solve problems and create opportunities [9].

To adopt an innovation, we need to introduce that innovation to adopter. The innovation is communicated through a communication channel. It is the means

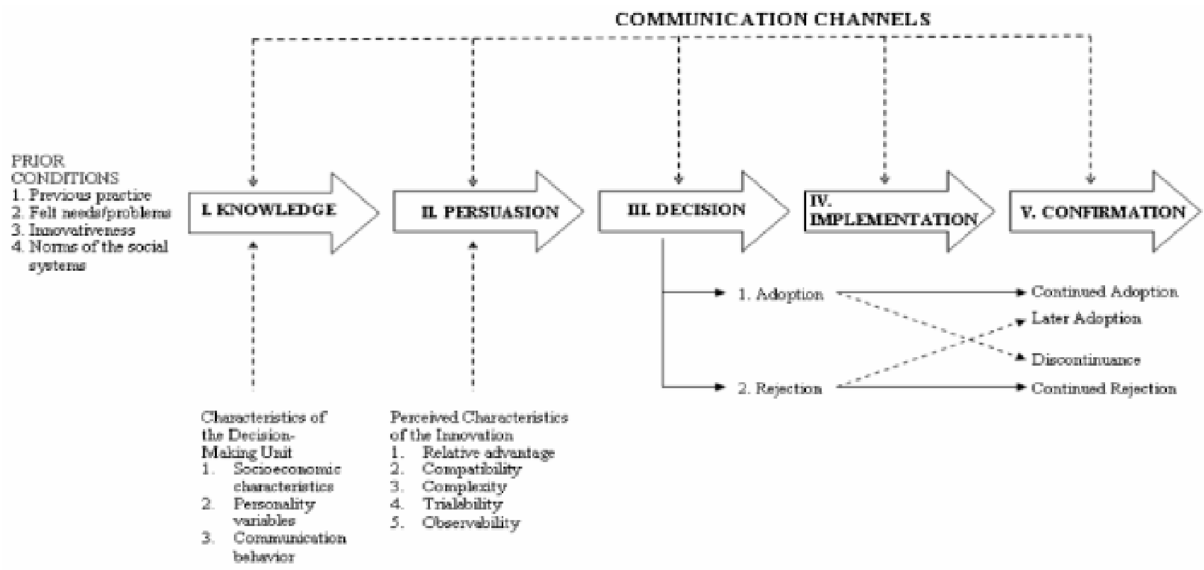

Fig. Innovation-Decision Process [8]

by which messages get from one individual to another in a social system. The members or units of a social system may be individuals, informal groups, organizations, and/or subsystems[8].

Another aspect of communication process is time. We can find time dimension of diffusion in the innovation-decision process. The innovation-decision process is essentially an information-seeking and information-processing activity in which the individual is motivated to reduce uncertainty about the advantages and the disadvantages of the innovation. Fig.1 shows five stages of innovation-decision process[8].

C. *User engagement*

User engagement is defined as the quality of the user experience that emphasizes the positive aspects of the interaction. It is not just about usability, but it also contains how users invest time, attention, and emotion when they are connected to the system [10]. It matters because it is related to the confirmation of adoption status. When a user is engaged, it means they will more understand about what they should do and they will admit the value of the whole system and there will be a chance to the innovation to be well adopted. User engagement has become an issue in the implementation of IT. It is a cognitive aspect which represents of positive emotion like enjoy and willingness [11]. It will suddenly cause the internal satisfaction. According to previous research, user engagement has 6 attributes, focused attention, aesthetic, novelty, perceived usability, endurability, felt involvement [12]. Table 1 shows the list of engagement attribute and description. This attributes will be used to measure user engagement of digital workplace.

TABLE I. DESCRIPTION OF USER ENGAGEMENT ATTRIBUTES

Attributes	**Description**
Focused attention	The state of total absorption in situation, and full attention [13]
Aesthetics	Aesthetics is defined as the feeling of beauty when a user interact to the system[14].
Novelty	The feeling of surprise and excitement when interacting to interface [12]
Perceived usability	Perceived usability is defined as the perception when system is not beautiful means the system is not usable. [15]
Endurability	This attribute is defined as the state when the system is worth to try again in the future [16]
Felt involvement	The feeling of self-relevance which affects attention [17]

III. DESIGN OF CONCEPTUAL FRAMEWORK FOR DIGITAL WORKPLACE DIFFUSION

In this part, we will introduce the conceptual framework for digital workplace diffusion. As we can see fig.2, this model contains integration between diffusion theory, digital workplace concept, and user engagement. These two concepts lie on control which responsible to manage risks, compliance, run good IT governance along digital workplace diffusion.

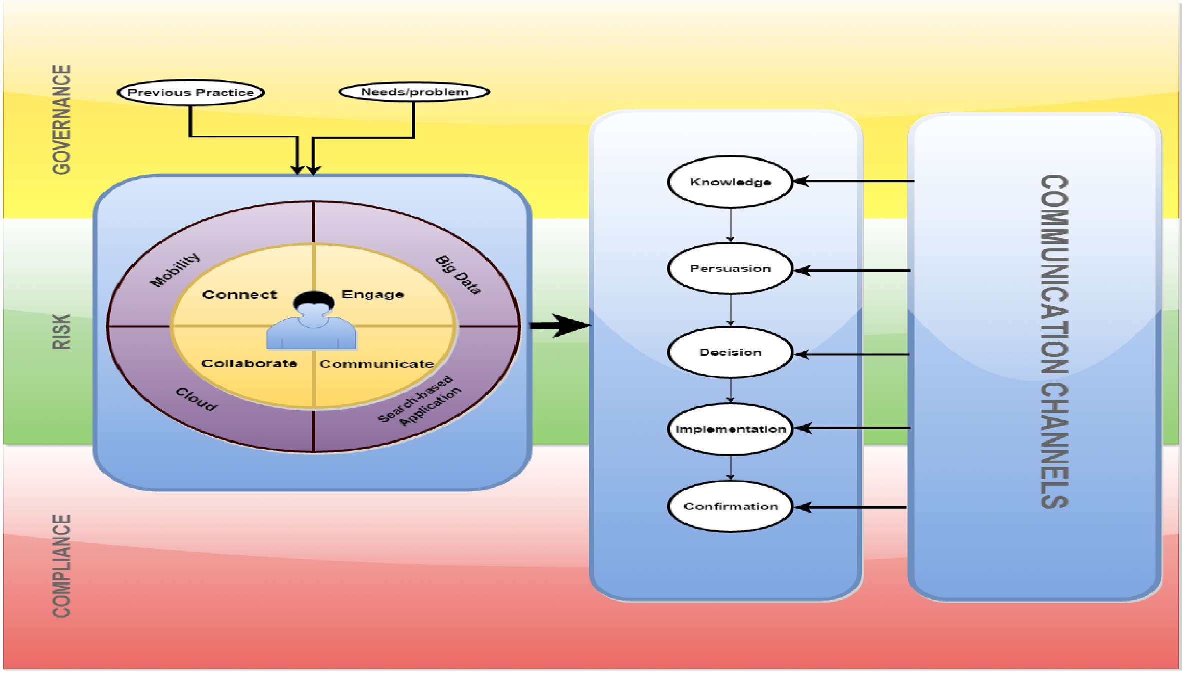

Fig.2. Conceptual framework of Digital Workplace Diffusion

A. Initial Stage

Before we decide to use the innovation, we have to identify where we are now. This activity enables us to define current status, future target, and how digital workplace helps organization to reach its goal. The common reasons of digital workplace diffusion are awareness and organization needs. It is an awareness if organization becomes more active in improving and monitoring the IT trends. In this state, organization is more open to the digital workplace because it has advanced understanding to the advantage of implementing digital workplace.

Organization needs are caused by dissatisfaction to current practice. In this state, organization would examine a new technology to fulfill its needs. Digital workplace diffusion enables the organization to improve the used of technology from a previous state to the desired state. It involves new technologies and employees' habits which are enable the core competencies of a digital workplace.

B. Digital Workplace Technology

Nowadays, many new technologies emerged to support business. Those technologies, such as cloud computing, big data, mobile, and search-based application, create an opportunity for organization to create a digital workplace. Along with that technology, we also need to consider business goal and capability of organization in order to build a digital workplace that can be accepted by employees and give a positive impact for organization.

Digital workplace is focused on developing applications in mobile environment. Cloud computing enables people to access data or application by using any devices which are already used by employee, such as PC, tablet or smartphone. It also enables people to connect and work together no matter where they are.

Organization also needs many data to support its business. These data comes from many different sources and in different format. Big data helps employee to collect, process and visualize data faster. Given the amount of data involved, we need to develop a search-based application in order to help employee access those data faster.Integration of those technologies will make a big improvement to organization's performance.

C. Innovation-Decision Process

In this part, we will discuss about innovation-decision process. It contains knowledge, persuasion, decision, implementation, and confirmation. In every activity, there will be communication channel to facilitate us to introduce and give an understanding about the digital workplace to employees as adopters.

1. Knowledge

In this stage, we will introduce the idea of Digital Workplace to the decision-making unit, it can be IT Manager and IT supervisor. This stage is defined as an important state because it determines how the process will proceed. This stage matters because if decision-making unit fails to receive the basic understanding about functions of Digital Workplace, they will reject to use it. According to this issue, we have to focus on how we present the idea. Moreover, we have to take concern at socioeconomics characteristic, personality variables, and communication behavior to identify the type of adopter. It matters to help us in choosing suitable communication channels.

Socioeconomics is about how to identify education level, income, and occupation. Moreover, personality variables are used to find personality type. At this activity, we can use five factor model (FFM). FFM is widely known as psychology model to identify human personality. This model contains openness to experience, conscientiousness, extraversion, agreeableness, neuroticism.

After we identify employees' personality type, we can continue to check on communication behavior. It contains three characteristics; communication quality, the extent of information sharing between employees, and participation in planning and goal settings [18]. After we analyze three characteristics of decision making unit, we can find the type of digital workplace adopter.

According to the type of adopter, we can choose communication channel to present the digital workplace. Earlier adopter will more interest in cosmopolite mass media channel. Meanwhile, later adopter will more interest in cosmopolite interpersonal channel.

2. Persuasion

Persuasion stage aims to form the attitude of decision-making units toward Digital Workplace. After they gain some understanding about Digital Workplace through knowledge stage, they will more involve with the idea of digital workplace. They will seek more information about digital workplace. Table 2 shows the characteristics of digital workplace which have to be covered. Finding the characteristics of digital workplace as innovation is important to decrease the degree of uncertainty for individual. In this stage, we can use interpersonal channel to communicate with decision-making unit.

TABLE 2. CHARACTERISTICS OF DIGITAL WORKPLACE

Characteristics	**Description**
Relative Advantage	How Digital Workplace increasing user engagement and job satisfaction which can lead to productivity and profit organization
Compatibility	Digital Workplace meets organization needs and must be compatible with existing value and system
Complexity	Digital workplace must be easy to be understood and used. Additionally, along the implementation and use, there will be many changes which come from business and technical, so digital workplace must be easy to customizable
Trialability	At the persuasion stage, the idea of digital workplace seems to be uncertain, to minimize the risk which will lead to rejection, we must add prototype. By using prototype, we can present and demonstrate digital workplace. Moreover, we must enable the users to try or even use it to prove the list of advantages.
Observability	As we discuss at trialability section, we enable the organization to try the prototype and understand the mechanism. It will ensure that organization are able to check on what digital workplace can do for the organization.

3. Decision

Organization will decide to adopt or reject the implementation of Digital Workplace for their organization at decision stage. In this stage, we will use interpersonal communication channel. If the organization decides to adopt, we can prepare the requirements to implement the suggested innovation. In contrary, the rejection can be a passive and active rejection. Passive rejection means the organization refuses to adopt innovation at the first presentation of innovation. Meanwhile, the active rejection is defined as the state where the organization considers adopting the innovation but at the end organization decides to not implementing the innovation [8].

4. Implementation

When an organization decides to adopt Digital Workplace, the next step is to implement it in organization. The organization and developer must discuss IT infrastructure, configuration, standard operations, and set up a support system to ensure the digital workplace runs properly. We also need to communicate the changes which are caused by implementation of Digital Workplace to all employees. In this stage, we can combine interpersonal and mass media communication channel.

5. Confirmation

After implementing Digital Workplace in organization, we still have to review and monitor impact of implementation to organization. It will ensure that organization make a right decision. Organization can reverse the decision if they find the implementation of Digital Workplace has failed to boost the desired performance.

In this stage, we also propose user engagement measurement. Implementation of IT, including digital workplace, needs full participation of employees as users. We propose the use of user engagement measurement which is formulated by Heather O'Brien. Using this measurement, we will measure every attributes of user engagement.

D. Control

This part shows the elements to help organization in managing, controlling, and ensuring the digital workplace diffusion align to the organization's standard, and external's standard such as government's regulation.

1. IT Governance

We need IT governance along the implementation of digital workplace as an innovation. IT governance helps organization to ensure the use of IT is effective, efficient, and meet its goals. IT governance enables organization to fix organization's culture issue along with digital workplace as IT product implementation and measure IT and business performance. Digital workplace as IT product requires an enterprise-wide focus, by implementing digital workplace organization must have solution for issues which are related to business and strategic. There are many frameworks which are related to IT governance, such as COBIT, ITIL, etc. Every organization is unique, so we can discuss with the board about what kind of IT Governance that might be suitable.

2. Risk Management

Every project has risk. If risk is being left uncertainty and ignored, it will become seeds of failures for organization. Digital workplace diffusion is big project with high level of cost, risk management must be run to help organization in identifying, monitoring, prioritizing, minimizing risk, formulating guide, strategy, and policy, and more important is providing solution of risks using technology, human and organization resources. There are many frameworks for IT risk management, such as Cobit, Coso's Enterprise Risk Management (ERM), Risk Management Framework (RMF), etc.

3. Compliance

The implementation and operation of digital workplace must comply to organization policy and regulation in which it is implemented. When we decided to use digital workplace, we must define all the relevant rules and regulations, such as privacy law, Data Protection Act (DPA) etc. Thereafter, those rules and regulations along with organization policy will serve as guidelines for developer to develop the digital workplace. In operation of digital workplace, we also need to monitor employee's compliance to organization policy and regulation to avoid misuse of digital workplace.

IV. CONCLUSION

In this proposal, diffusion theory is suggested as an approach in introducing and implementing digital workplace. We also propose the use of user engagement-centric and control. User engagement-centric helps organization to measure the degree of user involvement and acceptance to digital workplace. Meanwhile, control which contains IT governance, risk management, and compliance help organization to align digital workplace and its business, evaluate, and monitor the activities in digital workplace. Further research of this proposal is validation of the model and the impact of digital workplace to organization culture.

REFERENCES

[1] N.Dotson, Building a Better Digital Workplace., San Fransisco, CA, CMSWire.

[2] D.Lavenda, “What Gartner Wants You to Know About the New Digital Workplace.” Internet : http://www.cmswire.com/social-business/what-gartner-wants-you-to-know-about-the-new-digital-workplace/, May 28, 2015 [Sep. 5, 2015].

[3] M.I.Hwang and R.G.Thorn, “The effect of user engagement on system success: a meta-analytical integration of research findings.” *Information and Management Journal*, vol.35, pp.229-336, Apr.1999.

[4] J. Mcconnell, “Digital Workplace - Trends and Transformation,” Washington D.C, 2013.

[5] F. Herrera et al, “The digital workplace: Think, share, do,”Deloitte, Canada, 2011.

[6] M.White.”Digital workplaces: Vision and Reality”.*Business Information Review,*vol.29, 205-214, Des.2012

[7] M.S. Elliott and K.L. Kraemer. Computerization Movements and Technology Diffusion: From Mainframes to Ubiquitous Computing. New Jersey: American Society for Information Science and Technology. 2008.

[8] E.M.Rogers. *Diffusion of Innovations*. New York: Free Press, 2003.

[9] R.B. Tucker. Innovation is Everybody's Business: How to Make Yourself Indispensable in Today's Hypercompetitive World.Hoboken, NJ: John Wiley & Sons, 2011, pp.12-13.

[10] J.Lehmann, M.Lalmas, E. Yom-Tov, G.Dupret,”Model of User Engagement,”in *UMAP'12 Proceedings of the 20th international conference on User Modeling, Adaptation, and Personalization*,Montreal, CA, 2012, pp.164-175.

[11] B.Laurel, *Computer as Theater*. Boston, MA: Addison-Wesley Longman, 1993.

[12] H.L.O’Brien and E.G.Toms,”The Development and Evaluation of a Survey to Measure User Engagement,” *Journal of the American Society for Information Science and Technology*, vol.61, pp. 50-69, Jan 2010.

[13] M.Jennings, “Theory and models for creating engaging and immersive e- commerce websites,”in *SIGCPR '00 Proceedings of the 2000 ACM SIGCPR conference on Computer Personnel Research*, New York, USA, 2000, pp. 77-85.

[14] R.Hunicke, M. LeBlanc, R.Zubek, “MDA: A Formal Approach to Game Design and Game Research,” in *Proceedings of the 2004 AAAI Workshop on Challenges in Game Artificial Intelligence*, San Jose, USA, 2004.

[15] O.Eliav and T. Sharon, “Usability in Israel," in *Global Usability*, I.Douglas and L.Zhengjie, Eds.London: Springer, 2011, pp.169-194.

[16] Y.Hung, W.Li, Y.S Goh,“Integration of Characteristics of Culture into Product Design: A Perspective from Symbolic Interactions,”in*Cross-Cultural Design, Methods, Practice, and Case Studies*, vol. 8023.P.L.P.Rau, Ed. Berlin: Springer,2013, pp.208-217.

[17] R.L.Celsi and J.C.Olson.“The Role of Involvement in Attention and Comprehension Processes.”*Journal of Consumer Research*, vol.15, pp.210-224, Sep.1988.

[18] J. J.Mohr and R.E.Spekman, “Characteristics of Partnership Success : Partnership Attributes, Communication Behavior, and Conflict Resolution Techniques.” *Strategic Management Journal*, vol.15,pp.135-152, Feb.1994.

Effect of Electromagnetic Radiation on Rice Plant Growth in Microgravity Environment

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***Abstract*— The existance of wireless technology indirectly take huge positive and negative effect on all organism on this earth especially plant. On previous research held by Racuciu, Lftode, and Miclauss said there is an effect on corn plant growth which treated by electromagnetic waves in 1 GHz frequency. The objective of this research is to prove the effect of electromagnetic waves on the plant in microgravity environment. In this research, 2.4 GHz electromagnetic waves were used and the testing environment will be used clinostat. The result on this research proved that there is an effect between electromagnetic waves 2.4 GHz on plant especially rice plant in microgravity environment. The main effect that we discovered is faster growth than normal condition.**

Keywords—Electromagnetic, Microgravity, Plant.

I. Introduction

Earth is the place all of creature and organism live within e.g. human, animal and plant. Plant is one type of living creature that has many benefits for humankind. Indonesia is one of the agricultural countries where most of the plant population due to tropical climate and soil texture that encourages farming predominately. Furthermore, Indonesian staple food is rice that derived from rice plant (*Oryza Sativa*).

Nowadays, a lot of researchs have been done to develop superior rice seedlings aiming to produce more rice in order to meet the food demand of Indonesian people, as we know rice fields getting smaller due to a lot of building and housing, especially in urban areas. Along with the development of information technology, many new technologies that facilitate human life, one of them is the wireless communication technology. With this technology allows humans to communicate remotely by using electromagnetic waves as a medium. The increasingly widespread use of this technology in everyday life, the impact on the number of BTS (Base Terminal Station). Of course as more BTS established, it will bring a great impact to the living beings who live around the Tower. During these studies that examine the effect of radiation of electromagnetic waves only focuses on a man, so the researchers rarely examined the impact of electromagnetic radiation for animals and plants.

Outer space environment where humans rarely to think about the impact of the space environment on the lives of living beings. This is due to the difficulty to build the same conditions as in space. At this time, we can build a condition where the condition is nearly resembling the conditions in outer space with a tool called clinostat. According Fathona et al. [4] reported that Clinostat built with the intention of creating a zero-gravity environment. From the facts presented above, researchers interested to research that aims to find the relationship between the development of the rice plants for space environment and radiation of electromagnetic waves.

II. STATE OF THE ART

Research on the effects of electromagnetic waves on plant growth had previously been done by Racuciu, Lftode and Miclaus [1] with the electromagnetic wave frequency of 1 GHz on corn and produce data as well as the period of exposure to a different microwave can affect seed germination and plant development started with the process mobile, such as mitosis. Study the effects of microwave exposure is important because they are worried about the effects that occur not only on the human body or a different animal, but also in plant organisms. As a result of this experimental study, we can say that the lower thermal oven 1 GHz microwave exposure is capable of initiating mutagenic effects and inhibition of cell proliferation and differentiation in seeds exposed. This indicates that there is an influence of electromagnetic waves on the growth and development of plants, further on previous research conducted by Igor Petrov Yu, Tatyana V. Moisseva and Elvira V. Morozova about [5] the effect of microwaves on plant wave results showed that there was accelerated growth of the seeds to plant without changing the cell structure of the plant. From both research before we can know that there is the influence of electromagnetic waves on the crop but research to investigate more about the influence of electromagnetic waves on this plant. On this paper researchers try to apply electromagnetic waves effects to plants in microgravity environment / space environment.

A. Clinostat

As quoted in the book made by the United Nations Programme on Space Applications Clinostat can be defined [2] assorted clinostats been developed, differing in the number of axes of rotation and operation mode with respect to the speed and direction of Clinostat form there is a form of two-dimensional (2-D), or an axis, Clinostat has a single rotational axis, which runs perpendicular to the direction rotation. of gravity vector. A Clinostat three-dimensional (3-D) has two axes of rotation, which is perpendicular to one another. Clinostat this type runs at a constant speed and in a constant direction, specifically called Clinostat 3-D. However, if the two-axis rotating at different speeds and in different directions. The present study concentrates on the comparison between different devices in order to determine the condition of the simulation is right for the object being treated. Clinostat can be equipped with the capacity for microscopy, online measurement of kinetic or chemical fixation samples during rotation. A rotation on Clinostat often called "clinorotation".
Further [2] Clinostat movement analysis based on classical mechanics of the rotation and non-inertial frame. Rotation of

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Clinostat generate fictitious force because there is a framework that is rotated coordinate system and the coordinates of the stationary frame. Both frameworks Clinostat rotated at a rate equal to the angle of the rotary axis, which is depicted as a dotted line as shown in Figure 1.

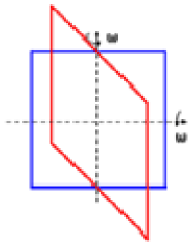


Figure 1. Clinostat Design

Figure above can describe with this equation:

$$F' = F - FCent - FCor - F'' \quad (1)$$

With F is the force relative to the frame of reference that is stationary, F 'style relative to the reference frame rotating which is two order clinostat, Fcent a centrifugal force, F' 'style which is caused by changes in rotational speed against time and Fcor is the Coriolis force generated when particle moving in a framework that is playing. Decrease Equation (1) is analogous to the two coordinate frame where XYZ is a framework that is silent and X'Y'Z 'framework that is playing is shown in Figure 2. The zero point of the second frame these coordinates coincide. Suppose framework X'Y'Z 'moves with a constant angular velocity ω, and we place a particle at the point P and the vector A.

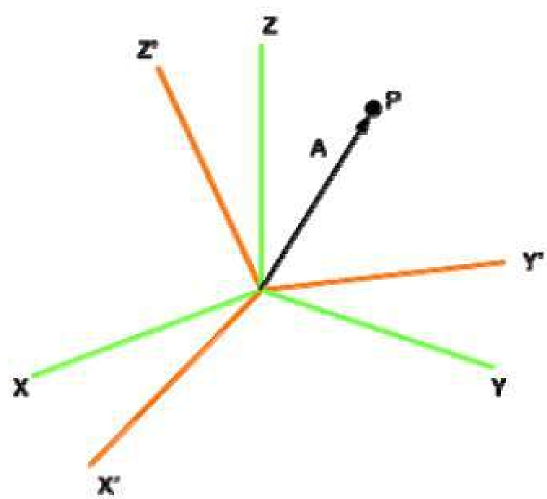


Figure 2. The intersection of the lines of force

By applying some conditions on Clinostat then some fictitious force would disappear or zero so that the fictitious force which is still in force is centrifugal force. Fcor koriolis style will be zero because the particles do not move to the coordinate frame is rotated; F " style caused by changes in rotational speed will be zero because the coordinate frame rotating speed ω be made permanent; and we assume no frictional force or the influence of external forces so that F is equal to zero. Thus the force experienced by the particles only fictitious centrifugal force which is directed away from the rotary axis radially with a large following.

$$F_{cent} = m\omega^2 r$$

where m is mass and r is the radius. Because the quantity of the desired radius and acceleration are known, then we can determine the angular velocity of the DC motor so that the sample is placed on Clinostat will accelerate considerably smaller than the acceleration of gravity. DC motor speed is set by applying a PWM method, namely the setting powering the motor by regulating the time the motor off and on, or so-called duty cycle settings D.

$$D = \frac{T_{on}}{T_{on} + T_{off}} 100\%$$

Ton is the time motto tons and Toff is the time the motor off. This setting performed by the microcontroller by providing digital state 0 and 1 on the DC motor during a certain period in accordance with the calculation of the duty cycle D in Equation (3). Thereby forming microgravity environment.

B. Rice Plant (Oriza Sativa)

The rice plant is a plant that is used as a staple food, especially for the people of Indonesia. Even plants in nearly all parts of Indonesia are rice paddy which illustrates that the importance of this for the people of Indonesia.

Today many breeding of rice so that rice varieties more and more stuff. In general, the rice began to grow around 5-6 months. [4] The growth of the rice plant is divided into three phases: (1) vegetative (early growth until the formation will panicles / primordia); (2) reproductive (primordia until flowering); and (3) maturation (flowering to mature grain).

Figure 3. Phase Growth Rice Plant

C. Gravity Effect on Plant

The effect of gravity was very influential on plant growth as quoted in the book made by the United Nations Programme on Space Application argued that Gravity-dependent growth is based on a highly complicated stimulus-response chain. The physical signal of gravity has to be transformed into a biochemical signal, which leads to a physiological response. Gravity is a force that acts on mass. A mass has to be transported in the gravitational field in order to create sufficient energy for the activation of a biological sensor. In order to transfer this signal at the level of a single cell, a mass which is denser than the surrounding medium must exist. This heavier mass sediment under the influence of gravity, thereby activating gravity specific receptors. Candidates for sedimenting mass are either intracellular statoliths or the entire cell mass (protoplast).9 The process that perceives gravity is called graviperception. Figure I shows a typical gravitropic response of a plant.

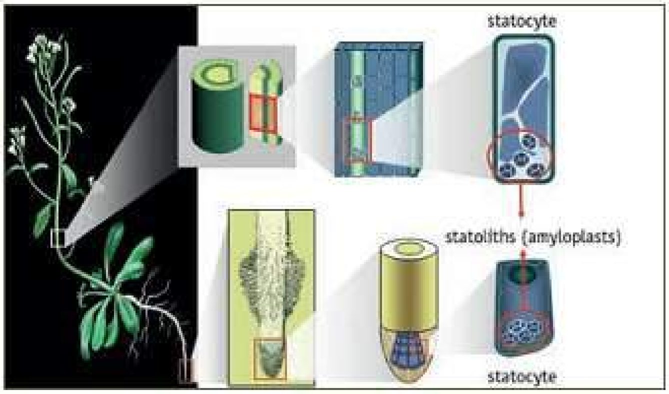


Figure 4. Growth of root plants with gravity vector

Furthermore, in United Nations Programme on Space Application also argued gravity is the stimulus that uses plants to grow roots toward the vector of gravity (below), retaining the plants in the ground, and to sprout in the opposite direction of the gravity vector (above), out of the land toward the sun. To understand the "up" and "down" is required for the survival of plants on earth.
It is also very necessary for all life on earth because photosynthesis necessary for the production of food and oxygen.

III. SYSTEM DESIGN

In the process of designing this experiment we will use three research objects and 2.4 GHz frequency electromagnetic waves to prove that there are significant electromagnetic wave frequency of 2.4 GHz on the growth of plants in the environment microgravity. The third subject may explain below:

1. Subject A = Rice plants are treated with electromagnetic waves and also microgravity environment.


Figure 5. Subject A

2. Subject B = Rice who just put in the microgravity environment.


Figure 6. Subject B

3. Subject C = Rice grown without the use of electromagnetic radiation and microgravity environment.

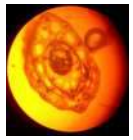

Figure 7. Subject C

IV. RESULT

Based by research above we have result in subject A growth of Rice plant is faster than subject B and subject C because as past research who did by Racuciu, Lftode and Miclaus that use 1 GHz electromagnetic waves used on corn plant it can make corn plant growth faster and result of subject A is related with past result research. In microgravity electromagnetic wave have a role to make plant growth faster in microgravity.
In Subject B that provide by microgravity environment growth of rice plant is more slow than subject B because microgravity take effect on rice plant.
In subject C the rice plant planted in normal condition and the growth of rice plant is growth normally.

Of the three subjects are studied, it can be obtained as follows:

Subjek A	Subjek B	Subjek C
1. Growth Faster.	1. Growth more slow than normal.	1. Growth normal.

2. Growth with Kemotropism (root move to the nutrition way)	2. Growth with Kemotropism (root move to the nutrition way)	2. Growth with Geotropism (root move because gravity)
3. Result of root cell  Figure 8. Root Cell Cell structure different with normal condition.	3. Result of root cell  Figure 9. Root Cell Cell Structure more smaller than normal condition.	3. Result of root cell  Figure 10. Root Cell Cell Structure normal.

V. CONCLUSION

Based on research that have been done, we encountered problem that this research has not been able to present more profound data because the limited facilities used during the study. In addition, the research data is still limited to rice plant. In general, the possibility of the result will be able to be applied to all plant because on previous research that performed by Racuciu, Lftode and Miclaus present the same data, but the research conducted by Racuciu, Lftode and Miclaus using normal environment. Different of past study with the present study we use a microgravity environment. With this research we add new knowledge about research in microgravity environment, because of that we can conclude :

1. With the 2.4 GHz frequency electromagnetic radiation can accelerate the growth of this is due to electromagnetic wave radiation heats the plant cells so that photosynthesis can be optimized.
2. With the 2.4 GHz frequency electromagnetic radiation can help plants to live in an environment microgravity (space) so as to allow for the existence of extraterrestrial life.

VI. FUTURE WORK

In future work research hope other work can prove if electromagnetic with different frequencies 1 GHz or 2,4 GHZ wil take same effect with past research. Also researcher hope research with space environment ore much that can be new knowledge about space environment and make a calculation to possible live in space.

REFERENCES

[1] R. Mihaela, l. Cora and M. Simona, “1GHz Low-thermal Microwaves Effect on Miotic Division of Vegetal Tissues,” International Conference and Exposition on Electrical and Power Engineering (EPE 2014), October 2014.

[2] Fathona LW, Surrachman A, dkk, “Clinostat Tiga Dimensi sebagai Simulator Microgravity untuk Berbagai Eksperimen Tanpa Bobot,” ITB Journal ISSN : 2085-2517, February 2011.

[3] H. L. Krauss, C. W. Bostian, and F. H. Raab, Teacher’s Guide to Plant Experiments in Microgravity, New York: United Nations, 2013.

[4] Makarim Karim dan Suhartatik E.. Morfologi dan Fisiologi Tanaman Padi. November, 2009.

[5] Petrov Igor Yu, Moiseeva Tatyana V. and Morozova Elvira V., “A Possibility of Correction of Vital Proccess in Plant Cell With Microwave Radiation”, International Conference and Exposition on Electrical and Power Engineering.August,1991

Development of Radio Telescope Receiver Based on GNU Radio and USRP

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***Abstract*—This paper presents the development of radio telescope receiver based on GNU radio and Universal Software Radio Peripheral (USRP). Usually an existing radio telescope receiver is only able to be used for fixed frequency and signal processing system; hence the other different frequency receiver needs to be carried out with different types of observation. Here, the development of radio telescope receiver includes the hardware system and the signal processing system. Then, the design of radio telescope receiver is developed based on GNU Radio and implemented by using USRP. From the result, it shows that the implemented radio telescope receiver which is applied for continuum observation can clearly detect signals at frequencies of 322MHz and 406MHz n which those frequencies are usually used in radio telescope.**

Keywords*—GNU radio; radio telescope receiver; software defined radio; universal software radio peripheral.

I. Introduction

Basically radio telescope is a device that is used by astronomer to observe celestial objects by radio waves [1]-[2]. It is comprised of some components typically used in radio communication such as antenna, amplifier and receiver [2]. In actual, there are several methods to observe celestial objects such as using light waves, infrared waves, microwaves, or gamma rays [3]-[4]. All of these tools have respective roles in the observation. Radiation of electromagnetic waves in radio frequency, for example, can not be detected by optical telescopes; however it is detectable by radio telescopes and vice versa. So the use of different methods for observing celestial object is complementary.

Furthermore, the advantage of using radio telescope rather than optical telescope is the observation could be held anytime. Meanwhile, the exiting radio telescope system is usually designed only for specific application for example spectral observation or spectrograph observation. Therefore, for observing other application, it requires another system. This circumstance gives a big restriction since the economical value of the radio telescope receiver is usually very expensive. One method in overcoming the problem is using software defined radio (SDR) technology which allows the user to design radio transceiver by software. There are some applications that can be realized using SDR, such as transceiver system, base transceiver station (BTS) system, and radar system [5]-[8].

In principle, an SDR is a radio device which its part of components can be set through the software. It usually consists of two components, namely software components and hardware components. Some SDR frequently used is WiNRADiO and GNU radio. By using SDR, the selection of operating frequency can be conducted freely and easily with the condition of antenna and amplifier having fairly wide frequency range. Therefore, an SDR can be an alternative solution to be implemented in the design of radio telescope system since it can be applied as a receiver that accommodates multiple techniques.

In this paper, the development of radio telescope receiver design using SDR is presented. Here, the SDR used for developing radio telescope receiver is based on GNU radio. Meanwhile, the implementation of radio telescope receiver is using USRP. The development of radio telescope receiver includes the hardware system and the signal processing system. Some brief description of telescope system is pointed out at first before the radio telescope receiver design. Then some discussion related the implementation of radio telescope receiver using USRP and its results are presented, and then followed by the conclusion.

II. Radio Telescope System

A. Basic System of Radio Telescope Receiver

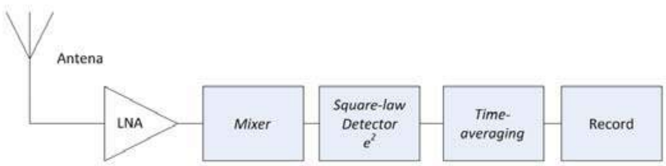

Figure 1. Block diagram of basic radio telescope receiver

Fig. 1. shows a block diagram of basic system of radio telescope receiver which comprises of antenna, low noise amplifier (LNA), square-law detector and time-averaging processor [1]. The signal from celestial objects is catch by the antenna and then amplified by the LNA. The square-law detector gives the power of received signal and the integrator takes a range of times to integrate the signal so it can be expected the mean value of signal at the integrator output. This

basic system is only can be used to observe the continuum properties of celestial objects. Meanwhile, to implement the receiver system for other properties, it is required to modify the receiver form basic system.

B. Radio Telescope Observation Techniques

There are two observation techniques which uses a single dish of radio telescope, namely continuum observation and spectral observation [1]. The continuum observation only observes the signal power of celestial objects, whilst the spectral observation is observation that gives us the spectral properties of a celestial object. All of those techniques have a different challenge in which the continuum observation is often used to make an image of the sky so the challenge is the resolution of the antenna, whereas the spectral observation is used to determine what chemical compounds that exist in the celestial object. For example at frequency of 1420MHz, it can be determined whether the object has a hydrogen component or others. The challenge of spectral observation is the power of the signal which is very weak.

III. Design of Radio Telescope Receiver

A. Radio Telescope Receiver for Continuum Observation

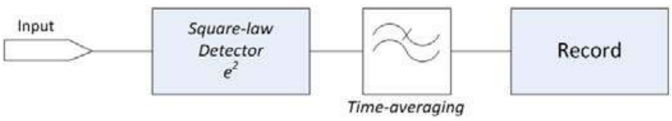

Figure 2. GNU radio flow graph of radio telescope receiver for continuum observation

In order to be implemented as radio telescope receiver for continuum observation, the basic system of radio telescope receiver flow graph shown in Fig. 1 should be modified. The modification of flow graph based on GNU radio is illustrated in Fig. 2 in which the flow graph of radio telescope receiver brings the signal from USRP to be processed. The first flow graph records the signal in frequency domain. The flow graph obtains the magnitude of signal and multiplies it by the signal itself. Then, the signal is convoluted by single-pole infinite impulse response (IIR) filter where the filter works as time-averaging. The difference equation of single pole IIR filter is expressed in (1).

$$y(n)-(1-\alpha)y(n-1)=\alpha x(n) \qquad (1)$$

It shows that the single pole IIR filter gives the past output and input in a specific weight that can be determined from the integration time. If the value of alpha equals 1 by sample rate, for example, it is obtained one second of integration time.

B. Radio Telescope Receiver for Spectral Observation

The main idea of signal processing in radio telescope receiver for spectral observation is in obtaining the averaging signal magnitude at various frequencies. So the flow graph of radio telescope receiver for spectral observation illustrates in Fig. 3 begins the process by transforming signal from time domain form to frequency domain. Here, the frequency domain signal is presented as a vector. Then the convolution is performed between the signal and single pole IIR filter. Fig. 4 shows how the convolution works. The output of the single pole IIR filter is an average signal in frequency domain. Thus, the sub sampling process is performed in order to push the recorded file size.

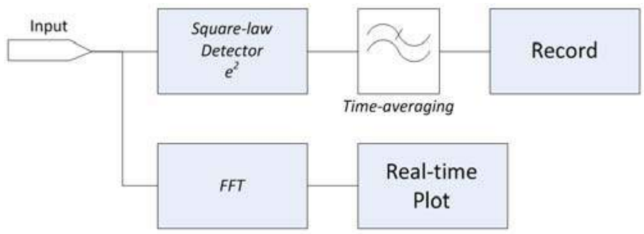

Figure 3. GNU radio flow graph of radio telescope receiver forspectral observation

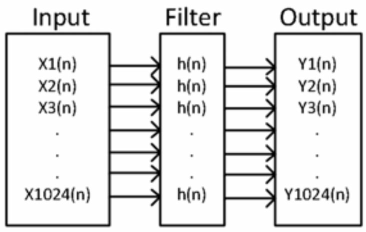

Figure 4. Convolution of a vector

C. Hardware System of Radio Telescope Receiver

As shown in Fig. 5, hardware system of radio telescope receiver is constructed by an antenna, LNA, jumper, and coupler. The important rules of arranging the hardware system are to amplify the weak signal and to keep the noise power as little as possible. To amplify the signal, the high gain antenna and high gain LNA are used for the system. To keep the noise low, it is required to minimize noise generation by hardware component. Therefore, an LNA with very small noise figure is employed; therefore the output SNR is not much different with the input SNR. The configuration of hardware system for radio telescope receiver is tabulated in Table 1, while the antenna and USRP used for implementation of radio telescope receiver are shown in Figs. 6 and 7, respectively.

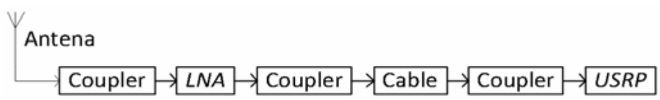

Figure 5. Block diagram of hardware system for radio telescope receiver

TABLE I. Hardware System of Radio Telescope Receiver

Component	
Antenna	LPDA antenna for continuum observation and parabolic antenna for spectral observation
LNA type	Mini Circuit ZX60-33LN+, RAS1420MHz LNA
Cable	LMR-400

Figure 6. Antenna for radio telescope receiver, LPDA for continuum observation (left), parabolic antenna for spectral observation (right)

Figure 7. USRP for implementation of radio telescope receiver

IV. Result and Discussion

Fig. 8 depicts the result of GNU radio based radio telescope receiver for continuum observation implemented using USRP. Two sinusoid signal corrupted by noise signal is set for the system. Both of those signals have the same amplitude. It is shown that the value recorded by the system provide a constant power value which means the system can detect the total power among given bandwidth. At the early samples, it is difficult to see the differences between noise and signal, however after so many integration, the system can determine whether the received signal or the noise.

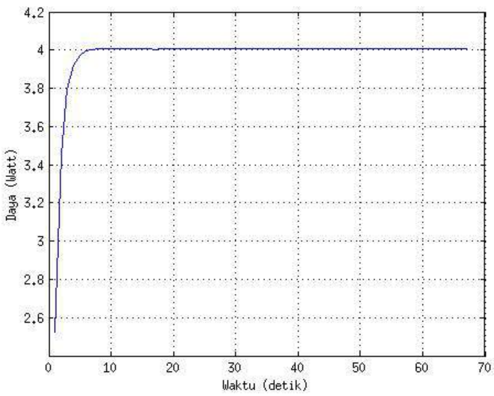

Figure 8. GNU radio result for continuum observation

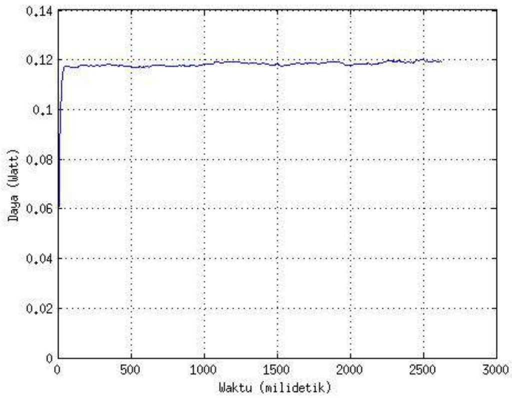

Figure 9. Average output power recorded at frequency of 322MHz

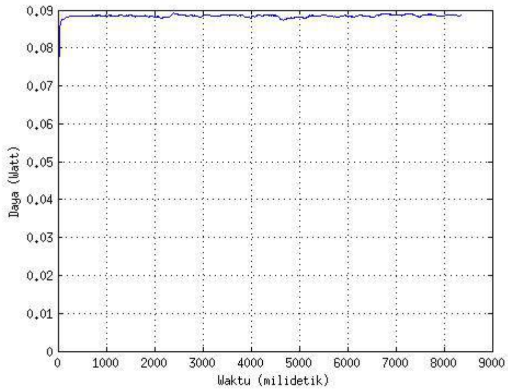

Figure 10. Average output power recorded at frequency of 406MHz

Figs. 9 and 10 plot the result of continuum observation at frequency of 322MHz and 406MHz with bandwidth of 5MHz, respectively. The figures show that the implemented radio telescope receiver successfully produces the total power of received signal. It indicates that the signal power at frequency of 322MHz is stronger than the signal power at frequency of 406MHz. It is noticeable that the result is similar to the galactic background noise which has a peak value at frequency of 30MHz and a minimum value at frequency of 1400MHz.

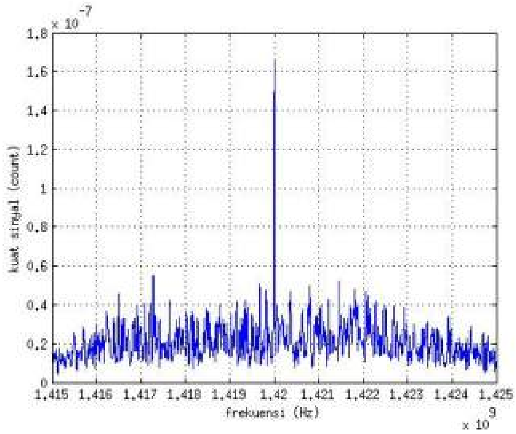

Figure 11. Integration result at first 500 samples

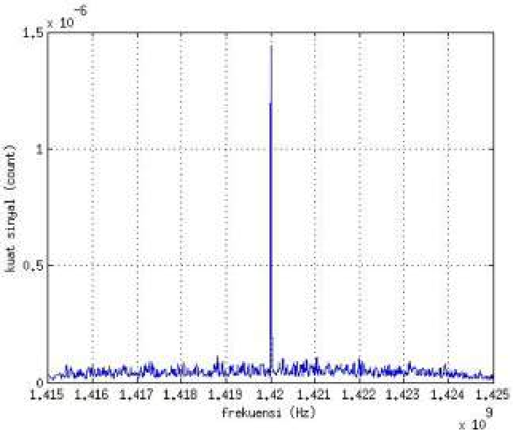

Figure 12. Integration result at first 6000 samples

Furthermore, Figs. 11 and 12 show the comparison between the result of integration at first 500 samples and 600 samples, respectively. It is noted that the longer integration time the noise is getting down. This complies with the prediction and can be proven by standard error equation in which the error equals with variance of- and inversely to sampling number.

V. CONCLUSION

The development of radio telescope receiver based on GNU radio and USRP has been presented. The developed system of radio telescope receiver has covered the hardware system and the signal processing system. It has been shown that the implemented radio telescope receiver which was applied for continuum observation can detect signals at frequencies of 322MHz and 406MHz which are usually used in radio telescope. In addition, to compensate the sensitivity of USRP, it is necessary to have a very high gain of. Meanwhile, the longer integration time of system is required since for more complicated system the noise will be higher.

REFERENCES

[1] J. D. Kraus, *Radio Astronomy*, 2nd ed., Cygnus-Quasar Books, 1986.

[2] T. L. Wilson, K. Rohlfs, and S. Hüttemeister, *Tools of Radio Astronomy* (*Astronomy and Astrophysics Library*), 5th ed., Springer, 2009.

[3] M. A. Covington, *Celestial Objects for Modern Telescopes: Practical Amateur Astronomy Volume 2*, 1st ed., Cambridge University Press, 2002.

[4] P. S. Harrington, *Star Watch: The Amateur Astronomer's Guide to Finding, Observing, and Learning about Over 125 Celestial Objects*, 1st ed., Wiley, 2003.

[5] J. R. MacLeod, T. Nesimoglu, M. A. Beach, and P. A. Warr, "Enabling technologies for software defined radio transceivers," in *Proceeding of MILCOM 2002*, Anaheim, USA, Oct. 2002, pp. 354–358.

[6] M. Ben-Romdhane, A. Maalej, R. Lahouli, and C. Rebai, "Test setup and spurious replicas identification in time-quantized pseudorandom sampling-based ADC in SDR multistandard receiver," in *Proceeding of 18th IEEE International Conference on Electronics, Circuits and Systems (ICECS)*, Beirut, Lebanon, Dec. 2011, pp. 180–183.

[7] P. G. Burns and M. C. Reed, "A complexity cost function for a network of WCDMA basestations using software defined radio with multi-user detection," in *Proceeding of IEEE 59th Vehicular Technology Conference (VTC)*, Milan, Italy, May 2004, pp. 2195–2199.

[8] A. Prabaswara, A. Munir and A.B. Suksmono, "GNU Radio based software-defined FMCW radar for weather surveillance application," in *Proceeding of 6th International Conference on Telecommunication Systems, Services and Applications (TSSA)*, Bali, Indonesia, Oct. 2011, pp. 227-230.

Designing Secured Data Using a Combination of LZW Compression, RSA Encryption, and DCT Steganography

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***Abstract*— Information security in mobile phones has become a challenge when large amounts of data exchanged on the Internet. Secure information transfer can be achieved by cryptography and steganography. Cryptography is a tool that provides encryption techniques for secure communication. Steganography is the technique of hiding information by way of the message of the picture, so that in addition to the person addressed, another person will not be aware of the existence of the message. In this study the issues to be discussed is the technique of message encryption with RSA (Rivest Shamir Adleman) method. Messages are encrypted after compressed by LZW (Lempel Ziv Welch) method, so it will reduce the size of the message that will be inserted and increase the capacity of messages that can be inserted. Messages that have been compressed and encrypted, is then hidden by DCT (Discrete Cosine Transform) techniques. With the incorporation of encryption techniques, steganography, and compression, the acquired information is more secure and its capacity is larger. For a 512x512 colored image there are 1536 characters enable for hiding. With LZW the amount of characters hiding will be about twice larger in this research.**

Keywords—steganography; cryptography; encryption; compression; LZW; RSA

I. INTRODUCTION

Information security is a key issue on the exchange of data in open networks like internet. Anyone can access the internet from any part of the world without being limited by geographic boundaries, national or international. While useful for a variety of purposes, but there are security risks associated with the transfer of information over the internet. Anyone can hack the information and then make the misuse of information or destroy information. Anyone can destroy the information if it is not fully guaranteed or protected.

Steganography and cryptography are very important role in information security. Steganography is a security tool that stores information confidential information in media files such a way that no one else except the sender and the intended recipient of the information can only guess where the information [1]. On the other hand, cryptography is enciphering and deciphering data and information with a secret code that can not be understood [2]. The purpose of both steganography and cryptography is the same but achieved in a different way. Good imperceptibility (difficult to detect hidden information) and sufficient data capacity (efficiency of hidden information) are two qualities that should be possessed by all steganography techniques [3].

Recent research conducted by the steganography include Rahna E. and VK Govindan on steganography techniques with unlimited payload [4], Saleh Saraireh [5], K. Challita and H. Farhat [6] proposed a combined scheme of steganography and cryptography that is resistant to attack. Research on DCT steganography method has been carried out by Prashasti Kanikar [7] and HB Kekre [8] where DCT for image steganography achieve better results than the DST (Discrete Sinusoidal Transform), Hartley wavelet, wavelet Walsh and Haar wavelet. However, there is no technique that combines encryption and data compression will be hidden in the picture. Compression is usually performed in the watermarking technique to cover image after the insertion process is done, as was done by Dr. Ajit Singh and Meenakshi Gahlawat [9]. In addition, these research will be implemented on mobile application.

This research will be conducted in the form of a text message encryption technique by RSA method. Messages are first compressed with LZW method for reducing the size of the message that will be inserted in order to increase the capacity of messages that can be inserted. Messages that have been compressed and encrypted, then hidden by DCT techniques DCT. The system would then be implemented on android. With the incorporation of encryption techniques, steganography, and the compression, the hidden message is expected to be more secure and its capacity can be larger.

II. THEORETICAL BACKGROUND

A. STEGANOGRAPHY

Steganography is derived from the Greek Steganos (hidden) and graphein (writing) [18], so steganography means hidden writing. Steganography is the art and science of hiding messages into a medium in a way that besides the sender and the recipient, no one actually knows or realizes that there is a secret message. In the modern steganography, steganography meaning evolved into withholding information on a digital

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media file, the media can include images, sound or video [10]. There are two methods of steganography, namely Spatial Domain and Frequency Domain [10].

- Spatial Domain Steganography

  Spatial domain technique uses a message on the intensity of the pixels directly. Least Significant Bit (LSB) is a first spatial domain steganography techniques most widely used. This technique of embedding bits of the message into the LSB of the pixel image. But the problem with this technique is that if the image is compressed, the data inserted may be lost. Thus, there is a fear for the loss of data that might have sensitive information [11].

- Frequency Domain Steganography

  In the frequency domain, the first image is converted and then the message is embedded in the image. When the data is embedded in the frequency domain, hidden data is in an area that is stronger, spread across the entire image, and provide better resistance against statistical attacks. There are many techniques used to change the image of the spatial domain to the frequency domain. The most common method is usually used in the frequency domain image processing is a 2D discrete cosine transform (DCT) [12] [13]. In this technique the image is divided into 8 × 8 blocks and DCT transformation on each block is done. DCT set the pixel image according to the value of the image frequency. Bits of data embedded in the high-frequency coefficients of DCT. In the 8x8 blocks after DCT transformation, the pixel in position 8th of row and 8th of column (8,8) is changed to δ for hidden information bit 1, and changed to $-\delta$ for information bit 0. δ is a positive value represents the hidden information bit. This embedding scheme causes only one bit embedded in 8x8 pixel. Thus, we can calculate how many bits maximum for an image file with certain resolution. If an RGB image with resolution MxN will be embedded by data, the maximum capacity for hiding data will be floor($MxN/8^2$)x3. If the colored image resolution is 1024x728, so the capacity will be floor($1024x728/8^2$)x3=27273 bits or about 3409 characters.

B. *RSA ALGORITHM*

RSA algorithm was introduced by three researchers from MIT (Massachusetts Institute of Technology), namely Ron Rivest, Adi Shamir, and Len Adleman in 1976. RSA encryption and decryption process based on the concept of prime numbers and modulo arithmetic. Both encryption and decryption keys are both integers. The encryption key is kept secret and is not known publicly (so-called public key), but kept secret key for decryption. Decryption keys are made of multiple primes together with the encryption key. To find the decryption key, must be factored a composite number into prime factors. In fact, factoring nonprime numbers into prime factors is not easy. There has been no efficient algorithms found in factoring it. The larger the number the more difficult to get the non prime factoring. The more difficult factorization, the stronger is the RSA algorithm [14].

RSA algorithm having magnitudes as follows:

- p and q are primes (secret)
- n = pq (not secret)
- $\Phi(n) = (p - 1)(q - 1)$ (secret)
- e (encryption key) (not secret)
- d (decryption key) (secret)
- m (plain text) (secret)
- c (cipher text) (not secret)

RSA algorithm is based on Euler's theorem which states that :

$$a^{\Phi(n)} \equiv 1 \pmod n \qquad (1)$$

with requirements :

- a must be relatively prime to n
- $\Phi(n) = n(1 - 1/p1)(1 - 1/p2) \ldots (1 - 1/pr)$, where p1, p2, ..., pr are the prime factors of n. $\Phi(n)$ is a function that determines how many of the numbers 1, 2, 3, ..., n which are relatively prime to n.

Based on the nature of $a^k \equiv b^k \bmod n$ for an integer $k \geq 1$, then the equation (1) can be written as :

$$a^{k\Phi(n)} \equiv 1^k \pmod n \qquad (2)$$

or $a^{k\Phi(n)} \equiv 1 \pmod n$

So that encryption and decryption are formulated as follows:

$$E_e(m) = c \equiv m^e \bmod n \qquad (3)$$

$$D_d(m) = m \equiv c^d \bmod n \qquad (4)$$

Algorithms which generate the key pair :

- Choose any two prime numbers , p and q
- Calculate n = pq (preferably $p \neq q$, because if p = q then $n = p^2$ so p can be obtained by numerical square root of n)
- Calculate :

  $$\Phi(n) = (p - 1)(q - 1) \qquad (5)$$

- Select the public key , e , which is relatively prime to $\Phi(n)$
- Generate a private key using the equation :

  $$ed \equiv 1 \ (\bmod\ \Phi(n)) \qquad (6)$$

C. *LZW Coding*

If any data file on a computer is viewed, character by character, one would notice that there are many recurring patterns. LZW is a data compression method that takes advantage of this repetition. The original version of the method was created by Lempel and Ziv in 1978 (LZ78) and was further refined by Welch in 1984. Like any adaptive/dynamic compression method, the idea is to first start with an initial model, secondly read data piece by piece and lastly update the model and encode the data as one go along. LZW is a "dictionary" -based compression algorithm, this means that instead of tabularizing character counts and building trees, as done in case of Huffman encoding, LZW encodes data by referencing a dictionary. Thus, to encode a substring, only a single code number, corresponding to that substring's index in the dictionary, needs to be written to the output file. It generally performs best for files with repeated substrings, such as text files [15],[16],[17].

The dictionary is assumed to be initialized with 256 entries (indexed with ASCII codes 0 through 255) representing the ASCII table. The compression algorithm assumes that the output is either a file or a communication channel. The input being a file or buffer. Conversely, the decompression algorithm assumes that the input is a file or a communication channel and the output is a file or a buffer.

LZW compression algorithm is completely described as following [18]:

1. Dictionary initialized with all the basic characters : {‘A’..’Z’,’a’..’z’,’0’..’9’}.
2. P ← The first character in the stream of characters
3. C ← The next character in the stream of characters
4. Are (P + C) string existing in the dictionary ?
   a. If yes, then P + C (combine P and C into a new string)
   b. If not, then :
      i. Output a string of code to replace P.
      ii. Add string (P + C) into dictionary and give the number / code that has not been used in the subsequent dictionary for that string.
      iii. P ← C.
5. Are there next character in the stream of characters ?
   a. If yes, then go back to step 2 .
   b. If not , then the output code that replaces the string P , the termination process (stop) .

On LZW decompression process carried out by the same principles as the compression process. The algorithm is given as follows [18]:

1. Dictionary initialized with all the basic characters : { ' A' .. ' Z ' , 'a' .. ' z ' , '0 ' .. ' 9 ' } .
2. CW ← first code from the code stream (pointing to one of the basic characters).
3. See the dictionary and output the string of code (string.CW) into a character stream
4. PW ← CW; CW ← next code from stream code.
5. Are CW string is existing in the dictionary?

   If yes, then :
   a. output string.CW is saved to character stream
   b. P ← string.PW
   c. C ← first character from string.CW
   d. add string (P+C) into the dictionary

III. System Design

Problem to be studied is the insertion technique on text by using a RSA Algorithm. The message inserted has been compressed by using LZW compression method so that the size of the message is smaller and the capacity of the inserted message is bigger.

The input message is a text used as the secret message that will be encrypted later. Customizable text input by the user. Text input will be encrypted using RSA method and using the key which will be used in the decryption process. Messages that have been encrypted earlier will be lossless compressed using LZW method so that the data size will be smaller. Results of the message that has been compressed to be inserted into the image with steganography using DCT. The image will be inserted chosen by the user. Image steganography results will be transmitted over data networks. Imagery will be received by the destination user and are processed to get the message that is needed. The image will be processed to get a text message hidden in the image. A text message has been obtained from the image is decompressed back to using LZW decompression technique. Text messages result from decompression was still not final results for the previous message encrypted with RSA method so that the necessary processes by using the decryption key that already exists. After the decryption process is complete we will get the actual text message.

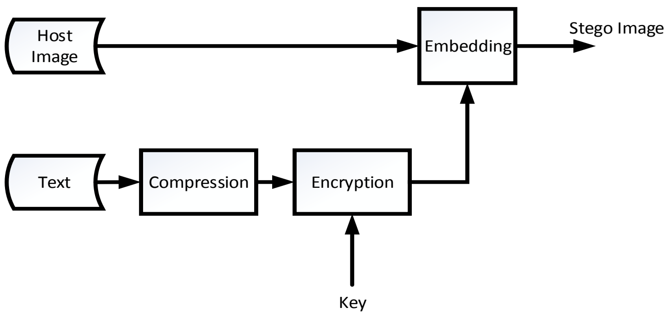

Figure 1. The process of embedding preceded by compression and encryption on text

Schematic diagram of the system proposed in this research proposal described by the scheme as shown in Fig 1. The scheme is effective to make the results of steganography become more secure and have a large capacity. Sequence of processes that occur from taking the text to be hidden is the compression process first, then a new encryption and data hiding is done. The sequence in this case is important, because the output of each process have different excesses when entering the next process is different. For example, if the sequence is converted into text-steganography-compression-

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encryption, the information carried compression process means not the text information, but the information related to the type of media cover or hosted on its stego image. If steganography used an image steganography, the type of cover that is a compressed image, as well as the encryption process. The encryption process is done on the output image compression process, not to a text content in terms of size lower than an image.

Based on Fig 2, LZW compression is used to reduce the size of the text is a more appropriate choice because of the effectiveness of compression will be better, rather than positioning the encryption process after the new text then the compression process [18]. After the compression process is the process of encryption, the encryption process encrypts data size becomes smaller than the original size and it is becoming one of the other advantages that cause a reduction in the processing time of encryption. After the encryption process, the next is the process of hiding the information encrypted on the cover image. At the receiver who will take back the hidden information, the process is the reverse process than there are at the sender as shown in Fig 2. Sequence of processes that occur is the data extraction process, followed by a data decryption process to restore data is encrypted, and the last new decompressed to restore the information to its original size.

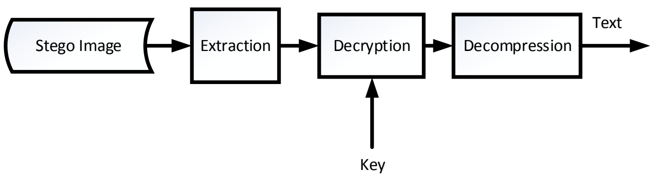

Figure 2. The process of embedding preceded by compression and encryption on text

IV. System Analysis

Data hiding capacity in this scheme of hiding method will have a formula floor(MxN/block^2)x3/8 characters. If block processing implemented in the simulation is 8x8 the formula will be floor(MxN/8^2)x3/8 characters. The capacity for data hiding will be as displayed in Fig 3. The capacity displaying in Fig 3 is amount of data as the encryption and compression result, thus the real data capacity for hiding will be larger than the capacity in Fig. 3.

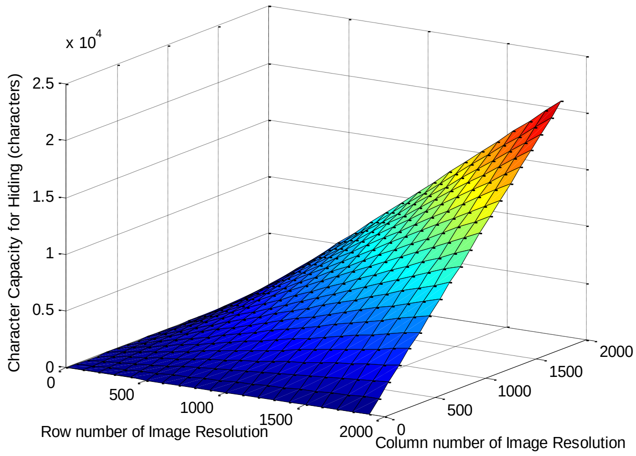

Figure 3 Image Resolution Impact for Data Hiding Capacity

Fig 4 displays the maximum integer value as outputs of the compression process. The higher the character number as inputs of compression, the higher the maximum integer value as compression outputs. There are 3 types of character input for comparison. The first type is random character input (red line color). The input character is randomly generated from 0-255 (ASCII number). The second type is repeated character input (blue line color) for 3 characters (e.g : ab ab ab ab .....). And the third type is normal text input (green line color) that we copy from the article and we process for compression. For first type the maximum integer value as compression output is also ups and down but in increasing trend. Normal type of input character tends to increase per groups of characters. And maximum integer value for repeated input characters is the same as normal type but keeps same integer maximum value in very long duration in each groups of characters. From Fig 4, also we can conclude that the integer value of output compression is more than 8 bits, depends on the input length. This case will certainly make an issue of encryption process, because in encryption process the bit length input will be limited to certain length of inputs bits which depends on p and q value of RSA.

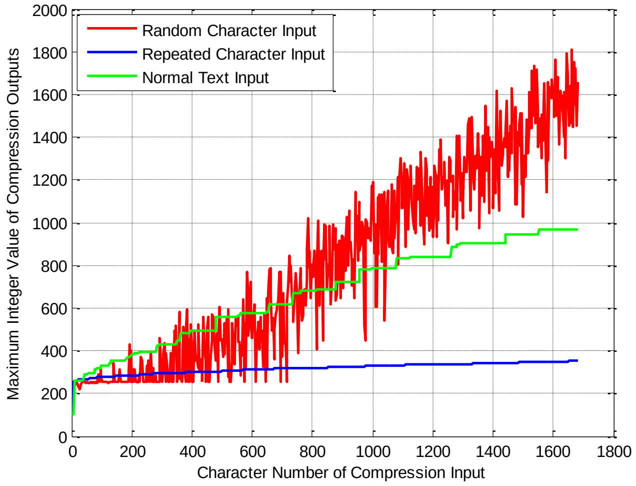

Figure 4. Effect of Input Number to Maximum Output Value of LZW Compression Process

The amount of character input before compression process will also affect to the compression output length as displayed in Fig 5. Random character input causes the length of output compression tends to be the same as the length of input compression. While for normal case, the length of output compression is about 50% of input length. It means the compression ratio for the normal character input is about 50%.

Besides the compression process, we must concern also about the encryption process. What is the maximum value of RSA encryption input and also what is the maximum value of RSA encryption output are the things that we must know for synchronization with compression and embedding process. In the appendix there is a table that explain about maximum integer value of input encryption value which depends on p and q RSA value. If the encryption input exceeds the maximum integer value in the table, then the decryption process will be failed. Because of this case, there are several things that need to be adjusted so that between the input and output of each process can be synchronous. These things include:

- Keep adjusting the output format of the compression process with input from the encryption process
- Need to adjust the amount or length of the output of the compression process with input from the encryption process
- Keep adjusting the output format of the encryption process with input from the process of embedding
- Need to adjust the amount or length of the output of the encryption process with input from the process of embedding

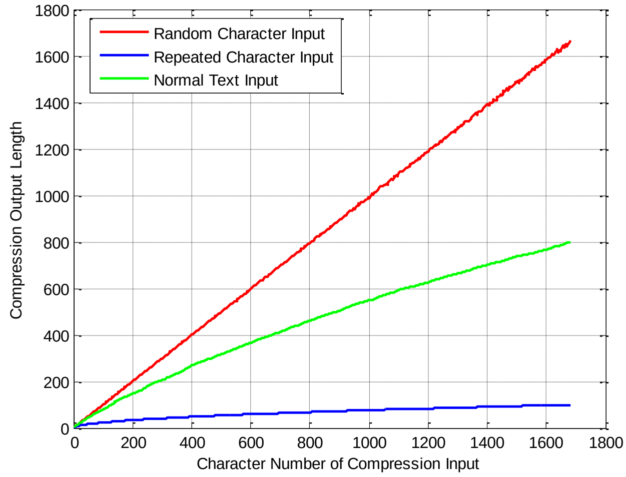

Figure 5. Effect of Input Number to Compression Output Length

Similarly, at the receiver, there should be an adjustment in contrast to the above points. Some of the problems that occur in addition to the above adjustment problems are associated with the following:

- Output compression process has the character length that is not fixed
- The value of p and q in the RSA encryption process must have a certain prime value
- The format of the data output from the encryption process is 16 bits unsigned integer
- The data will be hidden on the cover image should have a maximum length in accordance with the capacity of insertion of bits in the image

Those problems have an impact on the process of initial or preprocessing before data is processed on a block to next process. Solutions are being made to overcome the above problems, among others, as following :

- To address the problem that the output of the compression process has the character length is not fixed, it can be determined in advance how the maximum length of the text to be compressed by the compression ratio with a range of minimum-maximum, so that the Figs obtained compression results in a certain range that can be accommodated by the encryption process and embedding processes. So in this case there are special parameters that deal with setting a maximum character length in the beginning.
- The value of p and q in the RSA encryption process must have a certain value, then we must carry out several experiments to check the p and q values are suitable in the process of encryption. With the value of p and q, the encryption process can be performed on the data with a maximum length specified.

- To solve the problem of data output format of the encryption process which has 16 bits unsigned integer format, then it needs to do splitting process for 16 bit unsigned integer becoming two number with 8 bits unsigned integer before embedding process.

CONCLUSION

The simulation design of securing data using a Combination of LZW Compression, RSA Encryption, and DCT Steganography faces some synchronization problems between its blocks. There must be adjustment at input value of each blocks anticipating the failure of the algorithm in the next block. Some adjustments needed are there has to be maximum range of text length as compression inputs, p and q must be in certain range, and splitting the output number of encryption process before DCT embedding.

REFERENCES

[1] Rajyaguru, M. H., "Combination of Cryptography and Steganography With Rapidly Changing Keys", International Journal of Emerging Technology and Advanced Engineering, Vol.2, No.10, 2012, pp 329-332.

[2] Manoj, I. V. S., "Cryptography and Steganography". International Journal of Computer Applications (0975–8887), Vol.1, No.12, 2010,pp 63-68

[3] Sandeep Singh, Aman Singh. "A Review on the Various Recent Steganography", IJCSN International Journal of Computer Science and Network, Volume 2, Issue 6, December 2013

[4] E., Rahna dan V. K. Govindan. A Novel Technique for Secure, Lossless Steganography with Unlimited Payload. International Journal of Future Computer and Communication, Vol. 2, No. 6, Hal. 638-641. 2013

[5] Saleh Saraireh. "A Secure Data Communication System Using Cryptography And Steganography", International Journal of Computer Networks & Communications (IJCNC) Vol.5, No.3, May 2013

[6] Khalil Challita and Hikmat Farhat. "Combining Steganography and Cryptography: New Directions". International Journal on New Computer Architectures and Their Applications (IJNCAA) 1(1): 199-208. 2011

[7] Kanikar, Prashasti, Ratnesh N. Chaturvedi dan Vibhishek Kashyap. 2013. Image Steganography using DCT, DST, Haar and Walsh Transform. International Journal of Computer Applications, Vol. 65, No. 17, Hal. 34-37.

[8] Kekre, H. B., Sudeep D. Thepade dan Ratnesh N. Chaturvedi. 2013. Novel Transformed Block Based Information Hiding Using Cosine, Sine, Hartley, Walsh and Haar Transforms. International Journal of Advances in Engineering &Technology, Vol. 6, No. 1, Hal. 274-281.

[9] Singh, Dr. Ajit dan Meenakshi Gahlawat. 2013. Secure Data Transmission using Watermarking and Image Compression. International Journal of Advanced Research in Computer Engineering & Technology, Vol. 2, No.5, Hal. 1709-1715.

[10] Katzenbeisser, S. and Petitcolas, F.A.P. 2000, Information Hiding Techniques for Steganography and Digital Watermarking. Artech House, Inc., Boston, London.

[11] T. Morkel, J. Eloff, and M. Olivier, "An overview of image steganography", In Proceedings of the Fifth Annual Information Security South Africa Conference (ISSA2005), (Sandton, South Africa, Jun/Jul. 2005).

[12] M. Kharrazi06, H. Sencar, and N. Memon, "Performance study of common image steganography and steganalysis techniques," Communications of the SPIE and IS&T, 15, No.4, pp. 1017-9909, Oct-Dec., 2006.

[13] Nedal M.S. Kafari, Hani Y. Suleiman, "Bit-4 of Frequency Domain DCT Steganography Technique", First National Conference On Networked Digital Technologies, PP. 286-291, 2009.

[14] Håstad, Johan (1986). "On using RSA with Low Exponent in a Public Key Network". Advances in Cryptology — CRYPTO '85 Proceedings. Lecture Notes in Computer Science 218. pp. 403–408. doi:10.1007/3-540-39799-X_29.

[15] Singh, Dr. Ajit dan Meenakshi Gahlawat. 2013. "Image Compression and its Various Techniques". International Journal of Advanced Research in Computer Science and Software Engineering, Volume 3, Issue 6, June 2013, pp. 650-654

[16] Khalid Sayood, Introduction to Data Compression, Academic Press, 2000.

[17] D. Salomon. Data Compression: The Complete Reference. Springer, 1998

[18] Budiman, Gelar, and Novamizanti, Ledya. "White Space Steganography On Text By Using LZW – Huffman Double Compression", International Journal of Computer & Communication, Vol. 7, No. 2, March 2015.

APPENDIX

TABLE I. MAXIMUM INTEGER VALUE OF RSA ENCRYPTION INPUT

ID	p	q	Maximum Integer Value of RSA Encryption Input	ID	p	q	Maximum Integer Value of RSA Encryption Input	ID	p	q	Maximum Integer Value of RSA Encryption Input	ID	p	q	Maximum Integer Value of RSA Encryption Input
1	2	3	6	15	5	2	10	29	11	2	22	43	17	2	34
2	2	5	10	16	5	3	15	30	11	3	33	44	17	3	51
3	2	7	14	17	5	7	35	31	11	5	55	45	17	5	85
4	2	11	22	18	5	11	55	32	11	7	77	46	17	7	119
5	2	13	26	19	5	13	65	33	11	13	143	47	17	11	187
6	2	17	34	20	5	17	85	34	11	17	187	48	17	13	221
7	2	19	38	21	5	19	95	35	11	19	209	49	17	19	323
8	3	2	6	22	7	2	14	36	13	2	26	50	19	2	38
9	3	5	15	23	7	3	21	37	13	3	39	51	19	3	57
10	3	7	21	24	7	5	35	38	13	5	65	52	19	5	95
11	3	11	33	25	7	11	77	39	13	7	91	53	19	7	133
12	3	13	39	26	7	13	91	40	13	11	143	54	19	11	209
13	3	17	51	27	7	17	119	41	13	17	221	55	19	13	247
14	3	19	57	28	7	19	133	42	13	19	247	56	19	17	323

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Compact Antipodal Vivaldi Printed Antenna for Ultra Wideband Application

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Abstract—**In this paper, a design and implementation of compact antipodal Vivaldi printed antenna is proposed for ultra wideband (UWB) application. The antenna which is intended to work at frequency range of 1.5–3.5GHz is designed on an FR4 Epoxy dielectric substrate with the thickness of 1.6mm. In the design process, to have a compact antipodal Vivaldi printed antenna, the investigation is focused on its physical dimension with the reflection coefficient used as a key performance indicator. Two designs of antenna, i.e. single antenna and array of two antennas, are presented and then developed for the comparison. From the result, it shows that the realized single antenna has -10dB working bandwidth of 800MHz range from 1.6–2.4GHz. Meanwhile for the realized array antenna, it has -10dB working bandwidth of 1900MHz range from 1.6–3.5GHz.**

Keywords-antenna array; antipodal; ultra wideband; Vivaldi antenna.

I. INTRODUCTION

Recently, application of wideband is very large such as for military and commercial usage [1]. But not only on those two cases, other field such as medical, geology, archeology, radar, and many others also using a wideband systems [2]. One of the most critical parts of wideband system is the antenna. Currently, the demand for a small and compact antenna is dictated by the mobility requirements such as portable surface penetrating radar (SPR).

As is already, the SPR is a radar system usually used for object detection beyond the surface–mostly wall [3]. The SPR usually used in military or archeological field. SPR ways of work is similar to Ground Penetrating Radar (GPR). The device is processing the echo signal which is reflected by the object to get information from the object [4]. The information obtained can be varied, such as position and shape of the object, up to type of material that compose the object. In fact, SPR needs wideband systems to overcome the trade-off of resolution and signal depth of penetration [5]. Therefore a compact printed antenna with wideband response for such application is required.

There have been a great number of researches related to ultra wide band antenna, such as Vivaldi antenna, Bow-tie antenna, spiral antenna, and many potential modifications which can improve the characteristics of antenna [6]-[8]. In this paper, the design of antipodal Vivaldi printed antenna is proposed. Vivaldi antenna is chosen due to its ease in design as well as in fabrication. The second antenna that proposed in this letter is an antipodal Vivaldi array of two single antennas. The array of two single antennas is proposed to get higher gain than single antenna.

II. ANTIPODAL VIVALDI ANTENNA DESIGN

A. Design of Single Antenna

The initial design of antipodal Vivaldi printed antenna is using two ellipses as its exponential curve. The equation for each dimension is based on equations in [6]. The dimension of the initial design is 200.5mm x 182.6mm. It is approximately 0.67λ x 0.61λ. Through parametrical studies, a compact antipodal Vivaldi printed antenna is investigated. The design of antipodal Vivaldi printed antenna is shown in Fig. 1, and the value of each dimension is summarized in Table 1.

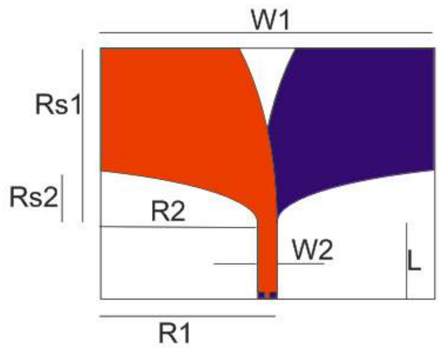


Figure 1. Design of a compact antipodal Vivaldi printed antenna

TABLE I. VALUE OF ANTENNA DIMENSION

Dimension	Value (mm)
W1	99.579
W2	5.751
R1	52.665
R2	46.914
Rs1	52.123
Rs2	15.614
L	22.877

The antenna dimension is 75mm x 99.6mm. This is approximately 0.25λ x 0.33λ, where λ is the wavelength of 1GHz. The antenna is fed from an SMA connector through a microstrip line. This microstrip line width is set 5.8 mm and its length is 22.9mm according to the results of parametrical study. This antenna is deployed on FR-4 Epoxy dielectric substrate with the thickness of 1.6mm and relative permittivity of 4.4. This proposed design reducing almost 80% area from initial design. The simulation results of reflection coefficient for of 2 designed antennas, i.e. initial design and proposed design, are plotted in Fig. 2. The simulation shows that the proposed design extends the low-end frequency more than 400MHz. Although the high-end frequency is much lower than the initial design, the proposed design still meets the specification.

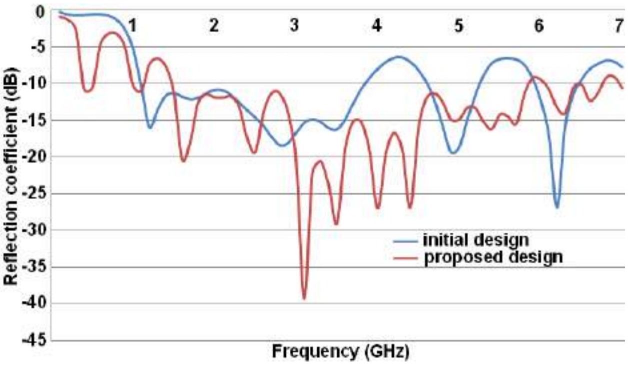


Figure 2. Simulation result of initial design and proposed design

B. *Design of Two Antennas Array*

Array of two compact antipodal Vivaldi printed antennas is linked together using T-junction. The figure of antenna array is shown in Fig. 3. SMA connector feeds a 50Ω microstrip line and then split into two 100Ω microstrip lines. Using a quarter-wave transformer, 100Ω microstrip line is transformed into 50Ω microstrip line that feeds each antenna. The width of each microstrip lines and a quarter-wave transformer is based on equations in [6]. The spacing between two antennas is set to be 35mm according to the result of parametrical study. The simulation result of this proposed array design is shown in Fig. 4. It shows that the bandwidth of antenna become smaller. The low-end frequency is rising from 1.1GHz to 1.7GHz. However, in general the return loss from 1.7GHz – 3.5GHz is deeper than the single antenna.

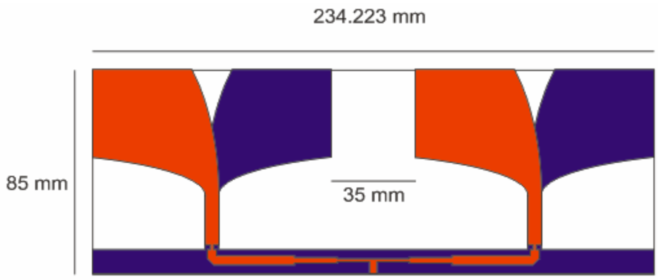


Figure 3. Design of two compact antipodal Vivaldi printed antenna array

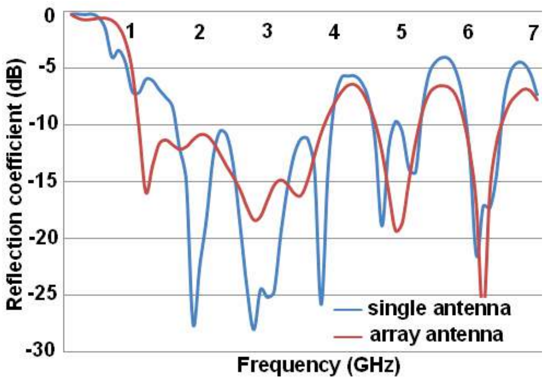
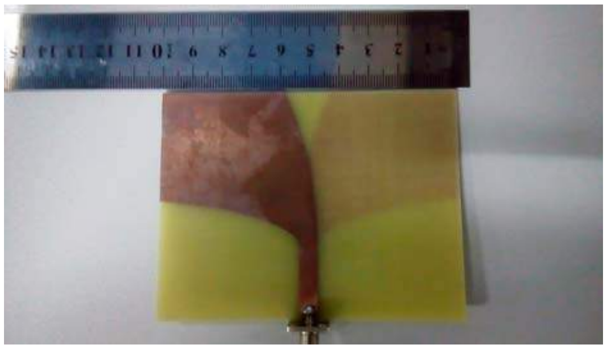


Figure 4. Simulation result of single antenna and array of two antennas

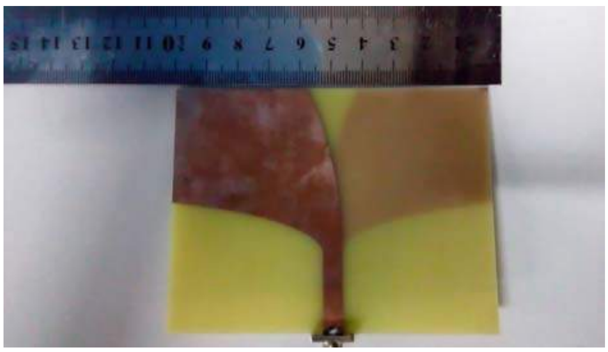
III. Hardware Realization and Measurement

A. *Realization of Single Antenna*

The design of compact antipodal Vivaldi antenna that was proposed is realized on 1.6mm FR4 Epoxy dielectric substrate through wet etching technique. The picture of realized single printed antenna is shown in Fig. 5. The reflection coefficient of realized antenna is measured using network analyzer and the measured result is plotted in Fig. 6. Unfortunately, the measurement result is very poor. The reflection coefficient is higher than -10dB for frequency of 1.6– 2.4GHz. The reflection coefficient also has higher value in any given frequency between 1.1–3.5GHz.



(a) front view of realized single antipodal Vivaldi printed antenna



(b) back view of realized single antipodal Vivaldi printed antenna

Figure 5. Picture of realized single antipodal Vivaldi printed antenna

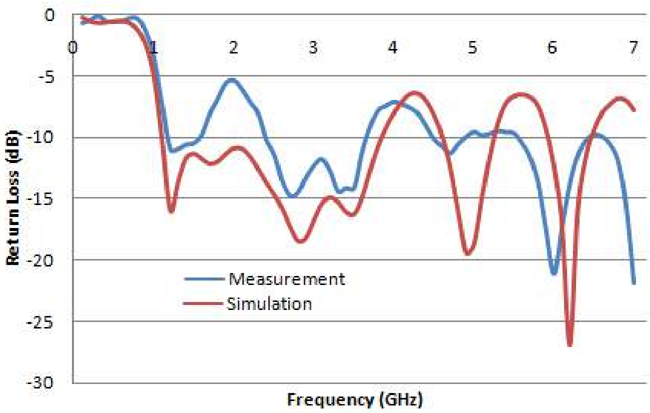
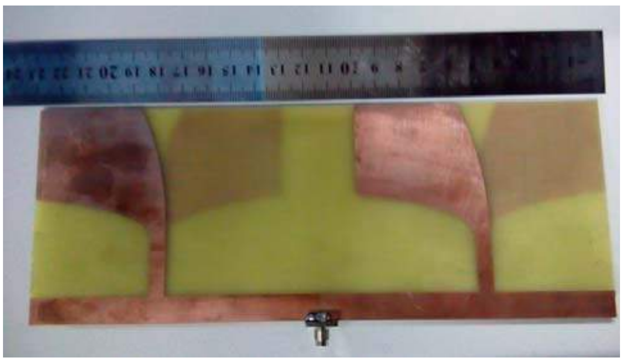


Figure 6. Comparison of reflection coefficient between simulation and measurement for single antenna

B. *Realization of Single Antenna*

The second antipodal Vivaldi printed antenna that was proposed in this paper is the array of two single antennas. The realized printed antenna is shown in Fig.7. The reflection coefficient measurement is depicted in Fig. 8. The graph shows that the -10dB working bandwidth antenna become smaller. The high-end frequency is about 500MHz lower than the simulation. The result also shows that almost in any given frequency from 1.6–3.5 GHz the measurement result is higher than the simulation result.

(a) front view of realized two antipodal Vivaldi printed array antenna



(b) back view of realized two antipodal Vivaldi printed array antenna

Figure 7. Picture of realized two antipodal Vivaldi printed array antenna

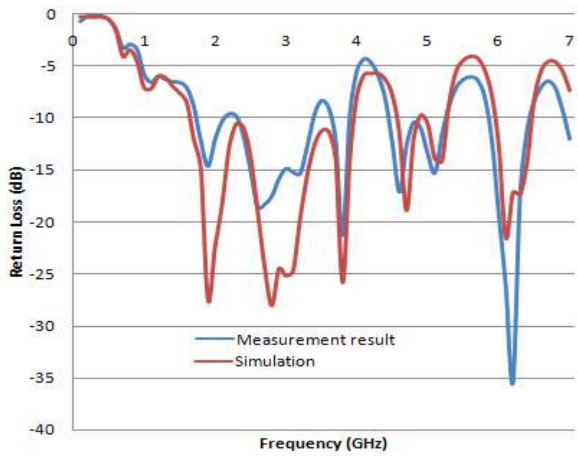


Figure 8. Comparison of reflection coefficient between simulation and measurement for two antennas array

IV. Conclusion

Simulation result and measurement result of two antenna design is presented. The single antenna design that proposed can reduce the area almost 80% area of initial design. The single antenna design also extends antenna low-end frequency more than 400MHz. The simulation result of antenna array has a smaller return loss almost in any given frequency from 1.6–3.9GHz. The smaller bandwidth of antenna array was caused by the use of T-junction to connect two single antennas due to T-junction usually used in narrowband. It might get better if we use power splitter such as Wilkinson power-divider to connect two or more UWB antenna, because Wilkinson power-divider is commonly used in UWB antenna array. Although the simulation shows a promising result, the measurement of two designs that proposed in this letter is not showing good result. The measurement result shows higher return loss than simulation result at almost any given frequency. There are so many causes such as the soldering which is not well attached to the patch. This also can alter the matching as well as the connection of SMA connector.

Acknowledgment

The authors wish to thank Mr. Zenal Aripin from School of Electrical Engineering and Informatics, Institut Teknologi Bandung (ITB) for supporting in the experimental preparation and characterization.

References

[1] R. Yogesh, *Antennas for Ultra-Wideband Systems*, LAP Lambert Academic Publishing, 2014.

[2] X. Li, *Body Matched Antennas for Microwave Medical Applications: Volume 72*, KIT Scientific Publishing, 2014.

[3] H. M. Jol, R. J. Dechaine, and R. Eisenman, "Archeological GPR Investigation at Rennes-le-Chateau, France," in *Proceeding of 9th International Conference on Ground Penetrating Radar (GPR)*, Santa Barbara. 2002; pp. 91–95.

[4] D. J. Daniel, *Ground Penetrating Radar*. 2nd ed.. London: IEE Radar, Sonar, Navigation and Avionics Series. 2004.

[5] R.B.V.B. Simorangkir and A. Munir, "Numerical design of ultra-wideband printed antenna for surface penetrating radar," *TELKOMNIKA*

(Telecommunication, Computing, Electronics and Control), vol. 9, no. 2, pp. 341-350, Aug. 2011.

[6] D.M.Pozar, *Microwave Engineering 3rd Edition*, Massachusetts, John Wiley & Sons, Inc, 2005.

[7] K. L. Shlager, G. S. Smith, and J. G. Maloney, “Optimization of bow-tie antennas for pulse radiation*,” IEEE Trans. Antennas Propag.***.** Vol. 42, No. 7, pp. 975–982, 1994.

[8] A.Z. Hood, T. Karacolak, and E. Topsakal, “A Small Antipodal Vivaldi Antenna for Ultrawide-band Applications,” *IEEE Antenna and Wireless Propagation Letter*, Vol. 7, 2008.

[9] P. Fei, Y.C. Jiao, W. Hu, and F.S. Zhang, “A Miniaturized Antipodal Vivaldi Antenna with Improved Radiation Characteristic,” *IEEE Antenna and Wireless Propagation Letter*, Vol. 10, 2011.

Impedance Matching Circuit for 30-88MHz Monopole Antenna with Automatic Matching

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***Abstract*—This paper presents an impedance matching circuit for monopole antenna in the frequency range 30-88MHz which is integrated by microcontroller with tuning algorithm as an automation component. An impedance matching type L is used for the circuit, while for the realization some trimmer capacitors and SMD inductors are employed. To get an input voltage for the microcontroller, some supporting circuits are designed including VSWR meter, comparator and amplifier with 2 levels. It shows that the circuit is successfully tested by replacing values of load impedance with $Z_A = 75 + 14.6952j$ and $Z_B = 75 + 25.6224j$, and tuning frequency values in frequency range of 41-45MHz. When impedance matching circuit is active, VSWR values for load impedance $Z_A = 75 + 14.6952j$ in the frequency 41MHz, 43MHz, and 45MHz are 1.34, 1.35 and 1.5, respectively and for load impedance $Z_B = 75 + 25.6224j$ are 1.37, 1.373 and 1.5.**

Keywords*—*impedance matching, VSWR, microcontroller.

I. INTRODUCTION

Information has become an important thing in human life. Delivering information from point to point should be done with more effectively, and commonly radio waves are used with different type of frequency range as a media. As a system, delivering information, at least a radio transmitter and receiver is required. In the radio transmitter and receiver, maximum power transfer is one of the most important things in transmission system. Maximum power transfer can be reached if there is an impedance match between source and load. Thus, impedance matching circuits are needed for balancing impedance between source and load [1]-[2]. Impedance matching is an essential part of an RF system that is used to maximize the power transfer from a generator to a load and to minimize the reflections from the load [3].

Impedance matching is an attempt to adjust the antenna impedance to the impedance characteristics of the source / channel. In order to transfer the energy from the transmitter to the antenna can be efficiently (no wasted energy or no reflected power), the antenna impedance is should match [3]. The worst effect of the impedance which is not the same is the emergence of reflectance (reflected power) from the antenna. Power reflected back to the transmitter will damage the transmitter circuit. To avoid damage, it is necessary to make impedance matching. The most ideal VSWR value is 1. The better VSWR value (closer to 1) then the signal will be passed to the antenna will be more optimal. Rate voltage standing wave ratio (VSWR) indicating how well the impedance matching is done. VSWR shows that the reflected signal is greater than the signal emitted by the antenna. During this time, antenna is designed as a transmitting and receiving antenna has specific frequency range and used for different purposes.

One of them is an antenna with working frequency in the VHF band (30MHz - 300MHz) which is used in materials research and also in the military field [4]-[5]. Based on that, it appears the idea to design an antenna with working frequency in the VHF band (30MHz - 300MHz). That requires impedance matching circuit that can be applied to the simple antenna that can be used in a specific frequency range to obtain optimal value to the transmission system [6].

II. DESIGN AND SIMULATION

A. System Description

The overall system consists of input voltage unit, user interface, controller unit, switching unit and main circuit. The block diagram of system is shown in Fig. 1

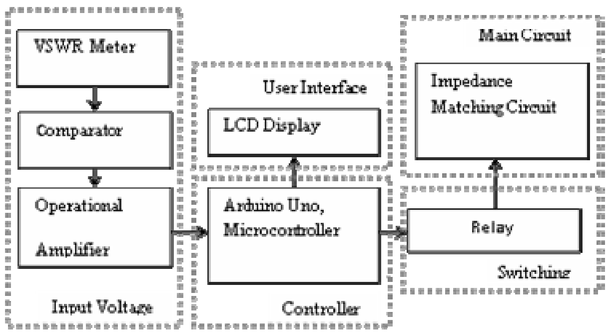

Fig. 1. Block diagram of overall system of automatic impedance matching

The user assigns load impedance and frequency value and then the VSWR meter will detect voltage value at source and load, and transform AC voltage to DC voltage. Comparator will give voltage difference between voltage value at source and load. Voltage will be strengthened 2 levels by operational amplifier (Ope-Amp) to be used as the input voltage for microcontroller. Then an Arduino Uno–based microcontroller gives an order to display the VSWR value on LCD display and also will choose which one is the right impedance matching circuit by using relay.

B. Hardware System Design

There are 3 main parts in the realization of this work: VSWR meter circuit, comparator and Op-Amp, impedance matching circuit based on VSWR value that has been determined. As shown in Fig. 2, initial voltage value is obtained using VSWR meter by taking voltage value at the point of source and load. By using diodes 1N60 and capacitor, AC voltage will be rectified and transformed into a DC voltage to be processed in the next stage.

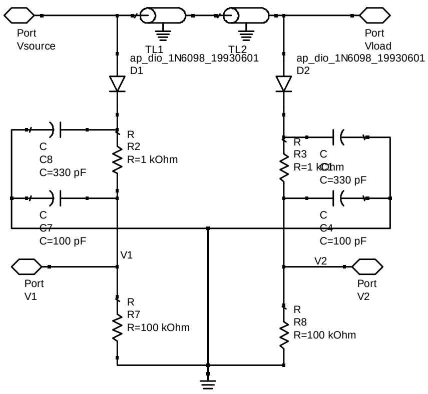

Fig. 2. VSWR meter circuit

Based on the simulation results, the voltage characteristic is successfully obtained in millivolts (mV), so that the voltage can be used as an input microcontroller. Here, a comparator circuit is used to get voltage difference between source and load produced by VSWR meter circuit as shown in Fig. 3. The output voltage is then strengthened using Op-Amp to get into microcontroller input voltage, that is 0–5Volt.

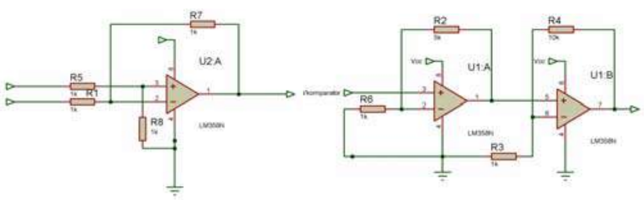

Fig. 3. Comparator and operational amplifier 2 levels

To get voltage level 0-5Volt, we need amplifier circuit. As previously known, that the output voltage of the comparator circuit obtained in millivolts at intervals of 10-50 mV, thus the minimum strengthening until 10x is required to get the voltage is in the interval 0-5Volt. In this work, the amplifier circuit designed with multilevel non-inverting amplifier with 50x voltage strengthening. For comparator and amplifier circuit, an integrated circuit (IC) LM358N is employed as it is suitable with specifications.

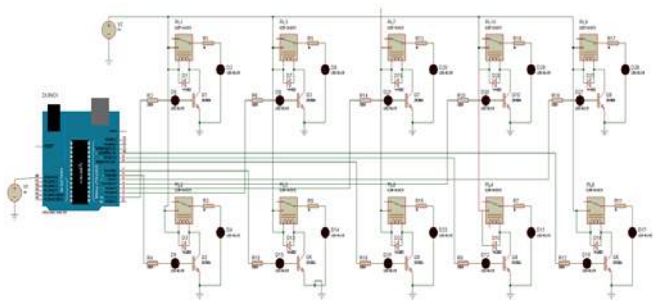

Fig. 4. Switching components

After the voltage is obtained, it can be used as input microcontroller. This voltage can be translated into algorithm as a command to select the appropriate circuit with the help of relay as a switching component as shown in Fig. 4. Impedance matching circuit is used for monopole antenna with a frequency range between 30MHz and 88MHz. This circuit using impedance matching circuit type L with inductor and capacitor arranged in parallel as illustrated in Fig. 5.

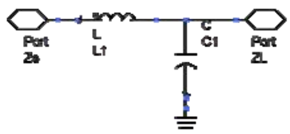

Fig. 5. Impedance matching circuit

The impedance matching circuit is simulated at frequency of 60MHz and refers to the range VSWR value that has been pre-determined. Reference VSWR values are presented in Table 1. To know the circuit has been working well, an LCD display as shown in Fig. 6 is used to shows the VSWR value which can be formulated based on the voltage difference value.

TABLE I. VSWR DATA FOR IMC

VSWR	Zs (Rs+jXs (Ω))	R_L (Ω)	L (nH)	X_L (Ω)	Z_L (R_L+jX_L(Ω))
1.5	50	75	0	0	75
1.532435	50	75	22	8.2896	75+8.2896j
1.598774	50	75	39	14.6952	75+14.6952j
1.779909	50	75	68	25.6224	75+25.6224j
2.059789	50	75	100	37.68	75+37.68j
3.067344	50	75	180	67.824	75+67.824j
3.731116	50	75	220	82.896	75+82.896j
6.126489	50	75	330	124.344	75+124.344j
10.43429	50	75	470	177.096	75+177.096j
13.96824	50	75	560	211.008	75+211.008j

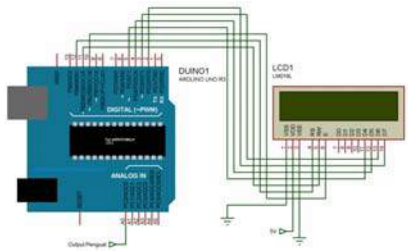

Fig. 6. LCD display to show VSWR value

C. Software System Design

Design software are related to the creation of algorithm that will be implanted into the microcontroller, in this case is arduino uno. The program contains instructions to be processed by the microcontroller to handle all the inputs and outputs in the system when it is being used. The working principle of algorithm is illustrated in a flowchart shown in Fig. 7.

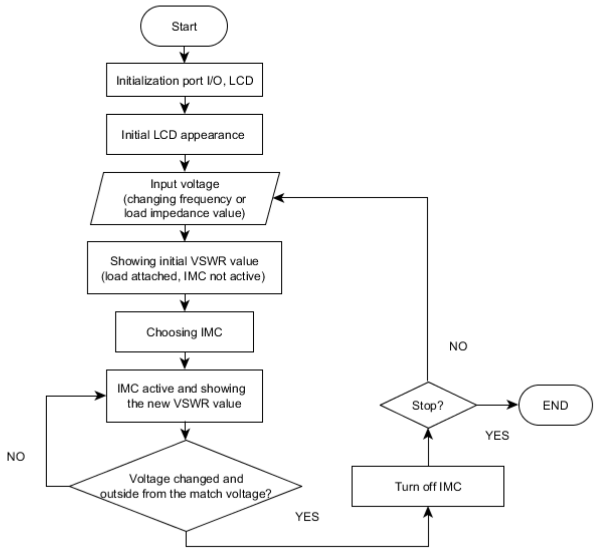

Fig. 7. Flowchart diagram of software design

III. Result and Discussion

This circuit is a combination of several circuits that have previously simulated, consisting of VSWR meter circuit, comparator circuit, amplifier circuit, microcontroller circuit and relays as switching devices, impedance matching circuit and LCD to display the value of VSWR as an indicator. The prototype of realized automatic impedance matching circuit is shown in Fig. 8.

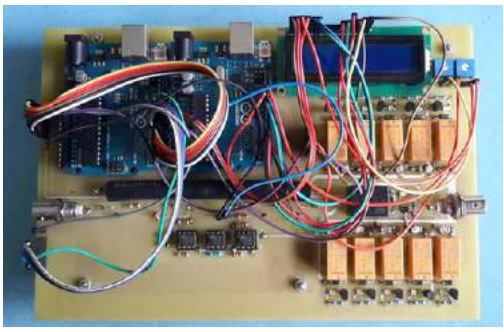

Fig. 8. Realization of the automatic impedance matching circuit

In principle, impedance matching circuit is used to provide the conditions for maximum power transfer for the transmission system. This is indicated by the value of power transferred by the source will be equal to the value of the power received by the load, this will happen when there is impedance match between them. Therefore, the value of the voltage difference between source and load can be used as an indicator of the maximum power transfer, as well as VSWR. The smaller value of voltage difference between the source and load indicates the presence of maximum power transfer conditions. In other words, the value of the voltage at the load will equal to the value of voltage transferred by the source.

To know the impedance matching circuit can work well, things to do is comparing value of the voltage difference when the load is used with the condition before and after the impedance matching circuit becomes active. By using a signal generator with a voltage source 120dBuV, this circuit has been successfully tested by replacing values of the load impedance with Z_A = 75 + 14.6952j and Z_B = 75 + 25.6224j and tuning frequency values in the frequency range 41- 45MHz. For the load impedance of Z_A = 75 + 14.6952j and Z_B = 75 + 25.6224j, the measured results are plotted in Figs. 9 and 10, respectively.

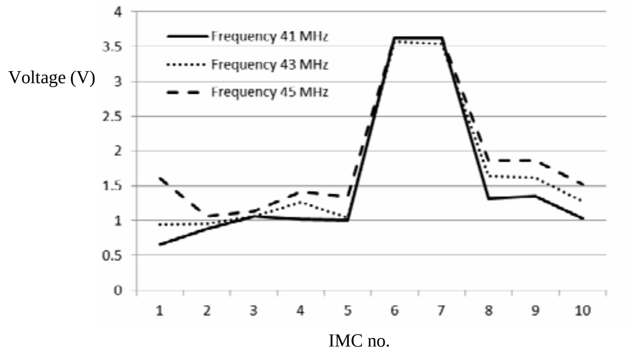

Fig. 9. Voltage difference for each IMC with Z_A = 75 + 14.6952j at frequency 41MHz, 43MHz, and 45MHz

Based on the voltage data, some conclusions for the characterization can be inferred. For the load impedance of Z_A = 75 + 14.6952j, when impedance matching circuit is active, voltage difference value in the frequency of 41MHz, 43MHz, and 45MHz are 1.02V, 1.04V and 1.54V, respectively. Meanwhile, for the load impedance of Z_B = 75 + 25.6224j, when impedance matching circuit is active, voltage difference value in the frequency of 41MHz, 43MHz, and 45MHz are 1.11V, 1.12 V and 1.52V, respectively.

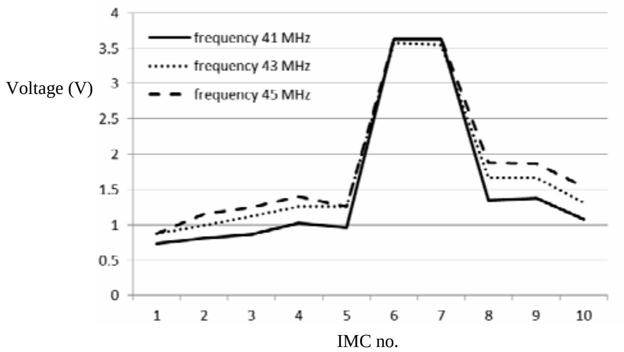

Fig. 10. Voltage difference for each IMC with Z_B = 75 + 25.6224j at frequency 41MHz, 43MHz, and 45MHz

If it is associated between voltage difference value and VSWR value, it can be calculated as follows.

$$|\Gamma| = \frac{Z_L - Z_s}{Z_L + Z_s};\ Z = \frac{V}{I}\ ;\ |\Gamma| = \frac{V_L - V_s}{V_L + V_s};\ \text{with VSWR value} = \frac{1+|\Gamma|}{1-|\Gamma|}$$

Based on these equations, the value of voltage difference that obtained earlier to VSWR values can be changed and it is summarized in Table 2. From the VSWR value, it shows that the circuit can work well for impedance value of Z_A = 75 + 14.6952j and Z_B = 75 + 25.6224j in the frequency range of 41-45MHz.

TABLE II. CHARACTERIZATION RESULT FOR DIFFERENT LOADS

Impedance load value	Frequency (MHz)	IMC no.	VSWR
Z_A = 75+14.6952j	41	4	1.34
	43	5	1.35
	45	10	1.5
Z_B= 75+25.6224j	41	4	1.37
	43	3	1.373
	45	5	1.5

IV. CONCLUSION

Impedance matching is an essential part of an RF system that is used to maximize the power transfer from a generator to a load and to minimize the reflections from the load. Automatic impedance matching circuit was designed using combination of several circuits; there are VSWR meter, comparator, microcontroller and amplifier with 2 levels. This circuit was implemented with PCB dimensions 25cm x 15cm and PCB designed by using Epoxy. This circuit has been successfully characterized by replacing values of the load impedance with Z_A = 75 + 14.6952j and Z_B = 75 + 25.6224j and tuning frequency values in the frequency range 41-45 MHz. When the impedance matching circuit is active, VSWR values for load impedance of Z_A = 75 + 14.6952j in the frequency 41MHz, 43MHz, and 45MHz are 1.34, 1.35 and 1.5, respectively. Meanwhile for load impedance of Z_B = 75 + 25.6224j are 1.37, 1.373 and 1.5, respectively.

REFERENCES

[1] D. M. Pozar, *Microwave Engineering 2nd Edition*, John Wiley & Sons, 1998.

[2] M. F. Iskander, *Electromagnetic Fields and Waves*, Waveland Press, 1992

[3] D. Misra, *Radio-Frequency and Microwave Communication Circuits: Analysis and Design*, New Jersey: John Wiley & Sons, Inc, 2004.

[4] M. Rolan and H. Witschnig, "Automatic Impedance Matching for 13.56 MHz NFC Antennas", *IEEE Communication Systems, Networks and Digital Signal Processing*, 2009.

[5] V. Parro and M. Pait, "Design of an Automatic Impedan.ce Matching System", *IEEE Microwave and Optoelectronics*, 2010.

[6] Robson N. Lima, "Étude d'un Système d'adaptation Automatique d'impédance RF et d'un Circulateur Actif RF en Technologie MMIC", Tese de Doutorado, ENST-Paris, 2001.

2-Stage Microstrip Combline BPF with Capacitor Chip Incorporation

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***Abstract*—In this paper, the design and implementation of 2-stage microstrip combline bandpass filter (BPF) is proposed with capacitor chip incorporation. The initial design of filter is taken from the existing one with some improvements in the performances to be suitable for the 2.4GHz application. The proposed filter is designed on a 0.8mm thick FR4 Epoxy dielectric substrate with the dimension of 8.9mm x 32mm. Two tuning capacitor chips of 2pF are incorporated into resonators of filter and another capacitor chip is inserted into inter-resonator to connect each resonator of filter. After obtaining an optimum design, a prototype of 2-stage microstrip combline BPF is realized through wet etching technique. From characterization result, it shows that the realized filter has center frequency of 2.375GHz with -3dB working bandwidth of 170MHz. The measured bandwidth is 30MHz narrower than the simulated one which has center frequency and -3dB working bandwidth of 2.4GHz and 200MHz, respectively.**

Keywords-2-stage microstrip combline BPF; -3dB working bandwidth; capacitor chip incorporation; center frequency.

I. INTRODUCTION

Nowadays the need of information is the most interest stuff for the modern society. Telecommunication technologies play a significant role as a one fundamental support especially for handset communication [1]. In wireless technology, transmitted electromagnetic signal have so many losses such as path loss, multipath, and noise [2]-[3]. In its application, wireless technology have filter as signal receiver and select the certain frequency as operation frequency [4]. To answer the challenges for better quality mentioned before, recently the research for robust filter became an interesting research chosen for any telecommunication technology [5].

As is already known, filter is a device that is used to pass a certain frequency by allowing the desired frequency signal in passband and block the unwanted signal in stopband. Basically, filter has an essential factor as a signal processing device in telecommunications. Whilst, bandpass filter (BPF) plays a significant role in wireless communication systems. Transmitted and received electromagnetic signals have to be filtered at a certain center frequency with a specific bandwidth. There are numerous techniques to create filters, such as using strip line and microstrip [6]-[7]. The latter mentioned, i.e. microstrip, is a technology to design a microwave filter at a high frequency (> 1GHz) which usually consists of patch, substrate and groundplane.

In this paper, a 2-stage microstrip combline BPF is designed and implemented with capacitor chip incorporation. The proposed microstrip BPF which is taken from the initial design in [8] is deployed on a grounded 0.8mm thick FR4 Epoxy dielectric substrate to have center frequency of 2.4GHz applicable for wireless communication system. In the design process, the performance of microstrip combline BPF is investigated intensively to achieve the optimum design of overall structure. Some parametrical studies upon the characteristic of filter are carried out to obtain the optimum design including the optimum value of capacitor chips. Meanwhile, the filter parameters such as return loss, insertion loss, and bandwidth response will be used as performance indicators in the evaluation of microstrip combline BPF characteristic.

II. DESIGN OF MICROSTRIP COMBLINE BPF

A. Initial Microstrip Combline BPF

The initial microstrip combline BPF is shown on Fig. 1. This filter is deployed on a Taconic-RF30 dielectric substrate with overall size and thickness of 12mm x 8mm and 0.75mm, respectively. A pair of coupled combline resonators with capacitors loaded is implemented in the structure of initial microstrip combline BPF. The capacitor at one end of each resonator is for frequency tuning whilst the capacitor across the two resonators is for controlling the inter-resonator coupling as shown in Fig. 2.

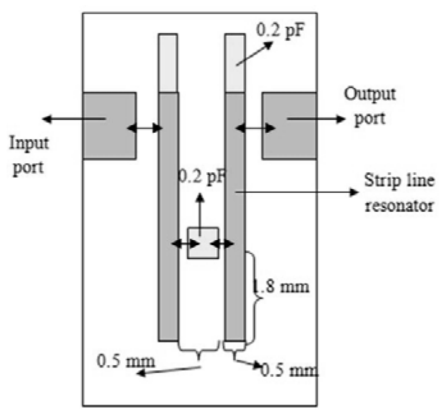

Figure 1. Structure of initial microstrip combline BPF

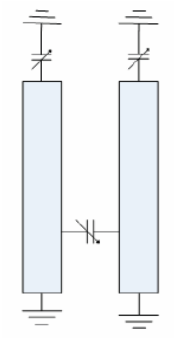

Figure 2. Schematic circuit for coupled combline resonators

The simulation is carried out to investigate the coupling characteristics based on the microstrip coupled combline resonators, where two probes associated with input and output ports are arranged so as to weakly excite the coupled resonators. For the simulations, the capacitor for controlling frequency has capacitance of 0.2pF. The spacing between the coupled resonators is fixed at 1.1mm. For electronically controlling the coupling, two capacitors are used in series for easy dc bias arrangement. Fig. 3 shows the characterization result of initial design of microstrip combline BPF.

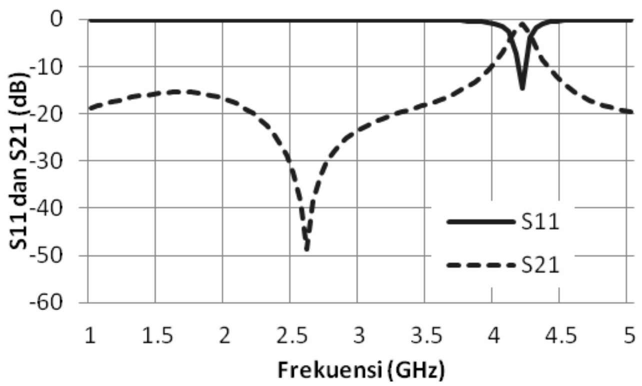

Figure 3. Characterization result of initial microstrip combline BPF

Table 1 summarizes the detail geometry of each filter element. The elements are made of metal copper on the top side of dielectric substrate as well as the ground plane on the bottom side. In order to have an accurate design, the losses of dielectric substrate and metal copper conductive for elements and ground plane are taken into account. The input/output signals for initial microstrip combline BPF are obtained from SMA connectors.

TABLE I. GEOMETRY OF EACH ELEMENT FOR INITIAL MICROSTRIP COMBLINE BPF

Element	Width (mm)	Length (mm)
Input Port	1.9	2.62
Output Port	1.9	2.62
Resonan 1	0.5	9.1
Resonan 2	0.5	9.1

B. Proposed 2-Stage Microstrip Combline BPF

Fig. 4 shows the structure of proposed 2-stage microstrip combline BPF modified from the initial design. During the process to find the best frequency response of insertion loss and return loss for proposed filter, several parametrical studiies are carried out intensively. In additions, study about changing or modifying BPF dimension such as size of the input and output port, length of resonator strip, changing the vertical position of inter-resonator capacitor and the value of capacitance of each capacitor chip is also attempted.

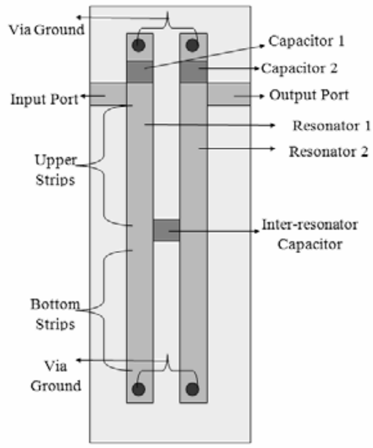

Figure 4. Structure of proposed 2-stage microstrip combline BPF

The proposed 2-stage microstrip combline BPF detailed dimensions are 1.3mm x 2.95mm for input and output port, 1.0mm x 25.2mm for both resonator strip, 8.9mm x 32mm board of substrate and groundplane. Two capacitor chips of 2pF are incorporated for tuning capacitors and another capacitor chip of 1pF is for inter-resonator capacitor. The gap between two resonators is 1mm, and the length of reach resonator is 12.6mm. Fig. 5 plots the characterization result of proposed 2-stage microstrip combline BPF. It shows that the proposed filter has a center frequency of 2.4GHz with the value of reflection coefficient of -16dB at center frequency and -3dB working bandwidth of 200MHz.

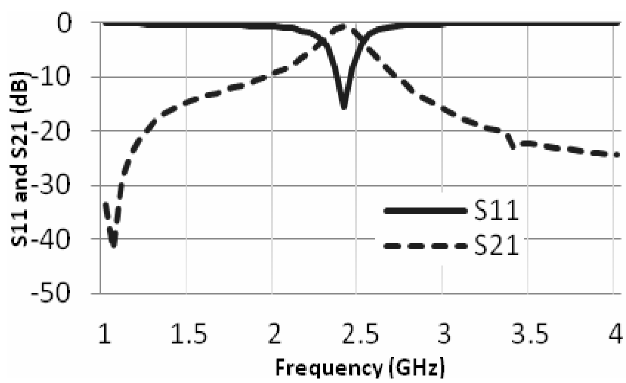

Figure 5. Characterization result of proposed 2-stage microstrip combline BPF with capacitor chip incorporation

III. PROTOTYPING AND EXPERIMENTAL CHARACTERIZATION

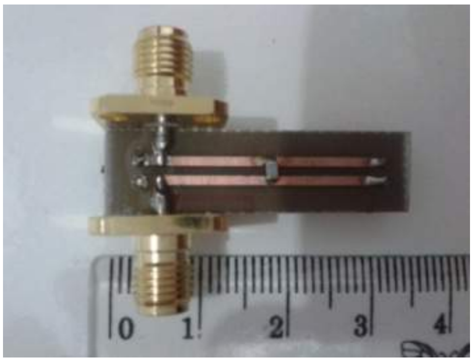

Figure 6. Picture of realized 2-stage microstrip combline BPF prototype

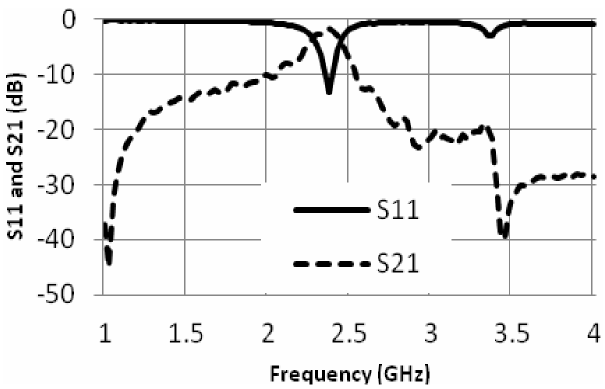

Figure 7. Experimental characterization result of realized 2-stage microstrip combline BPF with capacitor chip incorporation

Fig. 6 shows the realized prototype of 2-stage microstrip combline BPF with capacitor chip incorporation. The prototype is fabricated through wet etching technique on a 0.8mm thick FR4 Epoxy dielectric substrate. In order to characterize experimentally, 2 SMA connectors are soldered at input and output ports of realized filter. Fig. 7 depicts the experimental characterization result realized 2-stage microstrip combline BPF with capacitor chip incorporation. In general, it clearly shows that the measurement result is agreed with the simulation result plotted in Fig. 5.

The discrepancy between measured and simulated results is probably evoked by the different value of tangent loss of FR4 Epoxy dielectric substrate. From the results, it seems that the tangent loss of dielectric substrate for simulation is lower than for fabrication. It shows that the measured -3dB working bandwidth is 170MHz ranges from 2.29GHz to 2.46GHz with center frequency of 2.375GHz which this is 30MHz narrower than the simulated result. Nevertheless, it is noticeable that the realized prototype of microstrip combline BPF has complied with the designed one where the realized filter is suitable for the application which has working frequency of 2.4GHz such as for WLAN communications.

IV. CONCLUSION

The design and implementation of 2-stage microstrip combline BPF with capacitor chip incorporation has been presented. Physical parameter of filter structure has also been investigated to obtain the optimum design of filter. The 2-stage microstrip combline BPF has been fabricated on a 0.8mm thick FR4 Epoxy dielectric substrate with the total dimension of 8.9mm x 3.2mm. From the characterization result, it has been shown that the measured result has coincided with the simulated one. Some discrepancies have occurred due to the possibility of different parameter values of dielectric substrate for simulation and fabrication. It has been shown that the realized prototype of 2-stage microstrip combline BPF has center frequency of 2.375GHz with -3dB working bandwidth of 170MHz ranges from 2.29GHz to 2.46GHz which was comparable to the characterization result with the center frequency of 2.4GHz and -3dB bandwidth response of 200MHz ranges from 2.3GHz to 2.5GHz.

ACKNOWLEDGMENT

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REFERENCES

[1] R. E. Collin, *Foundations for Microwave Engineering*, McGraw-Hill, Inc., 1992.

[2] D. M. Pozar, *Microwave Engineering*, 2nd ed., John Willey & Sons, Inc., 2002

[3] D. K. Misra, *Radio-Frequency and Microwave Communication Circuits: Analysis and Design*, 2nd ed., John Wiley & Sons, Inc., 2004.

[4] K. Chang, *RF and Microwave Wireless Systems*, John Wiley & Sons, Inc., 2000.

[5] A. Sadiku. *Fundamentals of Electric Circuits*, McGraw-Hill, 2008.

[6] 81. H. Lukius, M. S. Arifianto and A. Munir, "Capacitor-based reconfigurable filter for 2.4GHz WLAN, UWB, and uplink GSM-900," in *Proceeding of 5th Indonesia Japan Joint Scientific Symposium (IJJSS) 2012*, Chiba, Japan, Oct. 2012, pp. 339-342.

[7] A. Munir and Y. Prasetio, "Characterization of ultra-wideband bandpass filter with balanced microstrip coupled-lines and dual open stubs perturbation," in *Proceeding of 2nd Asia-Pacific Pacific Conference on Antennas and Propagation (APCAP) 2013*, Chiang Mai, Thailand, Aug. 2013, pp. 199-200.

[8] J.-S. Hong and W. Tang, Reconfigurable Microstrip Combline Filter With Tunable Center Frequency and Bandwidth, Department of electrical and computer engineering, Heriott-watt University. 2011

Performance Analysis of Gigabit Passive Optical Network with Splitting Ratio of 1:64

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Abstract—**In this paper discusses about performance analysis of *Gigabit Passive Optical Network (GPON)* with splitting ratio 1:64. The network topology is simulated using software tool called Optisystem. The transmitter uses a laser components with NRZ modulation scheme and the transmitter power is adjusted to maximum power according to the ITU-T standards. The transmission channel, used a fiber optic cable Single Mode Fiber (SMF) G.652 with a length of 20 km, as maximum length of PON technology, and the cable attenuation of 0.35 dB/km for wavelengths 1310, 1490 nm and 1577 nm. The research concluded that based on link budget, rise time and BER analysis, the performance GPON with 1:64 splitting ratio is not good to be implemented for provide broadband services.**

Keywords—**Optical Network, PON, GPON**

I. INTRODUCTION

Needs of broadband servises is emerging in the near future. The broadband servises and co-existence with existing technologies are the general requirements from network operations to direct PON evolution. Operators worldwide are seeking to increase revenue by developing bandwidth-consuming services. New business models, an exemplified service is HDTV, which requires about 20 Mbit/s per channel such as broadcast TV and radio, video on demand, home online game, interactive E-learning, remote medical services and next-generation 3D TV will dramatically increase bandwidth demand. PON technology development was so diverse, among others is GPON which has bite rate up to 10 Gbps.

A passive optical network is a network that uses point-to-multipoint fiber to the end-points in which unpowered optical splitters are used to enable a single optical fiber to serve multiple end-points. A PON consists of an optical line terminal (OLT) at the originating service provider, transmission channel which consist of optical cable and splitter, and a number of optical network units (ONUs), near end users.

GPON is the development of PON technology that supports high data rate, an increase in security and the choice of 2 layer protocol (ATM and Ethernet). But in practice, the ATM protocol is not longer used.

This technology has a capacity of up to 2.5 Gbps data rate with a 93% efficiency using a GEM frame segmentation for Quality of Service (QoS). Established standards ITU-T G.984 GPON states that can transmit signals with a data rate of 1244.16 or 2488.32 Mbps on the downlink transmission. While on the uplink transmission, the data rate that can be used are 155.52 Mbps; 622.08 Mbps; 1244.16 Mbps; or 2488.32 Mbps. General block diagram of GPON topology using 1:64 splitter shown on figure 1. Downstream bite rate is 2,5 Gbps and upstream bite rate is 1,25 Gbps, according to ITU-T standard.

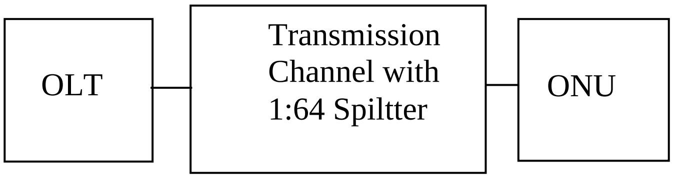

Figure 1. General block diagram of GPON

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A. Optical Line Terminal (OLT)

OLT consist of transmitters and optical amplifiers devices. The transmitter using a DFB laser with power 8 dBm. Modulator is Mach-Zender modulator with NRZ coding and extinction ratio is 25 dB. The circuit of OLT devices can be seen as in figure 2. The Optical amplifier is 20 dB and the upstream receiver device has -28 dBm sensitivity comply to the standard ITU-T. The receiver device uses a PIN photodetector and a low pass filter.

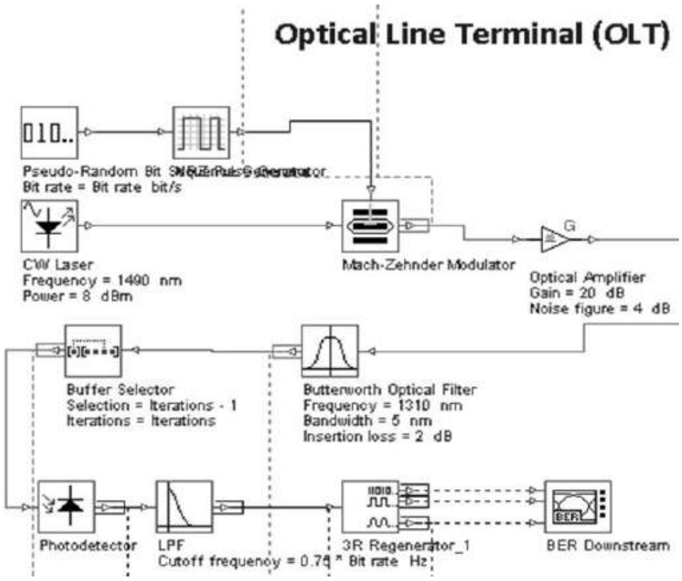

Figure 2. GPON Transmitter

B. Transmission Channel

GPON transmission channel uses the standard ITU-T G.652 fiber optic cable. The worst cable attenuation according to the ITU-T G.652 standard is 0.35 dB/km and the maximum length of the cable is 20 km. The splitter 1:64 places on the remote node as shown on figure 3.

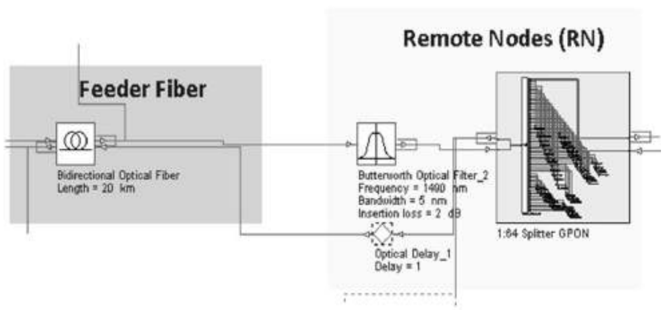

Figure 3. GPON transmission channel

C. Optical Network Unit (ONU)

Optical network unit consist of downstream and upstream receiver devices. The downstream receiver uses photodetector PIN and low pass filter, which has a sensitivity of -21 dBm according to the standard ITU-T. The upstream transmitter using a DFB laser with has power is 8 dBm. Modulator is Mach-Zender modulator with NRZ coding and extinction ratio is 25 dB. The signal is also amplified by the optical amplifier with gains of 20 dB. Configuring the GPON ONU can be seen in figure 4.

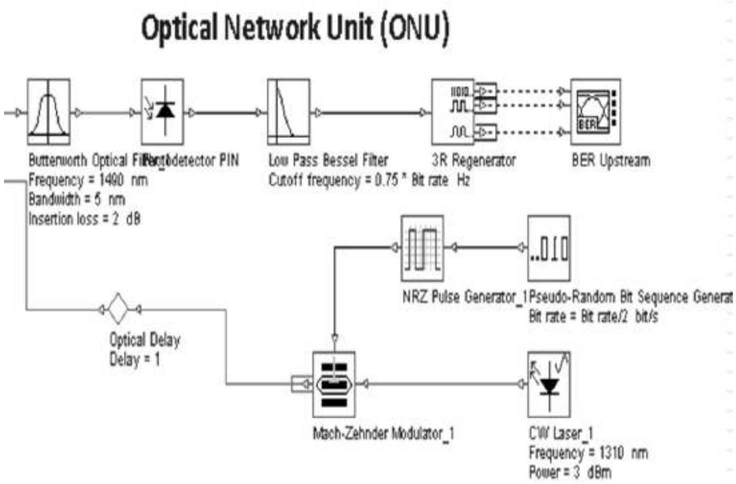

Figure 4. GPON Optical Network Unit

II. PERFORMANCE ANALYSIS

The analysis was performed by calculating the link budget and rise-time budget to see whether these parameters meet the requirements of the basic quality of the network connection. Link budget calculation to determine the suitability of the power to the receiver sensitivity, and rise time budget to test the feasibility of the signal shape and dispersion effect. The design is simulated with software Optisystem to test its feasibility. The parameters tested in the simulation is the power received in each component, the ratio of the signal shape in the receiver and the sender, BER and eye diagram form each receiver.

A.Link Budget

The measurement results indicate that on the downstream side, the average power at the transmitter is 4.9 dBm, and the power at the receiver is -24.1 dBm. The power on the receiver side is smaller than the sensitivity of the receiver (-21 dBm). Based on this fact that received signal is under the sensitivity threshold, performance of downstream network is declared unfit.

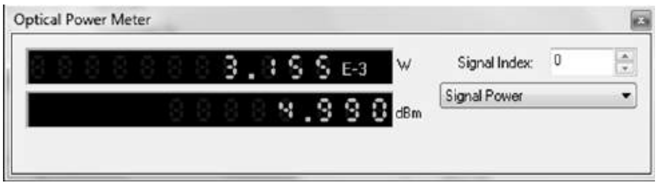

Figure 5. the downstream transmitter power

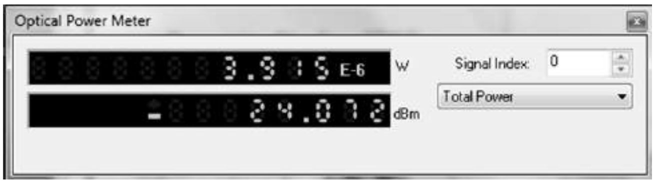

Figure 6. the downstream receiver power

On the upstream side, the link budget analysis is done by observing the power at the transmitter at ONU and the receiver at OLT. The observed power can be seen in figure 7 and figure 8.

Figure 7. the upstream transmitter power

Figure 8. the upstream receiver power

The measurement result on the upstream side that the transmitter power is -0.7 dBm and the receiver power is -27.8 dBm. Although the power at the receiver is greater than the sensitivity of the receiver (-28 dBm), the connections are not enough feasible because the required margin at least 4 dB.

Both the downstream and the upstream connections are not feasible due to the lack of transmitter power. This also proves the ITU-T standard for GPON should has the maximum splitting ratio 1:32 instead of 1:64.

B. Rise-time Budget

Rise-time budget analysis is done by observing the signal shape of a transmitter and receiver. Rise-time budget is said to be viable if the rise-time changes do not exceed 70% of the original time. Rise-time values can be observed from the time it takes to rise from 10 to 90% of the maximum value. Connection is feasible if the rise-time value change does not exceed 70% of the signal period. In the GPON connection, use a bit rate of 2.5 Gbps, the signal bit period is calculated:

$$T_b = \frac{1}{2{,}5 \times 10^9} = 0{,}4\,ns$$

So the rise-time change the maximum allowed is 70% Tb, which is 0.7x0.4= 0.28 ns.

Observations form of signals on the downstream side can be seen in figure 9 and figure 10, while on the upstream side can be seen in figure 11 and figure 12.

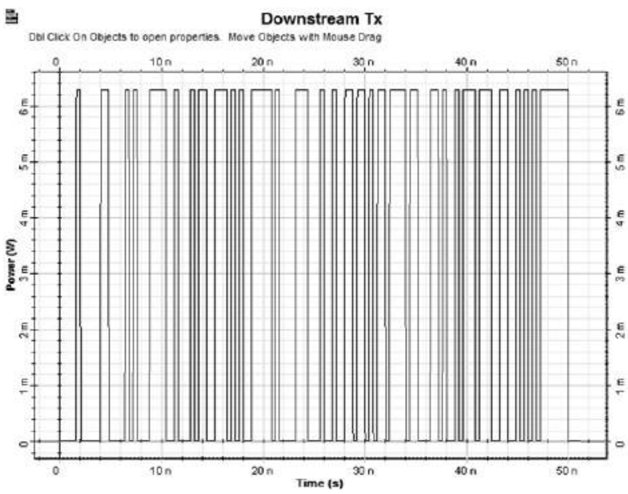

Figure 9. The downstream transmitter signal

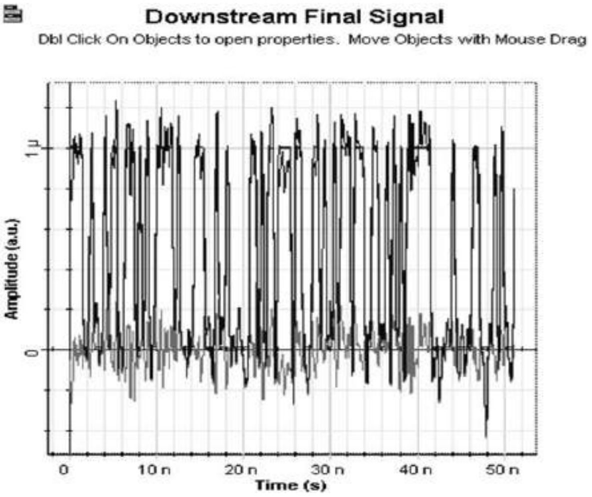

Figure 10. The downstream receiver signal

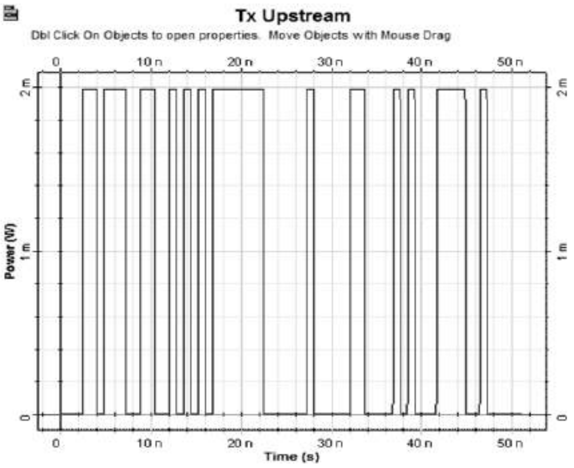

Figure 11. The upstream transmitter signal

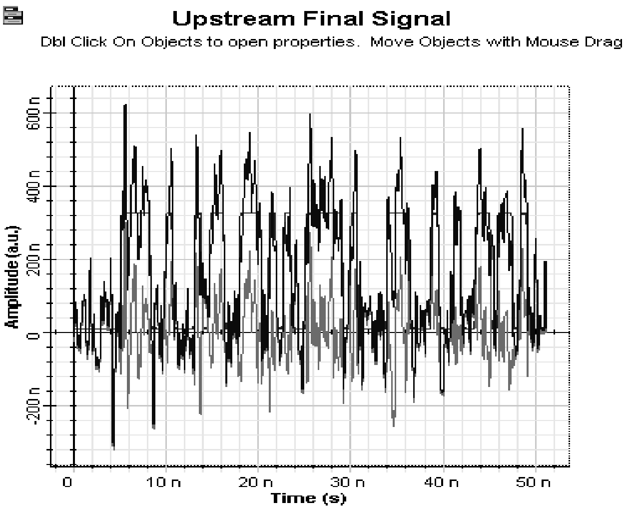

Figure 12. The upstream receiver signal

From the shape of the signal, the received signal is irregular. Although the rise and fall of bits look alike between the receiver and transmitter, but it is difficult to determine the rise time of the signal like this, the top and bottom signals are not visible due to the noise disturbed. In this case, the signal power is too small, so the effect of noise is too significant.

C. BER Analysis

The BER Analyzer generates the eye diagram on output downstream and output upstream. By analysed the eye diagram, the jitter and the BER can be determined. Figure 13 and figure 14 shown the output of BER analyzer.

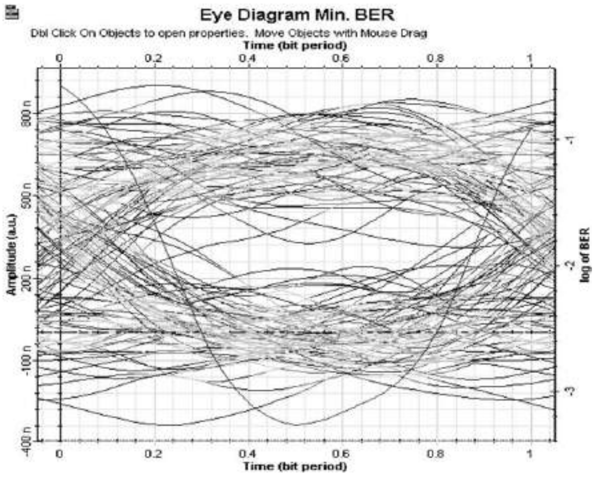

Figure 13. Eye diagram of output *downstream*

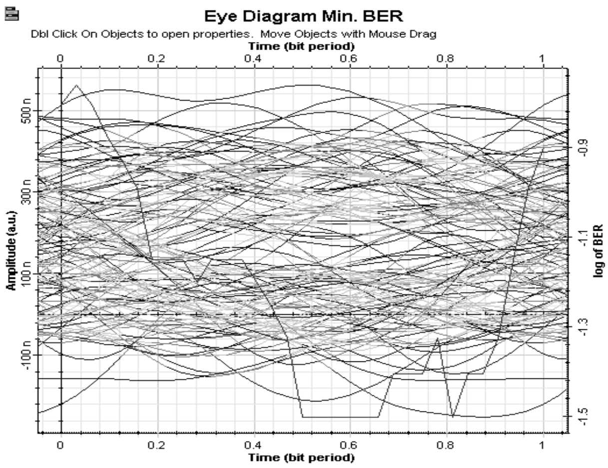

Figure 14. Eye diagram of output *upstream*

Eye diagram of the output BER Analyzer can be seen on Figure 13 and 14. There are a lot of interference on the signal. Small eye indicates Signal to Noise Ratio (SNR) is small. Receiver component will be difficult to read signals with eye diagram like this. Worse results are also shown on Figure 14. The downstream and upstream communication are seen that can not be distinguished between the rise time and fall time signal. Components of the receiver will not be able to read the signals like this.

III. CONCLUSION

The design of optical communication systems by using GPON technology with splitting ratio of 1:64 is not suggested to be implemented due to bad performance. The simulation results showed that the link budget analysis indicates that the received power at downstream and upstream receivers are -24.1 dBm -27.8 dBm respectively. Both numbers are smaller than the sensitivity of the receiver components. Rise-time analysis and BER also showed signs of having too many distractions and difficult to read.

REFERENCES

[1] G. Keiser, *Optical Fiber Communications 4th edition*. Singapore: McGraw-Hill, 2003.

[2] G. Agrawal, *Fiber-Optic Communication Systems*. New York: Wiley, 2002.

[3] A. Gençata *et al.*, “Overview of optical communication networks: Current and future trends,” dalam *The Handbook of Optical Communication Network*. Florida: CRC Press LLC, 2003.

[4] S.B. Weinstein *et al.*, *The Comsoc Guide to Passive Optical Networks*. NJ: John Wiley & Sons Inc., 2012.

[5] M. Kudo *et al.*, “ATM-PON for Optical Communication Transmission/Reception Optical Module,” *OKI Tech. Review* vol. 68, pp 64-67, Maret 2001.

[6] *Gigabit-capable Passive Optical Networks (GPON): Physical Media Dependent (PMD) layer specification*, ITU-T G.984.2, 2003.

[7] *10Gb/s Ethernet PassiveOptical Network*, IEEE 802.3av, 2009.

Characterization of Circular Spiral Patch for Wireless Charging Radiator

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***Abstract*—This paper presents wireless charging radiator design based on electromagnetic coupling resonant. The radiator which is developed based on microstrip technology is expected to work at center frequency of 13MHz. A circular spiral patch has been implemented to accomplish the specifications. Prior hardware realization, the performance of radiator is investigated through simulation. The optimum design for fabrication which is selected based on simulation result have dimension less than 50mm x 65mm, 2890.17mm patch length, 0.3mm patch width, and 0.3mm gap between patch. The experimental measurement is carried out to demonstrate the performance of two radiators as a wireless charging system, one as transmitter and another one as receiver. From simulation result, the radiator works at frequency of 13MHz and has S_{11} parameter value of -4.096dB. Meanwhile the resonant frequency and S_{11} parameter value of realized radiator are 13.2MHz and -3.53dB, respectively.**

Keywords*—Electromagnetic coupling resonant; radiator; wireless charging; circular spiral patch

I. INTRODUCTION

Portable electronic technology has been developed since 1980 and caused many changes in human life. The portable device uses battery as power source and must be recharged using a charger. At the beginning of its development, each device has its own charger and cannot be used on devices from different manufacture. Common charging method for portable device nowadays is charging cable. Different charger for different manufacture is redundant and cause the increasing amounts of electronic waste. Thus, the charger was standardized using the micro-USB connector that can be used on devices from different manufacturers [1]. Other standardization that has been developed is wireless charging which uses electromagnetic waves as medium.

Wireless charging techniques are currently in development. Starting from Nikola Tesla's experiment, wireless charging considered as the latest application of energy transfer without wire [2]. Wireless charging is more convenient than cabling charging by allowing its user used the device remotely while charging the battery. There are some transmission systems for wireless charging, such as direct contact, infrared, and radio frequency. However, this transfer system has technological limitations which are line of sight (LOS) between transmitter and receiver, harmful to the organism and susceptible to interference [1]. To avoid such a problem, a safe and practical energy transfer has been evolved called magnetic coupling resonance method. Magnetic resonance method generates better efficiency and the gap between transmitter and receiver is far enough [3]-[4].

II. BRIEF OVERVIEW OF RADIATOR DESIGN

Radiator is designed by using circular spiral patch. There are two radiators that will be placed facing each other so they will generate electromagnetic coupling. Resonant frequency is selected at 13 MHz. The length of patch line is very long due to the low frequency, so spiral patch is chosen to minimize the overall size of radiator. The desired dimension of substrate may not be greater than 50mm × 65 mm with circular spiral patch printed on it. Theoretically, operating frequency (f) can be calculated using (1).

$$f = \frac{c}{2L_{eff}\sqrt{\varepsilon_r}} \tag{1}$$

where c, L_{eff}, ε_r, are the speed of light in vacuum, the length of patch radiator, and the relative permittivity of dielectric substrate. Thus, it is inferred that the length of patch radiator is inversely proportional to the frequency.

The structure of radiator is shown in Fig. 1. The variation effect of patch length, width and gap between patch will be investigated in simulation which will be discussed later. Furthermore, the material used on the design is FR4 Epoxy dielectric substrate with relative permittivity of 4.4 and 0.8mm thickness. Whilst patch and groundplane employ copper with 0.035mm thickness, and SMA connectors 50 Ω is used as the port. The result of the design is expected to have S_{11} parameter as low as possible and the S_{21} parameter is closed to 0 dB.

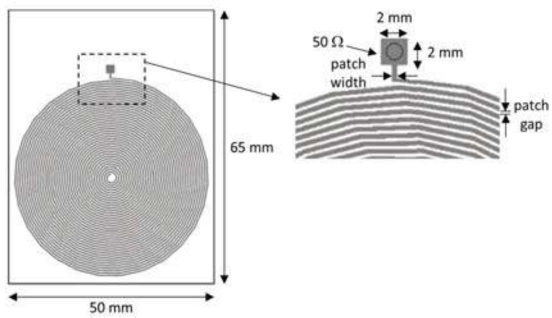

Fig. 1. Structure of circular spiral patch radiator and its geometry

III. NUMERICAL CHARACTERIZATION AND DISCUSSION

To investigate the characteristic of circular spiral patch as wireless charging radiator, some simulation had been done. The simulation only involves one radiator, so only the S_{11} parameter value is analyzed. Figs. 2, 3, and 4 show the effect of varied patch length, patch width, and gap between patch, respectively.

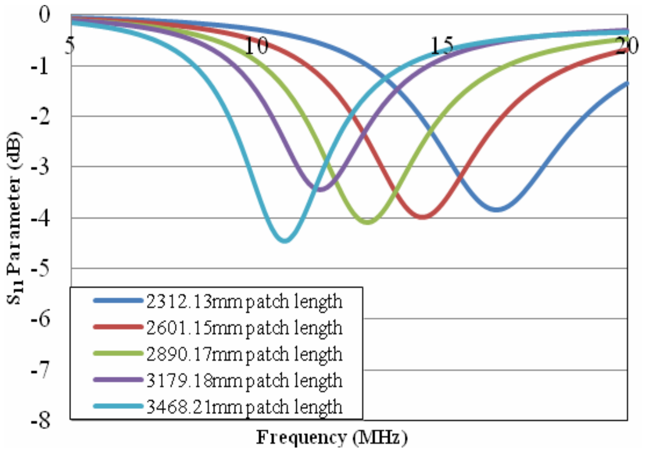

Fig. 2. Simulated results of varied patch length with patch width of 0.3mm and gap between patch of 0.3mm

From Fig. 2, it shows that the simulation result proves (1) which states the patch length is inversely proportional to frequency resonance. The longer the patch length, the frequency resonance will be lower. The resonant frequency of 13MHz is resulted from 2890.17mm. So for further simulation, 2890.17mm patch length is used.

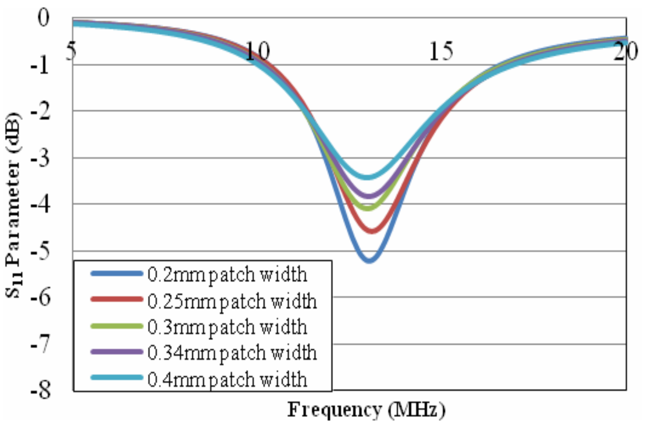

Fig. 3. Simulated results of varied patch width with gap between patch of 0.3mm and patch length of 2890.17mm

Furthermore, Fig. 3 shows that by varying the strip width, the resonant frequencies remain the same however the S_{11} parameter values change. There are 5 variations of patch width i.e. 0.2mm, 0.25mm, 0.3mm, 0.34mm, and 0.4mm. The lowest S_{11} parameter value is given by the patch width 0.2mm that is -5.20dB, which means the value of VSWR is 3.44.

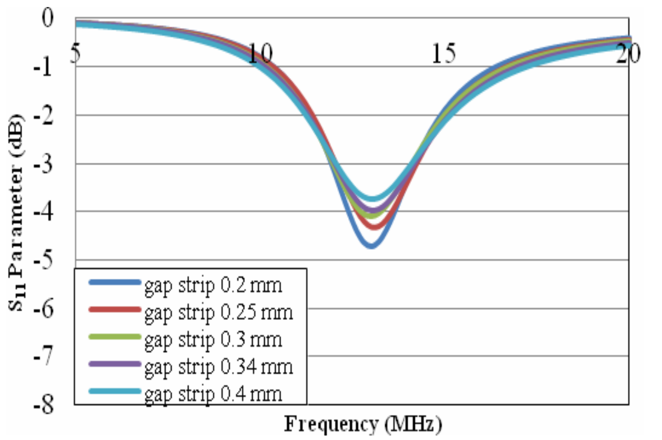

Fig. 4. Simulated results of varied gap between patch with patch width of 0.3 mm and patch length of 2890.17 mm

The simulation result of varied gap between patch is shown in Fig. 4. It shows that the result is the same as varied patch width in which it is unaffected the resonant frequencies but S_{11} parameter value changes. There are 5 variations of gap between patch i.e. 0.2mm, 0.25mm, 0.3mm, 0.34mm, and 0.4mm. The smaller the gap between patch will yields the lower S_{11} parameter value. The lowest S_{11} parameter value is given by 0.2mm gap that is -4.70dB, which means the value of VSWR is 3.79.

IV. FABRICATION AND EXPERIMENTAL CHARACTERIZATION

From simulation, it can be concluded that the best S_{11} parameter value is the one with 0.2mm patch width and 0.3mm gap between patch. However, due to the limitation in fabrication process, the realized hardware has 0.3mm patch width and 0.3mm gap between patch. The radiator is realized through wet etching technique. There are two kinds of wireless charging system to be investigated, namely symmetrical transmitter-receiver (both has right-hand turn) and asymmetrical transmitter-receiver (one right-hand turn, other left-hand turn). Both realized radiators are shown in Figs. 5 and 6.

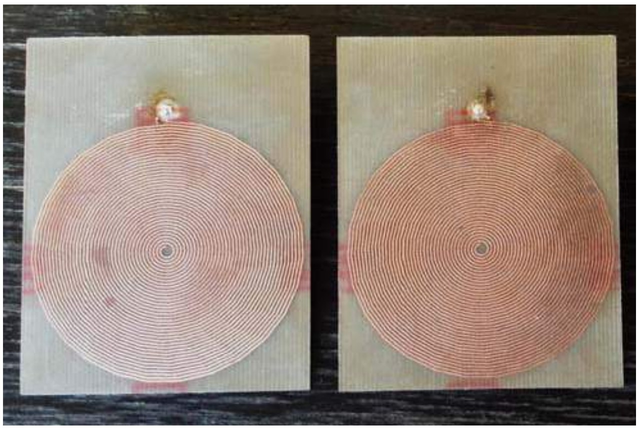

Fig. 5. Symmetrical transmitter-receiver (both right-hand turn)

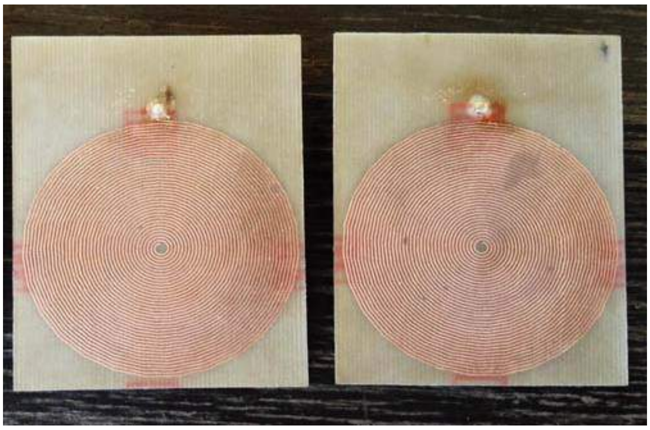

Fig. 6. Asymmetrical transmitter-receiver (right-hand and left-hand turn)

First to be investigated is the performance of one radiator. It is shown in Fig. 7. Experimentally, radiator with right-hand turn and left-hand turn generate similar results. The frequency resonance is 13.2MHz for both right-hand and left hand while the S_{11} parameter value is about -3.56dB. As to compare to simulation, the resonant frequency is 13MHz and the S_{11} parameter value is -4.096dB.

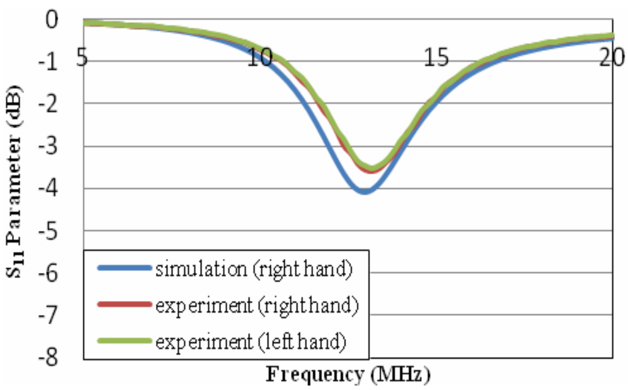

Fig. 7. S_{11} parameter value of realized radiator

Next, the effect of separation gap between two radiators is investigated. The observed separation gap is 0.4mm, 0.8mm, 1mm, 1.5mm, and 2mm. Table I summarizes experimental results for symmetrical transmitter-receiver. As the separation gap get closer, the S_{11} parameter value become lower. The S_{21} parameter value is about -2dB and the operating frequencies are on the range of 12 to 13 MHz.

TABLE I. SUMMARY OF EXPERIMENTED RESULTS FOR VARIED SEPARATION GAP IN SYMETRICAL TRANSMITTER-RECEIVER

Separation gap (mm)	Operating frequency (MHz)	S_{11} (dB)	S_{21} (dB)
0.4	12.95	-27.65	-2.00
0.8	12.88	-25.78	-1.99
1.0	12.13	-21.90	-1.89
1.5	12.58	-21.76	-2.04
2.0	12.28	-21.66	-1.88

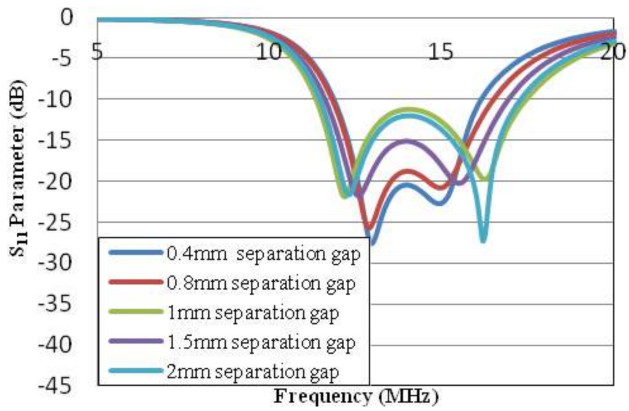

Fig. 8. Experimental result of S_{11} parameter value for varied separation gap in symmetrical transmitter-receiver

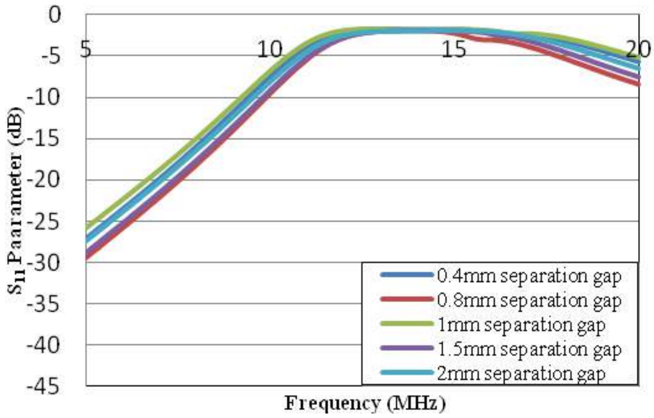

Fig. 9. Experimental result of S_{21} parameter value for varied separation gap in symmetrical transmitter-receiver

Table II shows summary of experimented result for asymmetrical transmitter-receiver. The S_{21} parameter value is about -3 to -5.4 dB and the operating frequencies are in the range of 13 to 14 MHz. The S_{11} parameter is also lower when the separation gap get closer. But in asymmetrical transmitter-receiver, as the separation gap increases there are significant changes in S_{11} and S_{21} parameter values.

TABLE II. SUMMARY OF EXPERIMENTED RESULTS FOR VARIED SEPARATION GAP IN ASYMETRICAL TRANSMITTER-RECEIVER

Separation gap (mm)	Operating frequency (MHz)	S_{11} (dB)	S_{21} (dB)
0.4	13.85	-30.41	-2.99
0.8	13.93	-14.97	-4.51
1.0	14	-11.47	-4.23
1.5	13.93	-9.43	-5.73
2.0	13.78	-7.52	-5.40

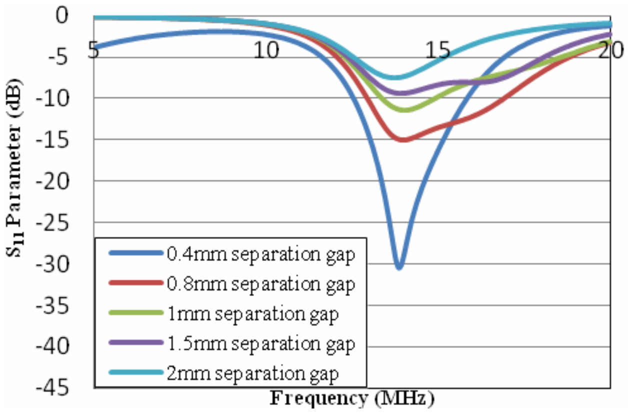

Fig. 10. Experimental result of S_{11} parameter value for varied separation gap in asymmetrical transmitter-receiver

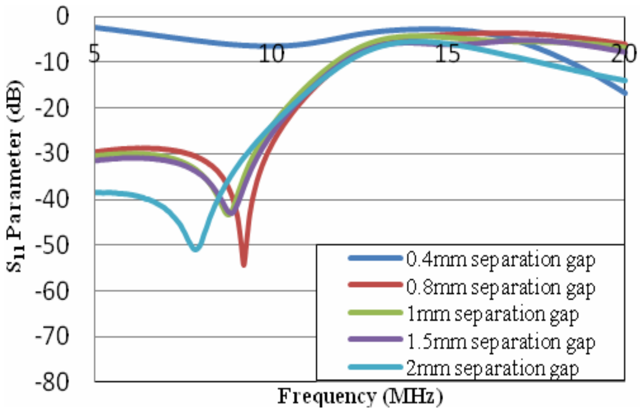

Fig. 11. Experimental result of S_{21} parameter value for varied separation gap in asymmetrical transmitter-receiver

V. Conclusion

The wireless charging radiator has been successfully designed as circular spiral patch and printed on 0.8mm thick FR4 Epoxy dielectric substrate. The dimension of dielectric substrate was 50mm x 65mm while the total length of the patch was 2890.17mm. The simulation had been done to investigate the performance of one radiator. From simulation, it was obtained that the resonant frequency was 13MHz and S_{11} parameter value was -4.096dB while from experiment, resonant frequency was at 13.2MHz and S_{11} parameter is -3.53dB. The performance of two radiators as wireless charging system was also investigated in experiment to analyze the effect of varied separation gap to S_{11} and S_{21} parameter value of radiators. In symmetrical transmitter-receiver, the S_{11} parameter is on the range of -21 to -27dB and S_{21} parameter is about -2dB. While in asymmetrical transmitter-receiver, the S_{11} parameter value was the range of -7dB to -30dB and the S_{21} parameter value was on the range of -3dB to -5.5dB.

References

[1] L. Ovitz, D. Vinko, T. Svedek, *Wireless Power Transfer for Mobile Charging Device*, MIPRO 2012 Croatia, pp. 141-145

[2] S.Y. Hui, *Planar Wireless Charging Technology for Portable Electronic Products and Qi*, Proceedings of the IEEE, 2013, v. 101, p. 1290-1301, 2013.

[3] G. Monti, L. Tarricone, M. Dionigi, and M. Mongiardo, “Magnetically coupled resonant wireless power transmission: An artificial transmission line approach,” in Proceeding of 42nd European Microwave Conference (EuMC) 2012, Amsterdam, The Netherland, Nov. 2012, pp. 233–236.

[4] J. Zhao, G. Xu, C. Zhang, Y. Li, X. Zhang, Q. Yang, Y. Li, and H. Yu, “The design and research of a new kind small size resonator used in magnetic coupling resonance wireless energy transmission system,” IEEE Trans. Magn., Vol. 48, No. 11, pp. 4030–4033, Nov. 2012.

Cylindrical Coordinate System-based Full Wave FDTD Computation for Resonant Frequency Calculation of Circular Cavity Resonator

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***Abstract*—In this paper, a full wave finite difference time domain (FDTD) computation method based on cylindrical coordinate system is proposed for calculating the resonant frequency of circular cavity resonator. The use of FDTD method with cylindrical coordinate system instead of Cartesian coordinate system is considered due to the geometry shape of analyzed structure, i.e. circular cavity resonator. In the computation, a hollow circular cavity resonator made of perfect conductor with the radius of 50mm and length of 100mm is discretized based on cylindrical coordinate system and numerically computed to determine its resonant frequencies. To verify the proposed method, the theoretical approach for the cavity is carried out by calculating its resonant frequencies for correspondent resonance mode. In addition, a simulation for determining resonant frequencies of the cavity is also conducted using finite element method (FEM) commercialized software. It shows that the FDTD computation result demonstrates an acceptable accuracy compared to the theoretical approach with the discrepancy less than 5% although it is worse than the result of FEM commercialized software.**

Keywords*—Circular cavity resonator; cylindrical coordinate system; full wave FDTD method; resonant frequency.

I. INTRODUCTION

Recently, a numerical analysis which is usually carried out by using simulation has been an inseparable step in the design of electromagnetics-based structure, such antenna, filter, waveguide, and resonant cavity [1]– [5]. The numerical process becomes significant for the designer since it can closely approximate the result before the implementation. As a result, the process can prune and mitigate the trial and error efforts. Therefore, a simulation algorithm or computational method should be as precise as possible with the theoretical approach. In addition, the ease of Maxwell's equations derivation and computational cost should be considered as well in choosing the simulation algorithm [6]– [7].

Basically, a computational method for analyzing electromagnetic wave problems derived from Maxwell's equations is commonly constructed in a Cartesian coordinate system [2]– [8]. However, for some structures with rounded shape cross-section such as circular cavity resonator, coaxial cable, or earth surface, the Cartesian coordinate system has no ability to exactly model the structure. As a consequent, the computational algorithm should be revamped to satisfy the shape of structures, otherwise the computational method should be altered using an appropriate coordinate system. Furthermore, the orthogonal grid structure of computational method implies the structures that have edges which follow the geometry of grid. This can pose a problem when the computation method is facing curved surface geometries. Some attempts have been resolved by using more numbers of computational grids especially for grid-based computational methods, as it could more precisely resemble the actual structure, however the effort costs the computational resources and time [9]– [10].

In this paper, to suit the rounded shape of structure, a full wave FDTD computation method based on cylindrical coordinate system is proposed for calculation of resonant frequency for a circular cavity resonator. The method of FDTD which has been introduced first time by Yee in 1966 [11], is a numerical method with the time and space discretizations in Maxwell's equations which leads to a recursive time-marching algorithm where the field solutions depend on values of field from the previous step [8], [10]. Due to the shape of structure, i.e. circular cavity resonator, the most appropriate coordinate system used for the analysis is the cylindrical coordinate system [12]. To compute the resonant frequency, a hollow circular cavity resonator made of perfect conductor is modeled in the FDTD notation and discretized in time and space based on cylindrical coordinate system (ρ, φ, z). While for verifying the full wave FDTD computation result, a theoretical approach for the cavity is carried out by calculating its resonant frequencies for correspondent resonance mode. In addition, a comparison of simulation result from a commercialized software employing FEM is also performed.

II. CIRCULAR CAVITY RESONATOR AND CYLINDRICAL COORDINATE SYSTEM-BASED FULL WAVE FDTD METHOD

Fig. 1 illustrates a hollow circular cavity resonator made of perfect conductor with the radius (a) of 50mm and length (d) of 100mm used for the resonant frequency computation. The resonance mode of Transverse Magnetic (TM) is applied for the computation, therefore the electric field is assumed

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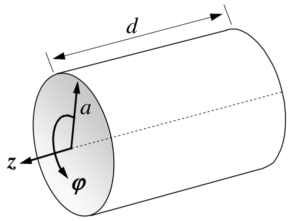

Fig. 1. A hollow circular cavity resonator used for full wave FDTD computation

to propagate in the $+z$ direction as $e^{(j\omega t - j\beta z)}$. Based on Maxwell's equations in differential form, the electric and magnetic fields in inside of cavity can be expressed in (1) and (2) [13], where μ_0 and ϵ_0 are permeability and permittivity of vacuum, respectively.

$$\nabla \times \mathbf{E} = -\mu_0 \frac{\partial \mathbf{H}}{\partial t} \tag{1}$$

$$\nabla \times \mathbf{H} = \epsilon_0 \frac{\partial \mathbf{E}}{\partial t} \tag{2}$$

The resonant frequencies for TM mode in the cavity can be calculated using (3) [13], where X_{np} denotes the p^{th} zero of the n^{th} Bessel function and q is an arbitrary integer. Table I tabulates a few of them for some lower order n.

$$f_{mnp} = \frac{c}{2\pi} \sqrt{\left(\frac{X_{np}}{a}\right)^2 + \left(\frac{q\pi}{d}\right)^2} \tag{3}$$

TABLE I. p^{th} ZERO OF n^{th} BESSEL FUNCTION FOR TM_{np} MODE

p	n					
	0	1	2	3	4	5
1	2.405	3.832	5.136	6.380	7.588	8.771
2	5.520	7.106	8.417	9.761	11.065	12.339
3	8.645	10.173	11.620	13.015	14.372	
4	11.792	13.324	14.796			

From the concept of curl operation for cylindrical coordinate system, 2 pairs of electric and magnetic fields in ρ, φ, and z axes can be obtained directly from (1) and (2). The pairs of fields are expressed in (4)–(6) and (7)–(9) which correspondence with (1) and (2), respectively.

$$\left[\frac{1}{\rho}\frac{\partial E_z}{\partial \varphi} - \frac{\partial E_\varphi}{\partial z}\right] = -\mu_0 \frac{\partial H_\rho}{\partial t} \tag{4}$$

$$\left[\frac{\partial E_\rho}{\partial z} - \frac{\partial E_z}{\partial \rho}\right] = -\mu_0 \frac{\partial H_\varphi}{\partial t} \tag{5}$$

$$\left[\frac{1}{\rho}\frac{\partial \rho E_\varphi}{\partial \rho} - \frac{1}{\rho}\frac{\partial E_\rho}{\partial \varphi}\right] = -\mu_0 \frac{\partial H_z}{\partial t} \tag{6}$$

$$\left[\frac{1}{\rho}\frac{\partial H_z}{\partial \varphi} - \frac{\partial H_\varphi}{\partial z}\right] = \epsilon_0 \frac{\partial E_\rho}{\partial t} \tag{7}$$

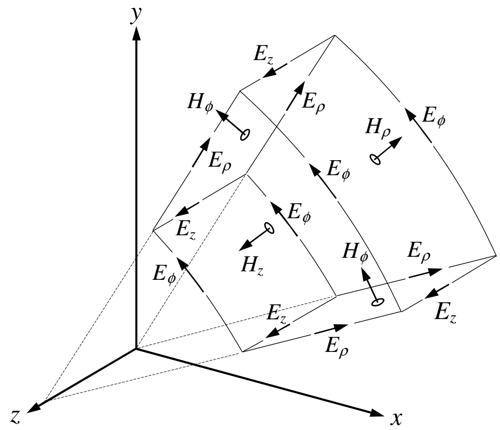

Fig. 2. Fields placement in full wave FDTD method based on cylindrical coordinate system

$$\left[\frac{\partial H_\rho}{\partial z} - \frac{\partial H_z}{\partial \rho}\right] = \epsilon_0 \frac{\partial E_\varphi}{\partial t} \tag{8}$$

$$\left[\frac{1}{\rho}\frac{\partial \rho H_\varphi}{\partial \rho} - \frac{1}{\rho}\frac{\partial H_\rho}{\partial \varphi}\right] = \epsilon_0 \frac{\partial E_z}{\partial t} \tag{9}$$

Then, a full wave FDTD notation in cylindrical coordinate system is applied for each field in the circular cavity resonator by discretizing the electric and magnetic fields in time and space. The full wave FDTD notation is expressed in (10)–(12) and (13)–(15) for electric field and magnetic field components, respectively [10]. It is noticeable that the notations have half-integer indexes which signifies both fields are on the same plane but different locations, as illustrated in Fig. 2.

$$\frac{H_\rho\big|^{n+\frac{1}{2}}_{i,j+\frac{1}{2},k+\frac{1}{2}} - H_\rho\big|^{n-\frac{1}{2}}_{i,j+\frac{1}{2},k+\frac{1}{2}}}{\Delta t} = \frac{E_\varphi\big|^{n}_{i,j+\frac{1}{2},k+1} - E_\varphi\big|^{n}_{i,j+\frac{1}{2},k}}{\mu_0 \Delta z} - \frac{E_z\big|^{n}_{i,j+1,k+\frac{1}{2}} - E_z\big|^{n}_{i,j,k+\frac{1}{2}}}{\rho_i \mu_0 \Delta\varphi} \tag{10}$$

$$\frac{H_\varphi\big|^{n+\frac{1}{2}}_{i+\frac{1}{2},j,k+\frac{1}{2}} - H_\varphi\big|^{n-\frac{1}{2}}_{i+\frac{1}{2},j,k+\frac{1}{2}}}{\Delta t} = \frac{E_z\big|^{n}_{i+1,j,k+\frac{1}{2}} - E_z\big|^{n}_{i,j,k+\frac{1}{2}}}{\mu_0 \Delta\rho} - \frac{E_\rho\big|^{n}_{i+\frac{1}{2},j,k+1} - E_\rho\big|^{n}_{i+\frac{1}{2},j,k}}{\mu_0 \Delta z} \tag{11}$$

$$\frac{H_z\big|^{n+\frac{1}{2}}_{i,j+\frac{1}{2},k+\frac{1}{2}} - H_z\big|^{n-\frac{1}{2}}_{i,j+\frac{1}{2},k+\frac{1}{2}}}{\Delta t} = \frac{E_\rho\big|^{n}_{i+\frac{1}{2},j+1,k} - E_\rho\big|^{n}_{i+\frac{1}{2},j,k}}{\rho_{i+\frac{1}{2}} \mu_0 \Delta\varphi} - \frac{\rho_{i+1} E_\varphi\big|^{n}_{i+1,j+\frac{1}{2},k} - \rho_i E_\varphi\big|^{n}_{i,j+\frac{1}{2},k}}{\rho_{i+\frac{1}{2}} \mu_0 \Delta\rho} \tag{12}$$

$$\frac{E_\rho|^{n+1}_{i+\frac{1}{2},j,k} - E_\rho|^{n}_{i+\frac{1}{2},j,k}}{\Delta t} = \frac{H_z|^{n+\frac{1}{2}}_{i+\frac{1}{2},j+\frac{1}{2},k} - H_z|^{n+\frac{1}{2}}_{i+\frac{1}{2},j-\frac{1}{2},k}}{\rho_{i+\frac{1}{2}}\epsilon_0\Delta\varphi} - \frac{H_\varphi|^{n+\frac{1}{2}}_{i+\frac{1}{2},j,k+\frac{1}{2}} - H_\varphi|^{n+\frac{1}{2}}_{i+\frac{1}{2},j,k-\frac{1}{2}}}{\epsilon_0\Delta z} \quad (13)$$

$$\frac{E_\varphi|^{n+1}_{i,j+\frac{1}{2},k} - E_\varphi|^{n}_{i,j+\frac{1}{2},k}}{\Delta t} = \frac{H_\rho|^{n+\frac{1}{2}}_{i,j+\frac{1}{2},k+\frac{1}{2}} - H_\rho|^{n+\frac{1}{2}}_{i,j+\frac{1}{2},k-\frac{1}{2}}}{\epsilon_0\Delta z} - \frac{H_z|^{n+\frac{1}{2}}_{i+\frac{1}{2},j+\frac{1}{2},k} - H_z|^{n+\frac{1}{2}}_{i-\frac{1}{2},j+\frac{1}{2}.k}}{\epsilon_0\Delta\rho} \quad (14)$$

$$\frac{E_z|^{n+1}_{i,j,k+\frac{1}{2}} - E_z|^{n}_{i,j,k+\frac{1}{2}}}{\Delta t} = -\frac{H_\rho|^{n+\frac{1}{2}}_{i,j+\frac{1}{2},k+\frac{1}{2}} - H_\rho|^{n+\frac{1}{2}}_{i,j-\frac{1}{2},k+\frac{1}{2}}}{\rho_i\epsilon_0\Delta\varphi} + \frac{\rho_{i+\frac{1}{2}}H_\varphi|^{n+\frac{1}{2}}_{i+\frac{1}{2},j,k+\frac{1}{2}} - \rho_{i-\frac{1}{2}}H_\varphi|^{n+\frac{1}{2}}_{i-\frac{1}{2},j,k+\frac{1}{2}}}{\rho_i\epsilon_0\Delta\rho} \quad (15)$$

The discretization in space produces the size of grid in ρ, φ, and z axes notated by $\Delta\rho$, $\Delta\varphi$, and Δz, respectively. The grid size depends on the wavelength applied in the computation, the smaller the grid size compared to the wavelength, the more accurate the outcome, but the excessive grid size means more computational cost required. It should be noted that H_z in (12)–(14) equals zero since the resonance mode used for analysis is TM mode. In order the computation has stability and convergency, a stability criterion for cylindrical coordinate system is used and expressed in (16) [9], where ν_{max} is the maximum phase velocity which can be substituted by a speed of light in case of hollow vacuum cavity.

$$\Delta t \leq \frac{1}{\nu_{max}}\left[\frac{1}{(\Delta\rho)^2} + \frac{1}{(\rho\Delta\varphi)^2} + \frac{1}{(\Delta z)^2}\right]^{-\frac{1}{2}} \quad (16)$$

III. Computation Result and Comparison

To excite the circular cavity resonator in order to observe the resonant frequencies, a wave source of sinusoidal-modulated Gaussian pulse with center frequency of 4GHz and bandwidth of 5GHz is used as an excitation source. The waveform of wave source in time domain is plotted in Fig. 3. The excitation source which takes the plane source is positioned in inside of the resonator at one grid cell away from the circular wall of resonator. The boundary condition for the walls of resonator is configured as perfect electric wall. Since the center frequency of wave excitation is 4GHz, the grid size for $\Delta\rho$, $\Delta\varphi$, and Δz is set to be 1mm, 1^o and 1mm, respectively. Therefore, the number of grids for the computation is 50 × 360 × 100 grids.

Fig. 4 shows the spectrum of TM mode resonant frequencies for the circular cavity resonator after performing Fourier transform for the time-domain data obtained from the full wave FDTD computation. The value of each

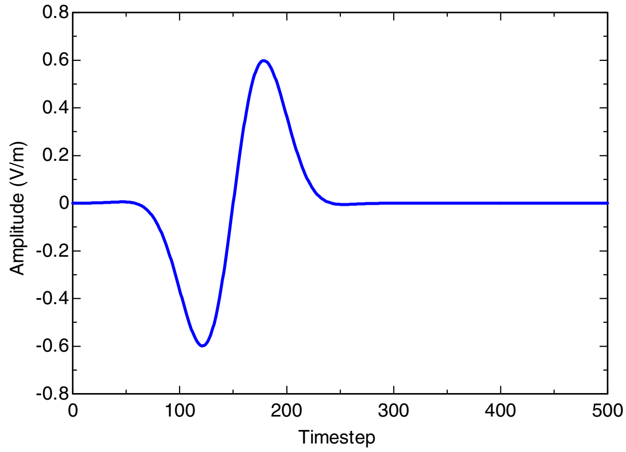

Fig. 3. A sinusoidal-modulated Gaussian pulse for excitation source in full wave FDTD computation

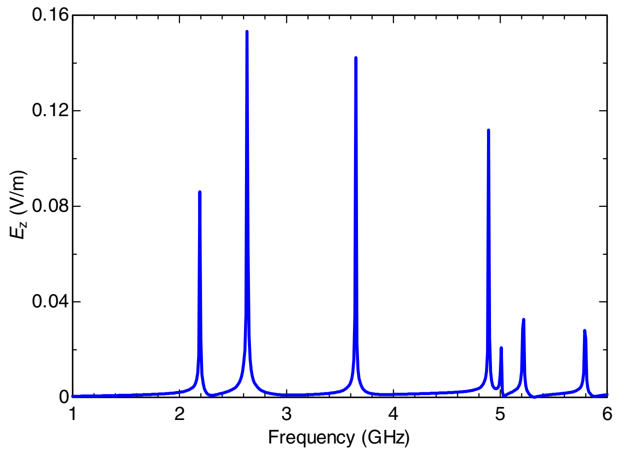

Fig. 4. Spectrum of TM mode resonant frequencies obtained from full wave FDTD method based on cylindrical coordinate system

TABLE II. TM mode resonant frequencies of cavity obtained from theoretical approach, FDTD method, FEM method and comparison of both methods to theoretical approach

$f_{npq,\ \mathrm{theory}}$ (GHz)	$f_{npq,\ \mathrm{FDTD}}$ (GHz)	$f_{npq,\ \mathrm{FEM}}$ (GHz)	Δf_{FDTD} (%)	Δf_{FEM} (%)
2.296	2.190	2.326	4.167	1.307
2.743	2.629	2.769	4.119	0.948
3.778	3.650	3.802	3.388	0.635
5.052	4.890	5.062	3.088	0.198
5.271	5.009	5.201	4.952	1.328
5.480	5.220	5.405	4.744	1.369
6.065	5.790	6.090	4.534	0.412

resonant frequency of TM mode and its comparison with the theoretical approach and the FEM commercialized software is summarized in Table II. The discrepancy between the results obtained from proposed FDTD method and the theoretical approach is probably evoked by some assumptions used in the FDTD method such as boundary condition at the wall of cavity resonator. Nevertheless, the discrepancy is less than 5% from the theoretical approach, indicating an acceptable accuracy when it is used in computing relatively for low frequency signal. Although the FDTD result seems to be worse than the result of FEM commercialized software, however it is expected that the proposed FDTD method could produce the comparable accuracy to the FEM commercialized software on curved surface structures versus stair-stepping edges.

IV. Conclusion

The cylindrical coordinate system-based full wave FDTD computation for resonant frequency of circular cavity resonator has been demonstrated. The theoretical approach has also been presented to verify the full wave FDTD computation result as well as the FEM commercialized software as a comparison. Although there was a discrepancy of the full wave FDTD computation result less than 5% to the theoretical approach, it has been shown that the proposed full wave FDTD method has an acceptable accuracy compared to the theoretical approach and has a good agreement with the FEM commercialized software. Therefore, it can be concluded that the proposed FDTD method based on cylindrical coordinate system could be effectively solving electromagnetic wave problems for some structures with appropriate shape.

References

[1] J. L. Volakis, A. Chatterjee and L. C. Kempel, *Finite Element Method Electromagnetics: Antennas, Microwave Circuits, and Scattering Applications*, 1st ed., NY: Wiley-IEEE Press, 1998.

[2] S. M. Rao, *Time Domain Electromagnetics*, 1st ed., Academic Press, 1999.

[3] W. C. Gibson, *The Method of Moments in Electromagnetics*, NW: Chapman & Hall/CRC, 2007.

[4] G. Pelosi, "The finite-element method, Part I: R. L. Courant [Historical Corner]," *IEEE Antennas Propag. Mag.*, Vol. 49, Issue 2, pp. 180-182, Apr. 2007.

[5] M. Randa and A. Munir, "3D-FDTD method for analysis of rectangular waveguide loaded with anisotropic dielectric material," in *Proceeding of International Conference on Electrical Engineering, Computer Science and Informatics (EECSI)*, Yogyakarta, Indonesia, Aug. 2014, pp. 500–503.

[6] E. Yamashita, *Analysis Method for Electromagnetic Wave Problems*, 2Rev. ed., MA: Artech House, 1995.

[7] D. M. Sullivan, *Electromagnetic Simulation Using The FDTD Method*, 1st ed., NJ: Wiley-IEEE Press, 2000.

[8] A. Taflove and S. C. Hagness, *Computational Electrodynamics: The Finite Difference Time Domain Method*, 3rd ed., MA: Artech House, 2005.

[9] N. K. Madsen and R. W. Ziolkowski, "Numerical solutions of Maxwells equations in the time domain using irregular nonorthogonal grids," *Waves Motion*, Vol. 10, Isuue 6, pp. 538-596, Dec. 1988.

[10] U. S. Inan and R. A. Marshall, *Numerical Electromagnetics: The FDTD Method*, NY: Cambridge University Press, 2011.

[11] K. S. Yee, "Numerical solution of initial boundary value problems involving Maxwell's equation in isotropic media," *IEEE Trans. Antennas Propag.*, Vol. AP-14, Issue 3, pp. 302-307, May 1966.

[12] R. Rahmatillah, Chairunnisa and A. Munir, "Numerical analysis for wave propagation in circular waveguide using cylindrical coordinate system-based FDTD method," in *Proceeding of International Conference on Advanced Informatics: Concepts, Theory and Applications (ICAICTA)*, Bandung, Indonesia, Aug. 2014, pp. 215–219.

[13] S. Y. Liao, *Microwave Devices and Circuits*, 3rd ed., NJ: Prentice Hall, 1996.

Beam Reconfiguration of Capacitor-based Square Patch Antenna Array

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***Abstract*—In this paper, a beam reconfiguration of antenna array composed of four elements microstrip square patch antenna is proposed by use of capacitor chips. The array of antenna which is deployed on an FR4 Epoxy dielectric substrate is intended to operate at center frequency 2.4GHz for wireless communication application. The four patch antennas are arranged in a horizontal array employing microstrip line as its feeding network. There are two configurations of capacitor chips incorporated into the antenna array, the first with capacitances of 1pF, 10pF, 10pF, 1pF and the second with capacitances of 0.5pF, 1.5pF, 1.5pF, 0.5pF for each feeding line connected the antenna patch. The beam reconfiguration is shown by characterizing the antenna array and comparing the result for an array without and with capacitor chips. From the measured results, it shows that the magnitude and phase of reflection coefficient as well as the radiation pattern of proposed antenna array are influenced by the presence of capacitor chips. The first configuration of capacitor chips changes the reflection coefficient at frequency of 2.46GHz from -17.18dB to -9.68dB, the phase shift into 81.48^o, and a slight deflecting beam maximum angle of 10^o. Whilst from the second configuration, the result shows the reflection coefficient of -7.46dB at frequency of 2.48GHz, shift of antenna phase into 73.89^o, and deflection of beam angle up to 20^o.**

Keywords*—Antenna array; beam reconfiguration; capacitor ships; radiation patten; reflection coefficient.

I. Introduction

To overcome the rapid globalization, almost over 2 last decades, telecommunication experts have been actively developing and upgrading communication technologies to be even more advanced. For telecommunication users which are more dynamic and have high mobility, the necessities of device with outstanding features are becoming unavoidable things. The development of wireless communication technology has enabled delivering the information without physical wire connection which helps the users in satisfying their needs to be conveniently active [1]. The mobility, flexible capacity, easier scheme, and lower cost of telecommunication devices have encouraged people to abandon the previously popular technology, wired communication, and spend an unreasonably huge amount on wireless communication devices [1]– [2].

New ideas in the design of communication device are affected by the expansion of communication technology particularly in wireless communication system [3]– [4]. This also affects to the design of antenna as an important part for the system. Based on the impacts of globalization, excellent performances in such system are something to be essential due to the increasing number of traffic in coming years. Therefore, it is necessary to have an excellent telecommunication performance; otherwise the compatibility of antenna may not be suitable for wireless applications [4]. To obtain the required performance, recently reconfigurable antennas have become one of the methods which are widely used in wireless communication system. The most common reconfigurable antennas were classified into polarization reconfiguration, frequency reconfiguration, and beam reconfiguration [5]– [6]. Moreover, reconfiguring the beam of antenna can be carried out to magnify the spectral efficiency and lessen the disadvantages of multipath propagation [7]. Hence, steering the beam of antenna into the direction of stronger signal could reduce the power consumption and relieve unwanted interferences [8].

In this paper, a beam reconfiguration of square patch antenna array designed by use of microstrip technology is investigated. As is already well-known, the microstrip technology has made a patch antenna becoming easy to be fabricated. The antenna can also be either a single element or a part of antenna array [9]. In practical, it is hard to steer the beam by using only a single element microstrip antenna, hence several microstrip antennas are employed as the elements of an antenna array. To be able minimizing the input power and widening the coverage area, an antenna array is a favorable design to accomplish the purpose to increase its efficiency, therefore it would be the appropriate method to develop a beam reconfigurable antenna.

In the current work, to have a beam reconfiguration of square patch antenna array, two slightly different configurations of antenna array with microstrip transmission line feeding network are proposed to be designed. Ideally, the beam of antenna array can be reconfigured from -90^o to +90^o in elevation plane. However, due to the use of patch antenna, the deflection of beam angle will be less than $\pm$90^o. The antenna array itself is deployed on an FR4 Epoxy dielectric substrate with the dielectric constant of 4.3. At first, a conventional antenna array consists of four patch antennas is arranged. Afterwards the second proposed antenna which has similar design as the first one is composed. However, for the second one there are capacitor chips in some configuration incorporated into the microstrip transmission line for each element of antenna array. By varying their capacitance values, the beam direction of proposed antenna array could be controlled towards the desired direction. The design and characterization result for both antenna arrays with and without capacitor chips are described. Some discussion related to characterization and measurement results will be presented.

II. Brief Overview of Antenna Array Design

Fig. 1 shows the design of square patch antenna array proposed for 2.4GHz wireless communication application. The array of antenna is constructed of four elements of square patch designed on the top side of dielectric substrate and fed by microstrip transmission line feeding network positioned at the bottom side of dielectric substrate. All the elements of square patch and the feeding network made of metal copper are implemented on FR4 Epoxy dielectric substrates with the thickness of 1.6mm. The groundplane is placed at the middle side between the patches and the feed line. The thickness of metal copper square patches and feeding network as well as the groundplane is set to be 0.035mm. A via wire is used to connect each antenna patch on the top side to the feed line end at the bottom side as illustrated in Fig. 2.

To obtain the beam reconfiguration characteristic, four capacitor chips with some configuration capacitance values are incorporated into the feeding network. A small gap is provided at each feed line for incorporating a capacitor chip. Meanwhile, for the conventional one, i.e. square patch antenna array without capacitor chips, the design is the same but without any gaps for capacitor chips incorporation. The dimension of each element of square patch is 29mm (l) $\times$ 29mm (w) which corresponds to resonant frequency of 2.4GHz. The four elements of square patch are configured in a horizontal array with the same distance between square patch elements (d) of 38.94mm to attain an optimum lobe of radiation pattern.

Furthermore, the microstrip line equations in [9] are used to determine the width of each line for the feeding network. Since each feed line has a different line impedance, hence it should be calculated individually starting from the feed line connected to the square patch element. Based on parametrical study, the feed line connected to the square patch element have line impedance of 94.25Ω. For the square patch antenna array with capacitor chips, the capacitor chips are incorporated in these feed lines. Then by setting a 100Ω line impedance into two 94.25Ω line impedances, a quarter-wave transformer is chosen to match both line impedances. Therefore, the line impedance of the quarter-wave transformer is 97.08Ω. Moreover, there is a feed line connected to the port, which has a line impedance of 50Ω which is connected to a parallel of 100Ω line impedances. Then, the 100Ω line impedance is split into two 50Ω line impedances, hence another quarter-wave transformer is required. Here, the line impedance of second quarter-wave transformer is set to be 70.71Ω. The detail parameter values of microstrip line for feeding network are tabulated in Table I.

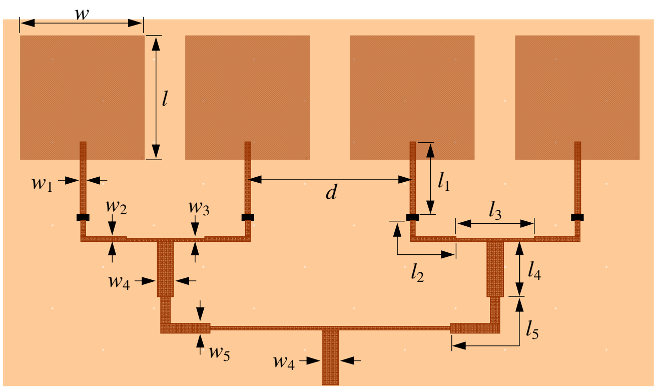


Fig. 1. Rough design of square patch antenna array for beam reconfiguration.

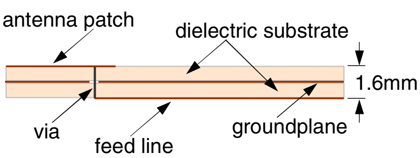


Fig. 2. A via wire for connection of each antenna patch to feed line.

TABLE I. Parameter values of microstrip line

Line impedance (Ω)	Width (mm)	Length (mm)
94.25	0.80 (w_1)	22.00 (l_1)
97.08	0.72 (w_2)	18.15 (l_2)
100.00	0.65 (w_3)	11.00 (l_3)
50.00	3.11 (w_4)	13.00 (l_4)
70.71	1.64 (w_5)	16.54 (l_5)

III. Characterization and Hardware Realization

A. Characterization and Discussion

To investigate the beam reconfiguration of square patch antenna array, at first, a characterization proposed antenna array without capacitor chips, called as Array-W/O, is carried out through simulation. Then, two proposed antenna arrays with capacitor chips in different configuration of capacitance values are characterized to be compared with the first one. For the antenna array with capacitor chips, the first configuration of capacitance value is 1pF, 10pF, 10pF, 1pF, called as Array-W1, and the second one is 0.5pF, 1.5pF, 1.5pF, 0.5pF, called as Array-W2. The capacitors values for both configurations, i.e. Array-W1 and Array-W2, are determined based on the parametrical studies in producing an optimum beam reconfiguration and satisfying the design requirements. Here, the discrepancies in resonant frequency and in value of return loss which are applied as design requirements should be satisfied less than 5% and 10dB, respectively. The characterization results of reflection coefficient i.e. magnitude and phase, and radiation pattern for 3 types of square patch antenna array are plotted in Figs. 3, 4, and 5, respectively.

It shows from Fig. 3 that the resonant frequencies of square patch antenna array with capacitor chips, i.e. Array-W1 and Array-W2, slightly shift from frequency of 2.4GHz for Array-W/O to the higher frequency. The resonant frequencies for Array-W1 and Array-W2 are 2.42GHz and 2.44GHz, respectively. Further, the minimum values of reflection coefficient for Array-W1 and Array-W2 are also increased to be worse. It indicates that the minimum value of reflection coefficient for Array-W/O is -30.58dB at frequency of 2.4GHz, while for Array-W1 and Array-W2 the minimum values of reflection coefficient are -16.31dB at frequency of 2.42GHz and -9.72dB at at frequency of 2.44GHz, respectively. The discrepancy occurs due to the incorporation of capacitor chips which affects to the reactance value of line impedance of impedance matching network. Hence, as the change of line impedance, the resonant frequency as well the value of reflection coefficient is also influenced.

From Fig. 4, it reveals that the result of reflection coefficient phase for 3 types of square patch antenna array resembles each other. The difference is slightly in the magnitude of phase, but the general shape is similar. The reflection coefficient phase of Array-W/O at frequency of 2.4GHz is 109.99°. Whereas for Array-W1 and Array-W2, the phases are 83.64° at frequency of 2.42GHz and 63.16° at frequency of 2.44GHz. Thus, there is a phase shift of 26.35° occurred due to the incorporation of capacitor chips with capacitances of 1pF, 10pF, 10pF, and 1pF, and a phase shift of 46.83° for the capacitances of 0.5pF, 1.5pF, 1.5pF, and 0.5pF, relatively to the square patch antenna array without capacitor chips. Therefore, related to the beam reconfiguration characteristic, the phase shift could be controlled by varying the configuration of capacitance values.

Moreover, the comparison of radiation patterns for square patch array antennas without and with capacitor chips incorporation is observable in Fig. 5. Here, the radiation patterns are taken at all elevation angles (θ) with azimuth angle (ϕ) of 270° for E-plane, and at all azimuth angles (ϕ) with elevation angle (θ) of 0° for H-plane. From this point of view, it can be observed that there are two main lobes with a slightly different angle for each radiation pattern. Both radiation patterns for Array-W/O and Array-W1 are very similar, it looks almost exactly the same. Meanwhile, for Array-W2 shows a different shape of radiation pattern. The maximum beam for Array-W/O, Array-W1, and Array-W2 is

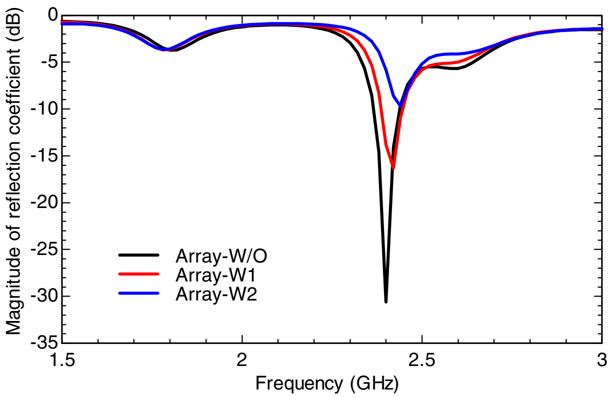


Fig. 3. Simulated result of reflection coefficient magnitude.

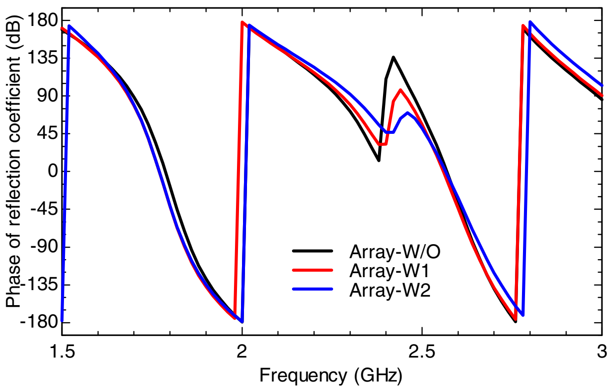
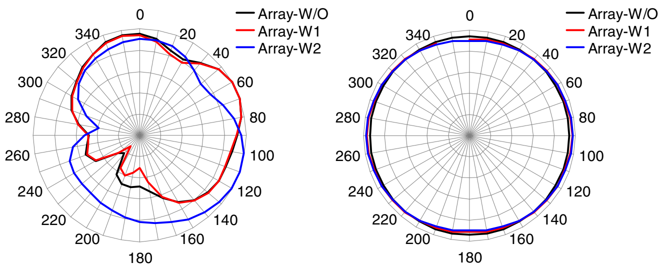


Fig. 4. Simulated result of reflection coefficient phase.



(a) E-plane (all θ, $\phi = 270°$) (b) H-plane (all ϕ, $\theta = 0°$)

Fig. 5. Simulated result of radiation pattern.

at 65°, 64°, and 113°, respectively. There is a slightly deflecting beam angle of 1° for Array-W/O and Array-W1, and 48° for Array-W/O and Array-W2. Hence, it can be concluded that the adjustment of capacitances values in a specific configuration is applicable to achieve the desired beam of antenna array. The capacitor chips incorporation of less than 1pF is most appropriate to significantly reconfigure the beam direction of antenna array.

B. Hardware Realization and Measurement

In order to demonstrate the feasibility of beam reconfiguration experimentally, a prototype of square patch array antenna is realized for characteristic measurement as shown in Fig. 6. The prototypes deployed on the FR4 Epoxy dielectric substrates are fabricated through wet etching technique. Figs. 7, 8, and 9 plot the measured reflection coefficient, i.e. magnitude and phase, and E-plane radiation pattern for 3 types of realized square patch antenna array, respectively. The minimum measured reflection coefficient for Array-W/O shown in Fig. 7 is -17.18dB at frequency of 2.45GHz. While for Array-W1 and Array-W2, the measured reflection coefficients are -9.68dB at frequency of 2.46GHz and -7.46dB on 2.48GHz, respectively. It shows that the measured resonant frequencies shift to the higher frequency compared to the simulation ones plotted in Fig. 3 which are mostly evoked by the dielectric constant for experimentation is lower than in the simulation. Whereas the worse value of measured reflection coefficient is almost influenced by the loss of dielectric substrate in experimentation which is higher than in the simulation. Unfortunately, some unexpected resonant frequency appears around frequency of 1.8GHz for all realized prototypes which is probably affected by the transmission lines at bottom side of dielectric substrate which act as radiators with lower resonant frequency.

(a) top side view (b) bottom side view

Fig. 6. Prototype of realized square patch antenna array for beam reconfiguration.

From the measured result of reflection coefficient phase depicted in Fig. 8, it shows that the measured phase of Array-W/O is 75.8° at frequency of 2.45GHz. While for Array-W1 and Array-W2, the phase shifts over Array-W/O are 81.48° at frequency of 2.46GHz and 73.89° at frequency of 2.48GHz. The discrepancy in measured results of reflection coefficient phase which are compared to simulation ones is mostly affected by the dielectric constant value and loss of dielectric substrate as explained above for measured results of reflection coefficient magnitude.

Furthermore, Fig. 9 shows the measured results of radiation pattern for all elevation angles with azimuth angle at 270°. It can be seen that the result revealed the change in the direction of maximum beam of each antenna, compared to the simulation results. The maximum beam direction for Array-W/O is 30°, which has more than 20° difference in angle compared to the simulated radiation pattern. While for Array-W1, it has maximum beam direction of 20° which discloses a 10° deflected beam over Array-W/O. The greater deflected beam of 20° is achieved by Array-W2 with the maximum beam direction of 10°.

Although the measurement shows different results compared to the simulation, it is evident about the role of capacitor chips in beam steering of antenna array. The array of antenna with lesser value of capacitances is able to deflect the beam angle greater than the higher value of capacitances. The incorporation of capacitor chips with value less than 1pF is considered as the key factor in beam steering process however it should be paid by the poor performance of impedance matching achievement which affects to the worse of reflection coefficient value.

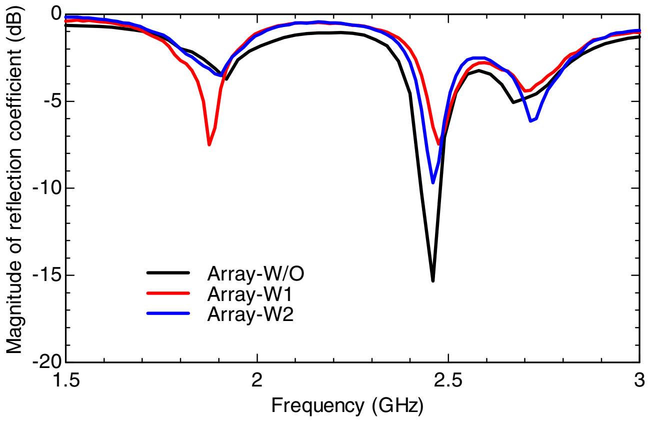


Fig. 7. Measured result of reflection coefficient magnitude.

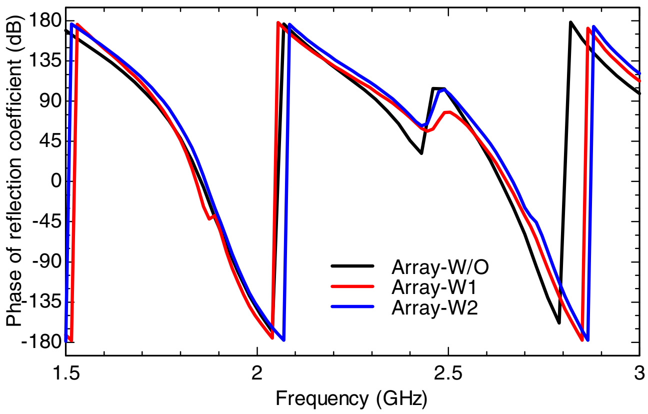
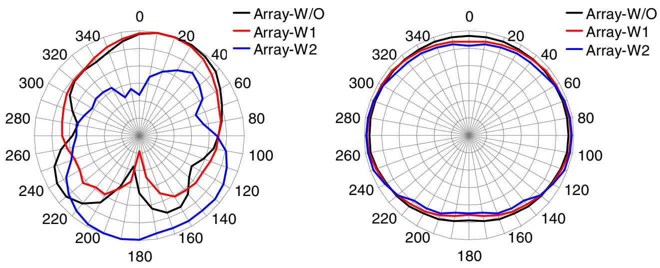


Fig. 8. Measured result of reflection coefficient phase.



(a) *E*-plane (all θ, $\phi = 270^{\circ}$) (b) *H*-plane (all ϕ, $\theta = 0^{\circ}$)

Fig. 9. Measured result radiation pattern.

IV. Conclusion

Beam reconfiguration of square patch antenna array has been investigated by incorporating capacitor chips with some configuration of capacitance values to obtain the desired beam direction. Four capacitor chips have been incorporated to the microstrip lines of feeding network with different capacitance value. From the characterization and measurement results, it has been confirmed that the capacitor chips incorporation has affected the characteristics of proposed antenna array. Capacitor chips with capacitance values higher than 1pF could only steer the beam direction of antenna array very slightly. On the other hand, the capacitor chips with capacitance values less than 1pF has had the ability to change the beam direction significantly, however, the antenna array would have a less favorable performance of impedance matching achievement. Therefore, by revamping the capacitance values of capacitor chips, beam reconfiguration of antenna array could be developed into more significant results.

References

[1] R. Horak, *Telecommunications and Data Communications Handbook*, 2nd ed., NJ: John Wiley & Sons Inc., 2008.

[2] S. H. Green, *The Irwin Handbook of Telecommunications*, 4th ed., NY: McGraw-Hill, 2000.

[3] T. S. Rappaport, *Wireless Communications: Principles and Practice*, 2nd ed., NJ: Prentice-Hall, 2002.

[4] D. M. Pozar, *Microwave and Rf Design of Wireless Systems*, 1st ed., NJ: John Wiley & Sons Inc., 2000.

[5] J. T. Bernhard and C. Balanis, *Reconfigurable Antennas (Synthesis Lectures on Antennas and Propagation)*, 1st ed., Morgan & Claypool Publishers, 2006.

[6] J. Volakis, *Antenna Engineering Handbook*, 4th ed., NY: McGraw-Hill, 2007.

[7] S-J. Ha, Y-B. Jung and C. W. Jung, "Reconfigurable beam-steering antenna using dipole and loop combined structure for wearable applications," *ETRI Journal*, Vol. 34, No. 1, pp. 1–8, Feb. 2012.

[8] W. Kang, M. G. Kim, J-H. Lee, K. Bahng and K. Kim, "A beam reconfigurable antenna using MEMS switches," in *Proceeding of IEEE Antennas and Propagation Society International Symposium (APSURSI)*, Charleston, USA, Jun. 2009, pp. 1–4.

[9] D. M. Pozar and D. H. Schaubert, *Microstrip Antennas: The Analysis and Design of Microstrip Antennas and Arrays*, 1st ed., NJ: Wiley-IEEE Press, 1995.

Analysis performance of DVB-H Over High Altitude Platform Station System (HAPS)

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Abstract—**Digital Video Broadcast-hanheld (DVB-H) can make mobile phones not only as a communication tool, but also be able to access multimedia services. DVB-H is transmitted through terrestrial and satellite infrastructures. Receiver receiving frequent loss because the dominant rayleigh channel conditions over terrestrial systems, so it need an innovation infrastructure that can support better performance. Recommended method of innovation infrastructure is HAPS. HAPS is a system that has the majority of the state of Line of Sight (LOS), and also has a non-LOS components, so that rician channel used. This thesis will analyze the effect of the use of HAPS on DVB-H technology. In this study, the effect will be measured at 2K, 4K, and 8K mode, different receiver speed, and the influence of modulation 16 QAM and 64 QAM on HAPS systems. Based on the research results, the performance of DVB-H technology on HAPS systems, working on mode 8K has highest value BER and Eb/N0 among the other working mode. User speed decrease performance. 16 QAM modulation has better performance than 64 QAM modulation.**

***Keywords*: HAPS, Rician, DVB-H, doppler frequency, Bit Error Rate (BER), E_bN_0, 16QAM, 64 QAM.**

I. INTRODUCTION

Media broadcast services based on telecommunications digital technology experiencing significant growth and potential in the past 10 years since the digitalization broadcast television. One technology that can support broadcast services received by mobile phone with good performance is Digital Handheld Video Broadcast (DVB-H) technology. DVB-H is an optimization of the Digital Video Broadcasting-Terrestrial (DVB-T) for existing digital TV and is intended to support the reception of broadcasting services on small terminals (hanheld) battery power in case of still or moving. DVB-H technology in 2004 standardized ETSI (ETSI EN 302 304 November 2004).

DVB-H technology is transmitted through terrestrial and satellite infrastructures, with the dominant rayleigh channel conditions and delay for satellite so receiver often receive loss. This technology require innovation infrastructure which can support service and has flexibility and better performance. One of recommended innovation infrastructure is *High Altitude Platform Station* (HAPS), which is also known as stratospheric repeaters [WRC-122, 97], [ITU-F.592, 02]. HAPs is defined by the World Radio Communications Conference (WRC) in 1997, and Radio Regulations (RR) No.S1.66A as "station located on an object with a height of 20-50 km that has the specifications, quantity, and relative coordinates fixed to the earth" [WRC-122, 97]. HAPS is a system that runs the majority state of Line of Sight (LOS), and also has a non-LOS components, then channel model is rician channel. At rician channel model, one of the main parameters which serve as an indicator of the state of the channel is used as a signal propagation medium is rician factor (K factor).

Modeling experiments conducted with some of the characteristics of this type of communication elements such as modulation techniques 16 QAM and 64 QAM. Then we will see 2K, 4K, 8K and doppler effect influence. Then BER performance of the system will be tested. BER performance will determine the coverage HAPS. In this thesis, the value of tolerance is still achievable BER is equal to 2.10^{-4}.

II. HIGH ALTITUDE PLATFORM STATION BASIC CONCEPT

Specification used in the HAPS system is strongly influenced by height of the platform, and the elevation angle between the platform and the ground stations, as well as the dimensions of the earth. HAPS coverage is a function of the height (h), the earth radius (R) is 6370 km, and the elevation angle (α) formed between platform and the ground station. Diameter (d) of the coverage area of the HAPS obtained from the calculation:

$$d = 2R\left(cos^{-1}\left(\frac{R}{R+h}.\cos\alpha\right) - \alpha\right) \quad (1)$$

The Doppler effect is the effect of the transmitted signal frequency shift due to the movement of user. Before the signal arrives at the receiver, the signal will go through a variety of pathways, causing a shift frequency. The mathematical equation of the Doppler frequency shift is as follows,

$$f_d = f_c \frac{v}{c}\cos(\alpha_n) \quad (2)$$

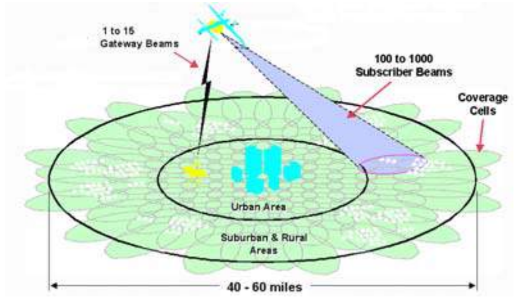

Fig. 1 HAPS coverage.

where f_d is *Doppler* frequency (Hz), f_c is frequency signal *carrier* (Hz), c denotes speed of light (m/s), and α_n is an incidence angle to the direct movement of the *user* signal.

Rician channel is the channel that describes the condition of LOS between transmitter and receiver and also a non-LOS component is smaller than the effect of multipath fading environment conditions, especially at small elevation angles. The following equation rician distribution is,

$$P(R) = \frac{R}{\sigma^2}^{\frac{(R^2+A^2)}{2\sigma^2}} Io\left(\frac{AR}{\sigma^2}\right) \text{; for (A≥0, R≥0)} \quad (3)$$

where R is the received signal, A is the amplitude of the dominant LOS signal, σ^2 is the variance of the average power of the multipath components and Io (.) Is the modified Bessel function by the first kind and zero order. One of the factors that affect is parameter K. K is defined as the ratio of the dominant LOS signal stronger and stronger average signal multipath components. PDF rician with K factor can be seen in the following equation,

$$p(R) = \frac{(K+1)R}{E[R^2]} \exp\left[\frac{-\{(K+1)R^2+KE[R^2]\}}{E[R^2]}\right].Io\left(2R\sqrt{\frac{K(K+1)}{E[R^2]}}\right) \quad (6)$$

Where,

$$E[R^2] = \frac{(A^2+2\sigma^2)}{2} \quad (7)$$

III. Introduction to Digital video broadcasting-Handheld (DVB-H)

DVB-H technology in 2004 standardized ETSI (ETSI EN 302 304 November 2004). The use of DVB-H standard allows the ability to receive signals that allow moving receiver to receive services well. In the DVB-H, a transmitter input MPEG-2 TS (Transport Stream) multiplex, which can consist of a TV program and there is also voice and data. DVB-H system using DVB-T standard, but there are additional elements in the link layer (layer above the physical layer). The addition of elements such as time slicing and coding forward error correction [5]. The working mode is optional DVB-H 4K mode but still supports 2K and 8K. The block scheme of DVB-H transmitter can be seen in Figure 2.

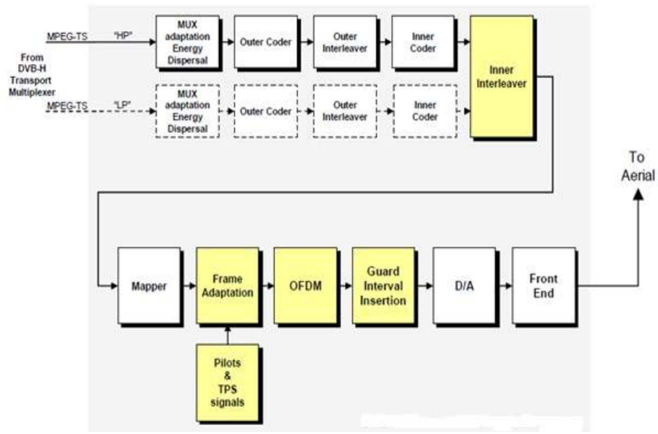

Fig. 2 DVB-H Adapter.

$$K = \frac{A^2}{2\sigma^2} \quad (8)$$

The relationship between the sampling frequency (τ), the number of subcarriers (NSC), and the symbol period (Tu) is as follows [20],

$$Tu = N_{sc}\ x\ \tau \quad (10)$$

Inverse discrete Fourier transform *(IDFT)* can be used to modulate the data in order to OFDM symbol. The mathematical equations of the IDFT is as follows [20],

$$y(n\tau) = \frac{1}{N_{sc}} \sum_{k=0}^{N_{sc}} X(k) e^{j2\pi kn\tau / N_{sc}} \quad (11)$$

While to restore the data symbol to OFDM symbol used discrete Fourier transform (DFT). The mathematical equations of the DFT is as follows,

$$X(k) = \frac{1}{N_{sc}} \sum_{n\tau=0}^{N_{sc}} y(n\tau) e^{-j2\pi kn\tau / N_{sc}} \quad (12)$$

The base period used in the DVB-H technology according to the following equation [20],

$$T = \frac{7}{8\ x\ Bw} \quad (13)$$

Where T is period on the basis of DVB (μs) and Bw is DVB Bandwith (5, 6, 7,atau 8 MHz).

IV. Design and Simulation

Simulations were performed at Digital Video Broadcasting Hanheld over HAPS systems. In this simulation, MATLAB (Matrix Laboratory) R 2011 is used. In this design, the receiver will be on two conditions, when the receiver is stationary condition (v = 0 km / h) and a receiver in motion (v = 5 km / h, v = 40 km / h, and v = 80 km / h). Design parameters can be seen in Table I.

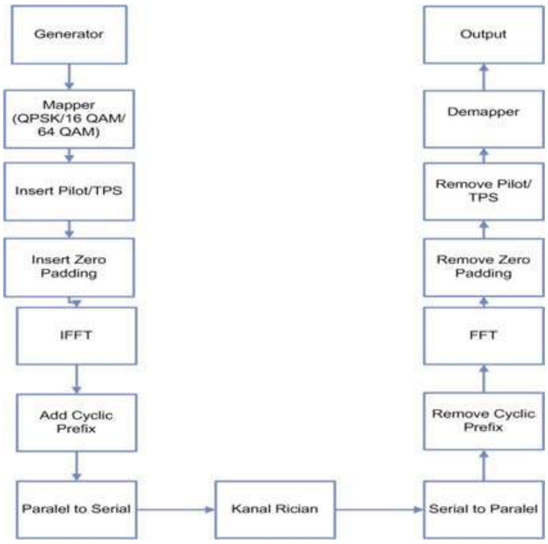

Fig. 3 Simulation Diagram.

TABLE I DVB-H ON HAPS PARAMETER.

Mode	2K	4K	8K
FFT size	2048	4096	8192
Guard Interval	¼	¼	¼
Mapping	QPSK, 16 QAM, 64 QAM	QPSK, 16 QAM, 64 QAM	QPSK, 16 QAM, 64 QAM
Bandwidth	8 Mhz	8 Mhz	8 Mhz
Perioda dasar	7/64 µs	7/64 µs	7/64 µs
Iteration	50	50	50
Jumlah simbol OFDM	4	4	4
Channel	Rician	Rician	Rician
Carrier Frequency	2.4 GHz	2.4 GHz	2.4 GHz

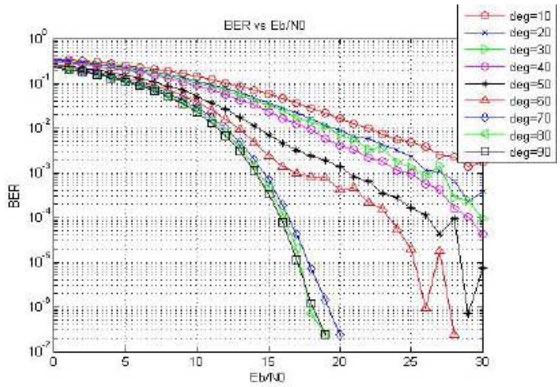

Fig. 4 2K 16 QAM.

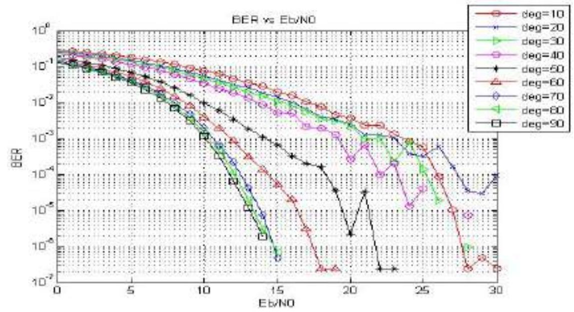

Fig. 5 4K Mode 16 QAM.

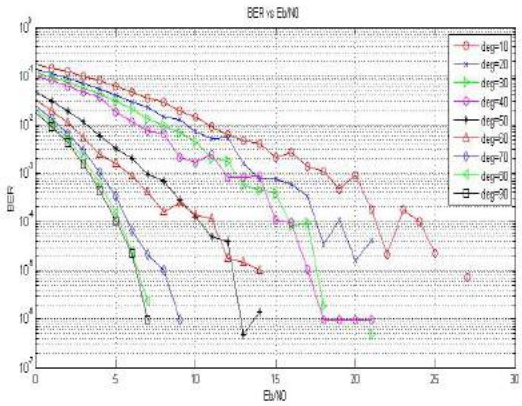

Fig. 6 8K mode 16 QAM.

III. ANALYSIS AND SIMUATION OF DVB-H OVER HAPS

The simulation result shows that the Performance comparison between mode 2K, 4K, 8K for 16QAM are as follow.

A. 2K 16 QAM

In the 2K mode at figure 4 using 16 QAM modulation can be seen that BER DVB-H standards is 2.10^{-4} for elevation angles $20^0 – 90^0$. In 2K mode, coverage HAPS is 108 km.

B. 4K 16 QAM

In the figure 5 4K mode using 16 QAM 16 QAM modulation can be seen that BER performance standards DVB-H BER is 2.10^{-4} for elevation angles 10^0-90^0. In the 4K mode with 16 QAM modulation, HAPS coverage is 215 km (diameter).

C. 8K mode 16 QAM modulation.

In the figure 6 8K mode using 16 QAM 16 QAM modulation can be seen that BER performance BER standards DVB-H which is on the corner of elevation 2.10^{-4} is 10^0-90^0. In the 4K mode with 16 QAM modulation, HAPS coverage is 215 km (diameter).

The Performance comparison of BER for 4K mode and for receiver (v=0), pedestrian (v=5 km/h), and v = 40 km/h and for v=80 km/h using 16 QAM are as follow,

A. For 4K Modulasi 16 QAM v=5 km/h

In figure 7 4K mode using 16 QAM modulation can be seen that BER performance BER standards DVB-H which is 2.10^{-4} on the corner of elevation $30^0 – 90^0$. Coverage HAPS is 68.7 km.

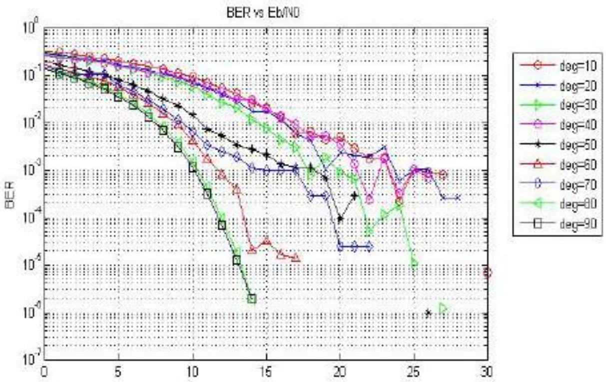

Fig. 7 BER vs Eb/N0 mode 4K QAM v= 5 km/h.

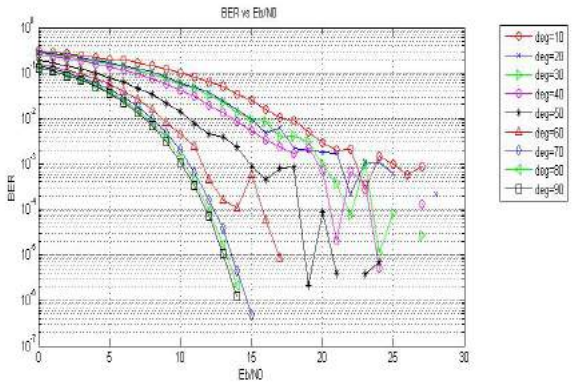

Fig. 8 BER vs Eb/N0 mode 4K 16 QAM v=40 km/h.

B. Mode 4K using 16 QAM v=40 km/h

In figure 8 4K mode using 16 QAM modulation can be seen that BER performance BER standards DVB-H which is 2.10^{-4} on the corner of elevation 30^0 – 90^0. Coverage HAPS is 68.7 km(diameter).

C. For 4K Modulasi 16 QAM v=80 km/h

In figure 9 4K mode using 16 QAM modulation can be seen that BER performance BER standards DVB-H which is 2.10^{-4} on the corner of elevation 40^0 – 90^0. Coverage HAPS is 48 km(diameter).

The Performance Comparison BER Mode 4K for receiver (v=5 km/h), v=40 km/h dan v=80 km/h pada Modulasi 64 QAM are as follow,

A. For 4K using 64 QAM pada v=5 km/h

In figure 10 4K mode using 64 QAM modulation can be seen that BER performance BER standards DVB-H which is 2.10^{-4} on the corner of elevation 40^0 – 90^0. Coverage HAPS is 48 km(diameter).

B. For Mode 4K using 64 QAM pada v=40 km/h

In figure 11 4K mode using 64 QAM modulation can be seen that BER performance BER standards DVB-H which is 2.10^{-4} on the corner of elevation 60^0 – 90^0 coverage HAPS is 22.9 km(diameter).

C. For 4K Modulasi 64 QAM pada v=80 km/h

In figure 12 4K mode using 64 QAM modulation can be seen that BER performance BER standards DVB-H which is 2.10^{-4} on the corner of elevation 70^0 – 90^0.

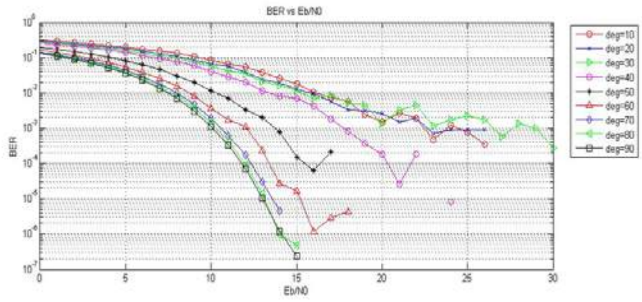

Fig. 9 BER vs Eb/N0 mode 4K 16 QAM v=80 km/h.

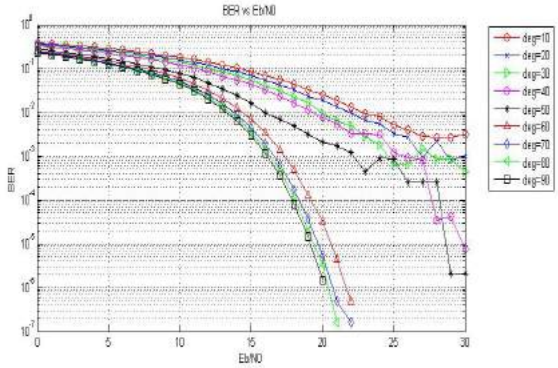

Fig. 10 BER vs Eb/N0 mode 4K 64 QAM v=5 km/h.

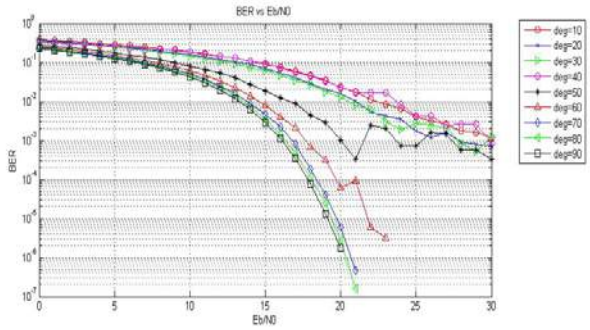

Fig. 11 BER vs Eb/N0 mode 4K 64 QAM v=40 km/h.

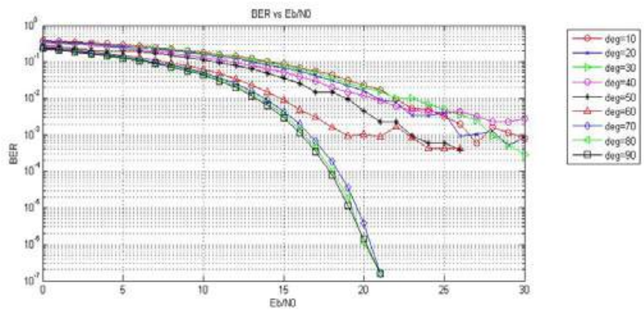

Fig. 12 BER vs Eb/N0 mode 4K 64 QAM v=80 km/h

Coverage HAPS is 14.5 km(diameter).

V. Conclusion

We have evaluated the performance of DVB-H over high

[7] Iskandar (2006). Channel Characterization and Performance Evaluation of Mobile Communication

Employing Stratospheric Platforms. IEICE Trans. COMMUN.School of Global Information and Telecommunication Studies (GITS) Waseda University. Japan .

altidute platform channel. For 2k mode scheme, 4K, and

[8] J.G Proaksis, Digital Communications, four ted., Mc. Graw-Hill, New York. 2000

[9] http://www.dvb-h-online.org/PDF/DIgiTAG-DVB-H Handbook.pdf

[10] Krzysztof Wesolowski, "Mobile Communication Systems", John Wiley and Sons. 2002.8K with modulation 16 QAM, 64 QAM and simulation results obtained that performance changes in various modes with the best performance in 8K mode. Coverage area of 16 QAM is larger (205 km) than 64 QAM (48 km). At different speed of user v = 5 km / h, v = 40 km / h, and v = 80 km / h show a decrease in BER vs Eb/N0 performance, with the better performance in the 16 QAM modulation than 64 QAM at various speeds.

REFFERENCES

[1] Alejandro Aragon dkk, *High Altitude Platform for Wireless Communications. Jhon Wiley and Sons.* United Kingdom. 2008

[2] David Cordero Diaz, "DVB-T Pilot Sensing Algorithms", Final Project in Linkoping University Departement of Electrical Engineering, Communications System Division, Linkoping, Juni 2010.

[3] dukemil.bme.duke.edu

[4] European Telecommunication Standard ETSI 300 744 v 1.5.1 "Digital Video Broadcasting Hanheld (DVB-H) : Framing Structure, channel coding , and modulation for digital terrestrial television"

[5] Gerard Faria, Jukka A. Henrikson, Erik Starq, and Pekka Talmda, "DVB-H : Digital Broadcast Services to Hanheld Devices, " proceedings of the IEEE transaction, vol. 94, no.1, January 2006.

[6] Jyri T.J. Penttinen, Petri Jolma, Erkki Aaltonen, and Jani Va're. "The DVB-H Handbook. The Functioning and Planning of Mobile TV",1' ed, John Wiley and Sons, 2009.

[11] Polak, Ladislav, Tomas Kratochvil."DVB-T/H Digital Television Transmission and its Simulation over Ricean and Rayleigh Fading Channels".Department of Radio Electronics, Brno University of Technology.Brno-Checz Republic.

[12] Samhany, Achmad Husnan (2010). "Pengembangan Equalizer DVB-T/H Untuk Aplikasi High Mobility". Program Studi Teknik Telekomunikasi Sekolah Teknik Elektro dan Informatika Institut Teknologi Bandung. Bandung-Indonesia.

[13] Schulze, Henrik , Christians Luders. "Theory And Applications of OFDM and CDMA". John Wiley and Son, West Sussex-England.ch 3 – 4 [3]Welch, Greg ,Gary Bishop (2006) . "An Introduction to the Kalman Filter".Department of Computer Science University of North Carolina, Chapel Hill.

[14] Setiawan Eddy, Wahana Terestrial Masa Depan, http://elektro Indonesia.com / elektro/assi0500.htmlASSI news-letter. Mei 2008.

[15] She, Yushi, Ed Martinez (2006). "Channel Estimation in OFDM ".FreescaleSemiconductor Application Note.

[16] Struzak. Ryszard, "Mobile Telecommunications via Stratosphere," International Communications Project, 2003.

[17] T.C. Tozer and D. Grace, "High Altitude Platforms for Wireless Communications" IEE Elec. And Commun Eng. J.,vol 13, no.3, pp. 127-137, June 2001.

[18] www.meted.uca.edu

[19] ----------------," Digital Video broadcasting (DVB): Framing structure, channel coding and modulation for digital terrestrial television ", ETSI EN 300 744 V1.6.1 (2009-01).

Performance Study of DVB-T Channel Estimation to Improve HAPS Coverage

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Abstract-DVB-T (Digital Video Broadcasting-Terrestrial) is an open standard in digital video transmission which has been internationally recognized. DVB-T standard issued by ETSI. This research will analyze the effect of DVB-T channel estimation for coverage improvement on High Altitude Platforms Station (HAPS)system. HAPS are known as stratospheric broadband telecommunications infrastructure is a standard similar to the satellite, which was placed in the stratosphere. HAPs channel can provide LOS relationship between HAPS platform with the user so that it can be simulated with rician channel. Based on the research results, the performance of DVB-T services in the HAPS system is influenced by the type of modulation used and the mode of work, the influence of channel estimation using least squares (LS) to increase coverage up to 34.4 km with a value of Eb/No 20 dB.

Keywords—DVB-T, HAPS, Rician, Channel Estimation.

I. INTRODUCTION

The growing need for telecommunications services requires the development of telecommunications infrastructure. High Altitude Platforms Station (HAPS) is an alternative solution that is able to provide multiple services on a single platform and can provide many advantages compared with other existing technologies (terrestrial and satellite). HAPS utilize transceiver station at altitudes well below the satellite, which is in the stratosphere [1].

DVB-T is a digital TV broadcasting standard which enables the delivery of terrestrial digital TV programs and other multimedia streaming [2]. With all the advantages offered, until now Indonesia has continued migrating from the use of analog TV to digital TV. This research will discuss about the performance of DVB-T on channel HAPS with areas that are still able to receive DVB-T with tolerance values specified BER [3]. HAPS is simulated with rician channel which is containing LOS and non-LOS components that have characteristics in each position / distance from HAPS platform, will further analyze the techniques that can increase the coverage [4]-[8]. This paper consists of an introduction, basic concepts of HAPS and DVB-T, then the system design and simulation results and analysis.

II. HIGH ALTITUDE PLATFORM STATION(HAPS)

A. *HAPScharacteristics*

Characteristics of the earth and height dimensions that are used on HAPS systems greatly affect the distance of the elevation angle of the earth and HAPS. LOS estimation quality also depends on the shape of the land.Diameter calculation can be referenced by the following equation:

$$d = 2R\left(cos^{-1}\left(\frac{R}{R+h}\,.\cos\alpha\right) - \alpha\right) \qquad (1)$$

Where d is the diameter of the area that is formed, R is the radius of the earth is 6370 km, h is the height of HAPs and α is the elevation angle in degrees formed between the earth station and the HAPS platform [9].

B. *Channel modeling HAPS*

This research will useRician channel model assumed LOS signals from HAPS directly to the receiver with few multipath. Attenuation factor due to the influence of weather and natural gas are also ignored because the use of low-frequency 2.4 GHz.

C. *Rician Channel*

Rician channel representation is most appropriate for small applied considering the influence of the multipath fading of the state of the environment, especially at small elevation angles [9]-[11]. At this situation, random multipath components coming from different angles can affect the signal received. Rician distribution is given as:

$$P(R) = \frac{R}{\sigma^2}^{-\frac{(R^2+A^2)}{2\sigma^2}} I_0\left(\frac{AR}{\sigma^2}\right), \text{ for } (A\geq 0, R\geq 0) \qquad (2)$$

$$P(R) = 0 \;\; ; \text{ for } (r < 0) \qquad (3)$$

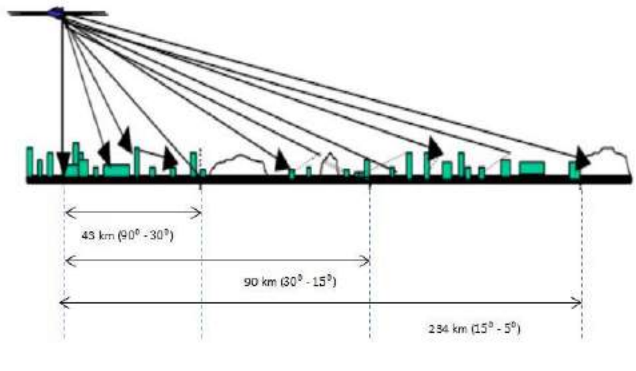

Fig. 1 Illustration corner of the HAPS coverage.

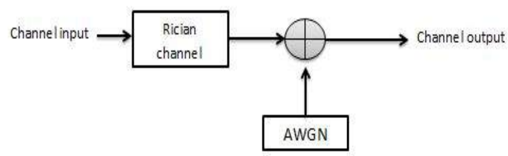

Fig. 2 HAPS channel simulated.

where A represents the amplitude and I_0 (.) Is the modified Bessel function by the first kind and zero order. One of the parameters that influence the parameter K, which is defined as the rate of a channel Rician LOS observed.

D. *Additive white Gaussian noise (AWGN)*
In a communication system signal always suffered damage caused by the noise caused by the amplifier. Damage caused by a signal only white Gaussian noise (AWGN), and the received signal using statistical deterministic AWGN (Gaussian distributed).

III. DIGITAL VIDEO BROADCASTING-TERRESTRIAL (DVB-T)

DVB-T is known as a digital television broadcast standard that is widely used in the world and is also being adapted in Indonesia due to several advantages, mainly due to reliability DVB-T is capable of delivering large amounts of data at high speed point-to-multipoint. DVB-T system which has direct broadcasting from a transmitter system Earth (terrestrial) to a receiver. Earth transmitter function is to transmit MPEG-2 digital data that has been modulated into a wave VHF / UHF antenna to be transmitted using a transmitter. Digital modulation system used in the DVB-T system is modulated OFDM (Orthogonal Frequency Division Multiplex) with a choice of modulation types QPSK, 16-QAM or 64QAM. By using this system, the bandwidth used (about 6 to 8 MHz) may be efficient to allow the use of the channel for some content.

A. Digital Modulation
Digital modulation is the process of laying the digital signal to the carrier signal. There are many types of digital modulation that can be used in digital signal transmission system, the following is a type of modulation to be used in this study,

a. Quadrature Phase Shift Keying (QPSK*)*:
Phase shift keying modulation is a digital modulation technique whereby the QPSK modulation using 90 increments in the phase which produces 4 kinds of logic state (symbol) in any condition. one symbol represented in two bits. QPSK modulation to increase the data transmission rate is higher than the binary phase shift keying (BPSK) in the same bandwidth but requires QPSK generation and detection systems are more complex.

b. Quadrature Amplitude Modulation (QAM)
Digital modulation Quadrature Amplitude Modulation (QAM) is a combination of ASK and PSK digital modulation which bit binary data expressed with different amplitude and phase.

B. Orthogonal Frequency Division Multiplexing (OFDM)
OFDM is a spread spectrum technique which uses multicarrier systems that transmit data in parallel on multiple subcarriers are mutually orthogonal [12]. OFDM is emerging as a solution to solve the problems in the propagation of waves in the air. Orthogonality concept of a signal can make the signal overlapping each subcarrier without having the effect of inter-symbol interference (ISI) and inter-carrier interference (ICI) which in turn will lead to efficient use of bandwidth [12].

OFDM technique used in the DVB-T transmission, with the parameters specified number of subcarriers corresponding working mode used in DVB-T transmission [13]. The parameters for DVB-T 2K and 4K modes are used as in Table 1.

C. Least square channel estimation
Channel estimation is a technique that aims to predict or estimate the channel impulse response (CIR) or impulse response of a channel to the signal sent. In general, there are many methods that can be used in channel estimation, the signal sent to the effects the changes generated by the channel was estimated that the detection signal becomes more accurate information. Estimates Least Square (LS) channel estimator is the simplest, this estimator works by dividing the received signal at the receiver after the channel through which the signal is transmitted through the channel before. Mathematically expressed as follows:

$$HLs = \frac{Y}{X} \qquad (4)$$

note:
HLS : Channel frequency response is obtained by least square estimate
Y : Signal at the receiver, expressed in the frequency domain
X : Signal at the transmitter, expressed in the frequency domain

TABEL 1. OFDM PARAMETERS OF THE DVB-T.

Mode	2K	8K
The number of subcarriers used	1705	6817
Number of K_{min}	0	0
Number of K_{max}	1704	6816
Sub-*carrierspacing*	1/(2048 x T)	1/(8192 x T)
Spacing between*carrier*	*Bandwidth* DVB	

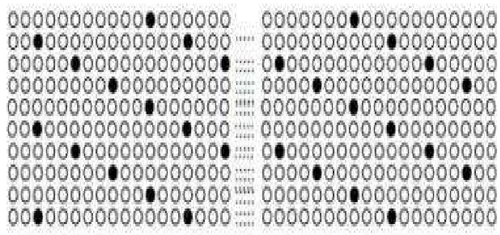

Fig. 3 Pilot location.

Estimation method is also the method of least squares estimation of pilot assisted channel estimation works by transmitting pilot symbols which is a bit stream that had previously been known to coincide with the data sent information. Pilot symbol was used to determine the

TABEL 2 VALUES OF K-FACTOR.

Elevation	K-factor
10	1.4
20	2.0
30	2.3
40	2.7
50	4.6
60	6.4
70	9.2
80	12.2
90	16.8

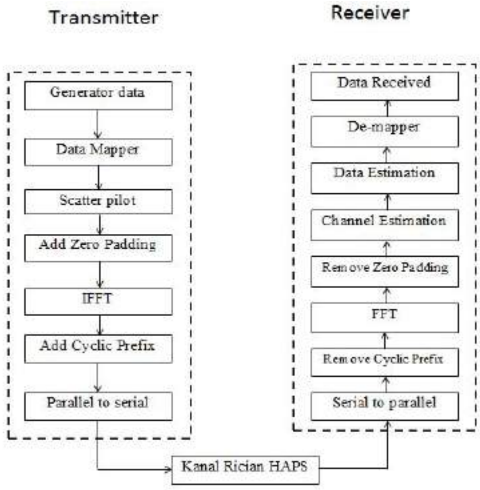

Fig. 4 Design System.

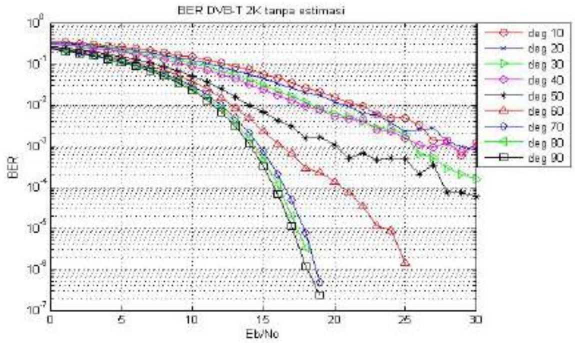

Fig. 5 Graph BER QPSK DVB-T 2K without estimation.

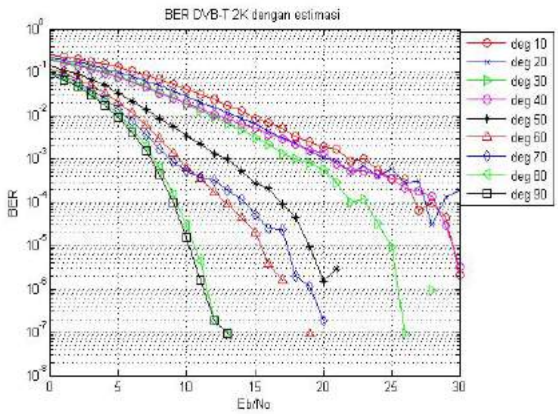

Fig. 6 Graph BER QPSK DVB-T 2K with estimation.

pattern of change is happening, then the pattern of these changes can be known impulse response channels. Basically pilot symbols used to have three types of pilot arrangement structure of which is the type of block, comb, and lattice. DVB-T is one of the services that use the lattice structure of the pilot arrangement. At this lattice type pilot arrangement, each OFDM symbol has a pilot signal periodically placed both in frequency and in time domain [14].

IV. SYSTEM DESIGN

In the simulation session will be an analysis of the performance of the DVB-T system on HAPS indicated by the BER values obtained for Eb/No. subsequently saw an increase in coverage is influenced by the use of channel estimation. To facilitate the process of simulation, then some of the assumptions made are as follows:

1. Conditions HAPs and receivers are considered in a static position, HAPS with a height of 20 km.
2. Doppler effect of the environment has no effect.
3. Simulation at the physical layer baseband signal.

Simulation will be conducted at several receiver positions angles. At each corner are observed have a different values of K-factor. K-factor values used in the simulations to follow the results of experiments that have been done previously. The value of K-factor at any elevation angle contained in Table 2. Generally, the system consists of a transmitter block, transmission channel and receiver blocks. Furthermore, from the data obtained de-mapper block output will be calculated bit error rate (BER), BER calculation is intended to determine the number of bits that have errors due to noise mixed. This calculation is done by comparing the data received by the receiving data transmitted by the transmitter. BER is a parameter used in the simulation to see the performance of the system.

V. ANALYSIS OF SIMULATION

A. *Performance of services by using the DVB-T channel estimation on HAPS*

Performance of the overall system be based on the BER values obtained of bit energy per noise value (EbNo) system. Good system performance will be shown with the smaller value of the BER. Fig. 5 shows the BER results obtained from the simulation of the DVB-T 2K with QPSK modulation on HAPS channel without using channel estimation. The results showed that the best BER graph is when the highest value of K-factor, which means the receiver is at elevation angles90^0. The position is appropriate under the platform so that the maximum transmit power received. This is also indicated by the value of K-factor on the LOS component.

Fig. 6 shows the results obtained from the BER simulation 2K DVB-T with QPSK modulation on HAPS systems by adding the use of LS channel estimation on the block to improve BER can be achieved. In the Fig. 7 will show simulation results on DVB-T 8K QPSK, which

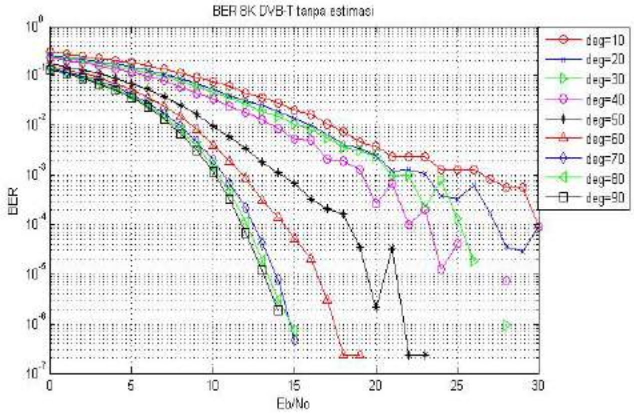

Fig. 7 BER of QPSK DVB-T 8K without estimation.

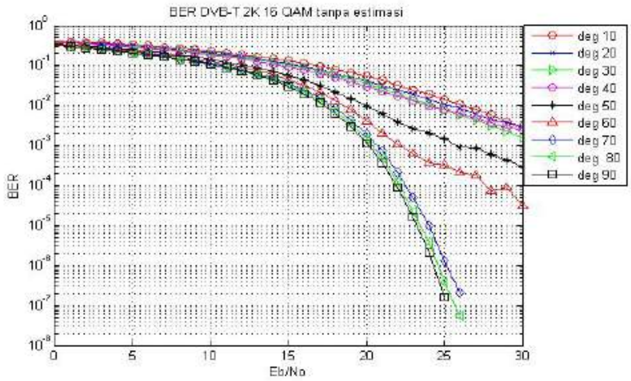

Fig. 9 BER of 16-QAM DVB-T 2K without estimation.

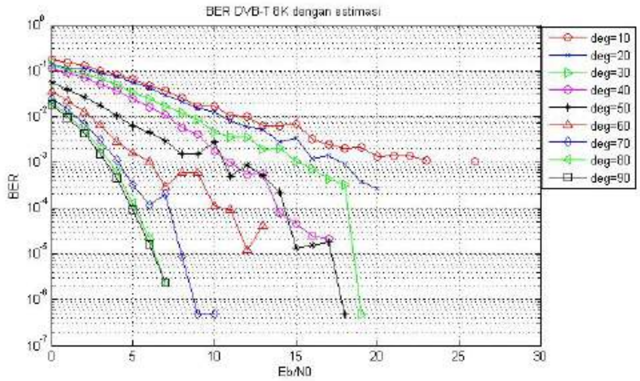

Fig. 8 BER of QPSK DVB-T 8K with estimation.

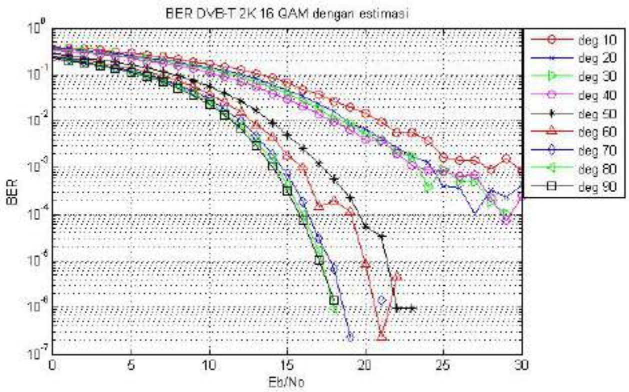

Fig.10 BER of 16-QAM DVB-T 2K with estimation.

TABLE III DIAMETER COVERAGE OF HAPS WITH AND WITHOUT ESTIMATION.

DVB-T		Diameter	
		Without estimation	With estimation
QPSK	2K	14.5 km	33.4 km
	8K	33.4 km	67.8 km
16 QAM	2K	1 km	22.9 km
	8K	14.5 km	33.4 km

Further simulations performed on the DVB-T 2K and 8K mode with 16 QAM modulation. Based on figure 9, graphics BER obtained still showed the same pattern where the smallest BER is on the corner of90^0but the value of BER10^{-4}can be achieved with a value of Eb/No is higher, thus indicating use QPSK modulation indicates better performance. It can be caused by the distance between the QPSK constellation symbols is greater than the distance between the constellation symbols on 16-QAM symbols errors resulting opportunities that occur on QPSK modulation is smaller than the chance of symbol errors that occur in 16-QAM modulation.In the figure 10 shows the BER values improved with the addition of LS channel estimation.

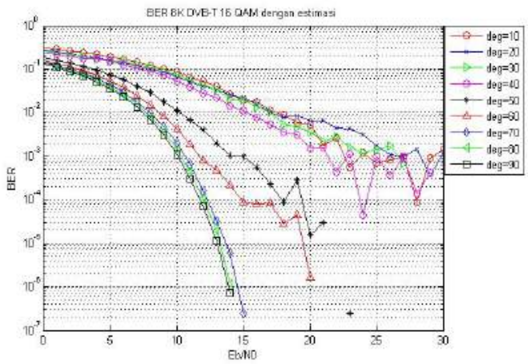

Fig.11 BER 16-QAM DVB-T 8K without estimation.

B. *Extensive analysis of the DVB-T service coverage on HAPS*

Extensive analysis of the DVB-T service coverage on channel HAPS can be determined from the simulation results that have been obtained. To calculate the area of coverage and the increase will be calculated on a value Eb/No with equation (1). The Eb/No maximum value used is 20 dB with the result that coverage can achieve BER values$10^{-4}$are shown in Table 3. Based on the simulation results, the mode is used DVB-T effect on the resulting performance. For service delivery area coverage is more suitable 8K mode, but this mode has a higher complexity with the use of subcarriers and FFT points more, for the simulation also takes a longer time.

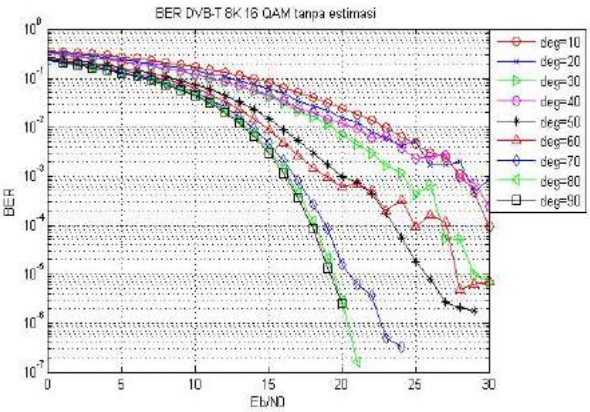

Fig. 12 BER 16-QAM DVB-T 8K with estimation.

VI CONCLUSION

The purpose of the study is to see the performance of Digital Video Broadcasting services - Terrestrial (DVB-T) in theHigh Altitude Platform Station (HAPS)system. Based on simulation results, showing a good performance, where the coverage of the BER tolerance10^{-4}reached a diameter of 68.7 km, the use of channel estimation DVB-T services greatly affect increase in coverage that can be achieved with the tolerances specified BER, the LS channel estimation services using the DVB-T 8K mode and QPSK modulation can increase coverage up to 34.4 miles on EbNo 20 dB.

REFFERENCES

[1] S.Karapantaziz, F.T Pavlidou," Broadband Communications Via High Altitude Platforms a Survey" IEEE Commun, vol. 7, no. 1, 2005.

[2] Brito Luciano, Jose, dkk. "Design, Simulation and Hardware Implementation of a Digital Television System: System Overview". IEEE Syhmposium on Spread Spectrum Techniques and Application.2006.

[3] ETSI EN 302 307 V1.1.1 (2004-06), "Digital Video Broadcasting (DVB) Second generation framing structure, channel coding and modulation systems for Broadcasting, Interactive Services, News Gathering and other broadband satellite applications". 2004-2006.

[4] Iskandar and S. Shimamoto, "Channel characterization and performance evaluation of mobile communication employing stratospheric platform," IEICE Transactions on Communications, vol. E89-B, no. 3, pp. 937-944, March 2006.

[5] Iskandar, "Reverse link CDMA system capacity evaluation for stratospheric platform mobile communications," Jurnal Makara Seri Teknologi, Universitas Indonesia, vol. 11, No. 2, pp. 49-58, November 2007.

[6] Iskandar and S. Shimamoto, "On the downlink performance of stratospheric platform mobile communications channel," in Proceedings of IEEE Global Telecommunication Conference (GLOBECOM), ISSN: 1930-529X, DOI: 10.1109/GLOCOM.2006.922, pp. 1-5, 2006.

[7] Iskandar and S. Shimamoto, "Analysis of CDMA capacity for multiple stratospheric platform mobile communications under imperfect power control and fading," in Proceedings IEEE Wireless Communications and Networking Conference (WCNC), ISSN: 1525-3511, DOI: 10.1109/WCNC.2007.468, pp. 2512 - 2516 , 2007.

[8] Iskandar, A. Kurniawan, E. B. Sitanggang, and S. Shimamoto, "Step Size Optimization for Fixed Step Closed Loop Power Control on WCDMA High Altitude Platforms (HAPs) Channel," in Proceedings of IEEE Global Telecommunication Conference (GLOBECOM), vol. 1, pp. 1-5, Dec. 2008.

[9] F. Dovis, R. Fantini, M.Mondin, and P. Savi, " Small-scale fading for high altitude platform (HAP) propagation channels, " IEEE J.Sel. Areas Commun., Vol. 20 ,No.3, pp.641-647, April 2002.

[10] Iskandar, A. Kurniawan, and H. Syaputra, "On the Evaluation of Fixed Step Closed Loop Power Control for CDMA High Altitude Platforms (HAPs) Communication Channel," in Proceedings of IEEE International Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC), vol. 1, pp. 1-5, Sept. 2008.

[11] Iskandar, A. Kurniawan, and M.E. Ernawan, "Performance of Variable Step Closed Loop Power Control in CDMA High Altitude Platforms Communication Channel," Proceedings of The 2010 IEEE 72nd Vehicular Technology Conference (VTC Fall 2010), DOI 10.1109/VETECF.2010.5594401, pp. 1-5, Sept. 2010.

[12] Cho, Y.S., Kim, J.,Yang, W.Y., Kang, C.G., " MIMO OFDM Wireless Communication with Matlab", John Wiley and Son, 2010.

[13] Osvaldo Simeone, "Pilot-Based Channel Estimation for OFDM System by Tracking the Delay-Subspace," IEEE Trans. Wireless Commun. Vol. 3, No. 1, pp. 315 – 325, January 2004.

[14] Prasad, Ramjee. OFDM for Wireless Communication.Universal Personal Communication. 2004.

Statistical Analysis on Aggregate and Flow Based Traffic Features Distribution

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Abstract – **Anomaly traffic detection is one method to detect attack in internet, especially Distributed Denial of Service (DDoS). Here, traffic analysis which reveal the observed traffic pattern is key important process as the anomaly decision was taken based on traffic analysis. This research analyzed several statistical analysis of traffic datasets categorized as normal, DDoS attack and flashcrowd. Analysis done in aggregate and per-flow traffic level which showed similarity and difference in each category. Windowing technic used to measure instantaneous statistical value. The result showed several statistical difference which could be used to categorized types of anomaly, especially to identify normal threshold. Flow to normal average flow distance distribution perform not followed Gaussian distribution.**

Key words: traffic, anomaly, DDoS, flashcrowd, distribution.

I. Introduction

DDoS attack is a kind of internet attack which had big impact for service provider and all user in network. High rate traffic from botnet made almost all users in network suffered in gaining the server service. Quality of Service (QoS) decreased as botnet traffic flooding the network. So its important to detect the DDoS attack to provide security mechanism to protect the network and service.

But high rate traffic not only came from DDoS attack. Hugh number of legal service request in flashcrowd also made the QoS decrease as there were limitation in hardware infrastructure. For example in breaking news, or product launching, increased number of legal user resulting high rate traffic for server. Anomaly traffic analysis is well known research to detect the anomalous traffic event.

Both event visually almost have the same high rate traffic. If only seen from visual analysis of these phenomena, there is possibility to wrongly classify the anomaly. So it's important to analyze in deeper analysis to avoid miss classification. It's because DDoS and flashcrowd must treated differently. From several solutions, not all anomaly detection proposal pay much attention to real traffic model. Past researches implement certain method based on believed traffic distribution. It made the solution limited to certain limited condition when proposed believed traffic distribution fulfilled.

This research was a pre-research of our traffic anomaly detection classification. We did several statistical analysis empirically of captured traffic in dataset. Data analysis done via windowed and streaming technic in aggregate and per-flow data. Normal aggregate traffic represented by packet trace network level in CAIDA anonymous 2014. Flashcrowd traffic reconstructed from binary server log application level from Worldcup 1998 dataset. DDoS aggregate traffic analysis done from raw packet trace in CAIDA 2007. For per-flow analysis, we did analysis on well known KDDCup 1999 dataset, which contain normal and attack labeled flows.

From this research, contributions were gained from several traffic statistical analysis in aggregate and flow level analysis. Statistical distribution in aggregate traffic showed similar distribution from flashcrowd and normal traffic, but different from DDoS attack. From per-flow analysis, normal flows still showed the same self similar long range dependent behavior same as aggregate traffic, but DDoS showed the opposite. Per-flow mahalanobis distance to average flow analysis provided information of distance behavior of normal and DDoS flow, which were not Gaussian normal distribution.

II. Related Work

Anomaly detection research focus to identify changes from normal patterns or behavior. The normal pattern obtained from daily historical or empirical experiment which believed to be normal traffic. Any pattern which not follow the known normal pattern will be identified as anomaly. Several method has already proposed in anomaly detection research, such as statistical, information theory, forecasting, data mining, etc.

Traffic modeling has been done in several research. In [1] empirically analyzed internet traffic and shifting the perception of some researchers who originally occupy Poisson distribution modeling into Pareto modeling in internet traffic model. Research [2] showed that the nature of self-similar behavior could be used as traffic model. Recently, distribution and analysis of traffic occupied in several traffic anomaly detection researches, especially DDoS detection.

Normal Hurst exponent for normal traffic is believed between 0,5 to 1, and followed by others Hurst exponent based anomaly detection. In [3] detected attacks by analyzing the nature of the Long Range Dependent (LRD) of internet traffic. Research in [4] used the properties of Self-Similarity in detecting DDoS attacks, which performance result depended on the estimated Hurst exponent value and sensibility system against congestion. FRT technique to estimate Hurst exponent used in [5] to estimate abnormal Hurst exponent value.

From the traffic features, several research did analysis from traffic aggregate based and flow based traffic data. The used of traffic aggregate based data especially to detect anomaly of DDoS and flashcrowd. And flow based data commonly used not only to detect the present of anomaly but also detect types of anomalous event. In [6] did comparison of clustering-based detection and histogram-based Pearson correlation based on the packet to detect anomaly. Statistical characteristic based on the traffic rate used to detect anomalies in [7]. Meanwhile, [8] and [9] separated the anomalous type of DDoS and flashcrowd with flow similarity and correlation function, even at a particular feature. Research that has detected and classified types of anomaly was found at [10] and [11] which used flow based traffic data.

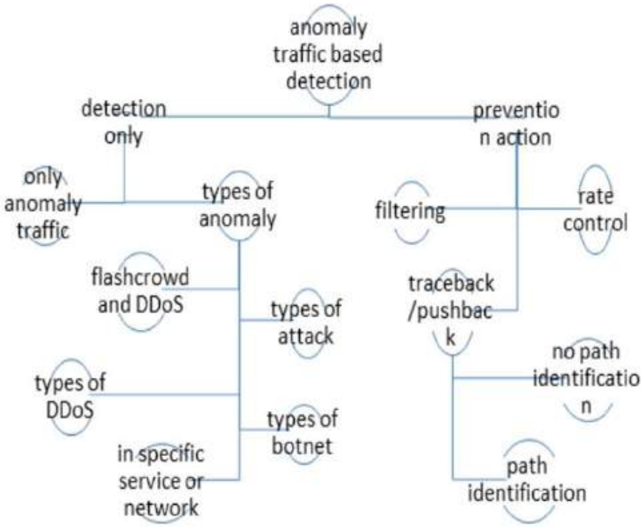

Figure 1 Traffic anomaly detection capability focus taxonomy.

In the development of traffic anomaly detection, research contribution made by each research have grown, as seen in taxonomy [12]. From several anomaly detection research based on traffic models, not many which tried to explored many of traffic features which can be extracted from data traffic. Generally, the methods developed based on one particular feature of the traffic (typically *lambda* or rate) which resulted in only able to detect anomalies in traffic but can not classify the type of anomalies. Though it could be seen that different features could result in different traffic analysis, which could clarify the types of anomalies that arise when analyzed further.

The used of feature-level traffic flow in [10], used covariance matrices of the many features (23 to 31) from KDD Cup 1999 to identify the type of attack. The results showed a very high accuracy in the analysis of the type of homogeneous traffic group. This research was studied further by [13] by adding the triangle area of the map features as an input traffic detection on all continuous features. This step done with per-flow analysis for the detection of attacks despite the heterogeneous traffic anomaly occured.

However, several anomaly traffic detection research only based on assumptions of used traffic features distribution. The consequences might be any limitations in implementation of the proposed solution, only when the underlying assumptions were fulfilled. So it is very important to analyze first the actual traffic feature distribution which appears in traffic features.

III. Methodology

This research carried out in stages as in Figure 2 as part of next research in types of anomaly classification research. In our next research, the detector will be placed at the target-end. So all trace in datasets must be selected that only lead to a specific destination address.

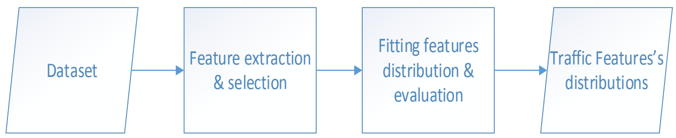

Figure 2 Framework

A. Features Extraction and Selection

Each dataset passed through the process of extraction and feature selection to determine the traffic features which suitably used as detection rule traffic characteristic. Such features were as follows.

1) Network level (packet trace) dataset, normal CAIDA anonymized 2014 , DDoS CAIDA 2007 [14] :
   - packet rate, number of data packets arrived at specific time intervals.
   - Delta, time between packets arrival.
2) Application level (log-server) dataset, flashcrowd WorldCup 1998 [15] :
   - packet rate.
3) Extracted features from normal and DDoS traffic in KDDCup 1999 [16], which mainly focus on traffic features which are feature number 23 and 24 :
   - 23 (Count), number of packets of certain flow at the past 2 seconds.
   - Srv_count, number of packets of the same flow to the same service.

To calculate the rate packet feature, the field "date" in the dataset used to calculate the number of packets within a specific time interval. This research used a 1 second interval to accommodate the same granularity of the three datasets. CAIDA 2007 and 2014 packet trace data reached 10^{-6} second granularity, while WorldCup 1998 server log only at second.

To conform detector position at target-end destination in network, normal dataset Caida Anonymized 2014; which was backbone level dataset; filtered at certain IP_destination 189.116.21.85. While for DDoS Caida 2007, we choosed filtered "ddostrace to the victim" dataset. Likewise in WorldCup 1998 dataset which has already in the form of server targeted dataset. It was important to analyze the traffic models corresponding to the position of the detector, due to possibility of different behavior at different network position.

B. Statistical Analysis

Histogram plots and estimated statistical value were analyzed in statistical analysis. We did empirical cumulative distribution fuction (ecdf) fitting, at first, second and third order of statistical analysis of aggregate traffic. In per-flow analysis, we added information theoretical analysis by measure distance between flows. The distribution of the data obtained empirically from the data and evaluated by the engineering chi-square test.

IV. RESULT AND ANALYSIS

A. Aggregate Traffic

a. Traffic Rate

From traffic rate feature, statistical analysis was done simply by calculating the empirical cumulative distribution function (ecdf). Analysis was performed on the data traffic directly, or the relationships among the data. In this analysis, the relationship between the data carried on with a one lag distance difference. The result shown in table 1.

Table 1 Contoh fitting distribution

Dataset	Time frame	Best fitting distribution	Chi-square test		
			DoF	Statistic	P Value
Normal CAIDA Anonymized 2014 IP_Dest 189.116.21.85, time 130000-131000)	1st 60 s	log normal, s=0,07745 m=7,9589	5	2,8476	0,7235
	2nd 60 s	log normal, s=0,08603 m=7,9799	5	3,5327	0,6184
	3rd 60 s	log normal, s=0,13157 m=7,8946	5	3,5004	0,6233
	4rt 60 s	log normal, s=0,19366 m=7,471	4	6,2201	0,161
	5th 60 s	log normal, s=0,23655 m=6,8805	5	1,182	0,2854
	5 minute	Gen.. Pareto, k=-1,8913 s=5077,1 m=503,97			
Normal WorldCup 1998 day 66	2nd 60 s	log normal, s=0,0801 m=5,8224	5	2,7143	0,7439
	3rd 60 s	log normal, s=0,06571 m=5,6815	5	6,6148	0,2509
	4rt 60 s	log normal, s=0,05273 m=5,7271	5	2,9801	0,7031
	5th 60 s	log normal, s=0,08426 m=5,6018	5	5,2946	0,381
	1st hour	log normal, s=0,09904 m=5,9075	11	18,872	0,0634
	2nd hour	normal s=26,993 m=323,74	11	12,778	0,3081

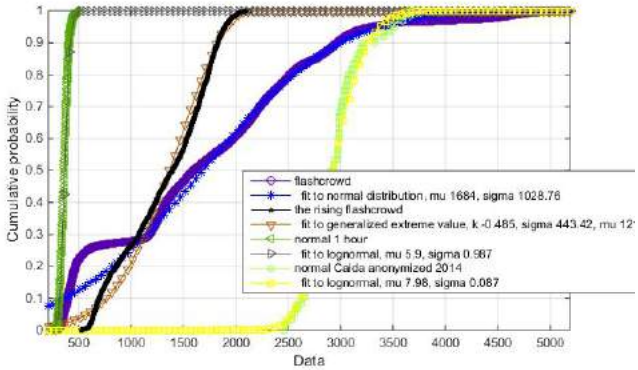

Figure 3 Ecdf rate normal and flashcrowd traffic

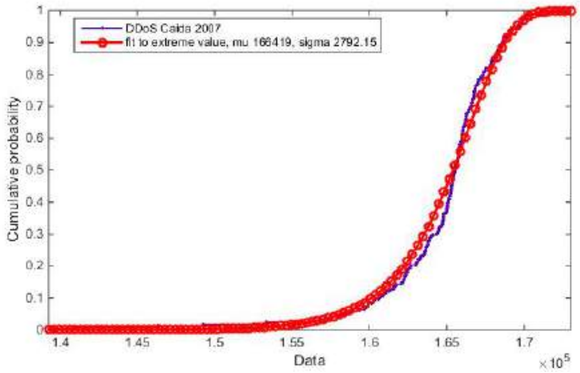

Figure 4 Ecdf rate DDoS CAIDA 2007at steady rate

Normal traffic distribution in different interval had the same traffic distribution despite produced different first-order distribution parameter values. It showed that the high rate had no significant effect on the data distribution model. Whereas when compared with flashcrowd traffic on the same observation interval, traffic could be approached with the same distribution but with a dominant outlier on low rate of traffic (Figure 3). This shows the nature of flashcrowd where high traffic rate was generated by similar traffic distribution to normal traffic, but generated by very large sources. DDoS traffic showed traffic distribution that was different than normal and overall flashcrowd, but had the same distribution as the rising flashcrowd event as both suitably approach by extreme value distribution (figure 4). It showed a high rate of traffic on DDoS occured because the traffic generated by the different distribution by many traffic sources (botnet).

Second order statistical analyzed by autocorrelation analysis from rate data traffic and rate data traffic with one lag difference. From autocorrelation analysis, normal traffic and flashcrowd have autocorrelation value which tends to long range dependence (LRD) compared with DDoS short range dependence (SRD) as shown in Figure 5. Analysis of long-range dependence will be more visible in stationary data with the use of one lag difference as shown in Figure 6.

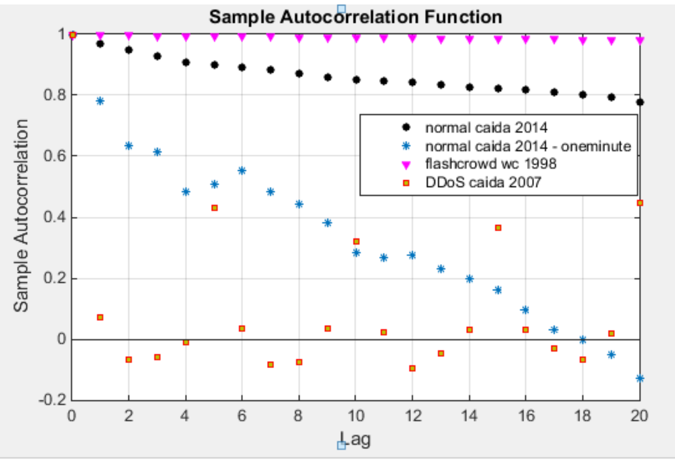

Figure 5 Autocorrelation of rate feature

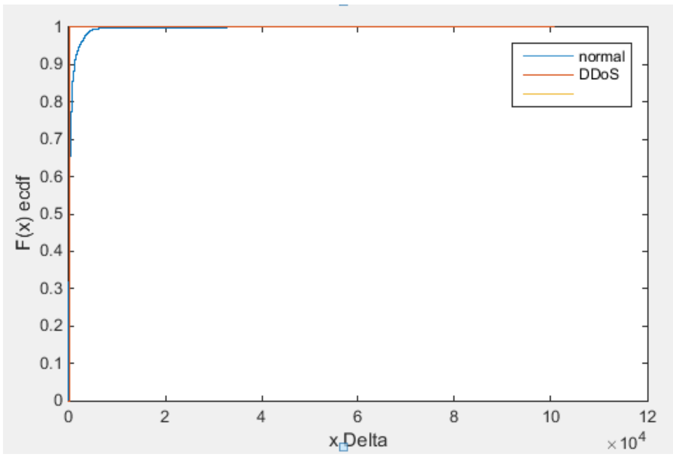

Figure 7 Ecdf delta DDoS and normal

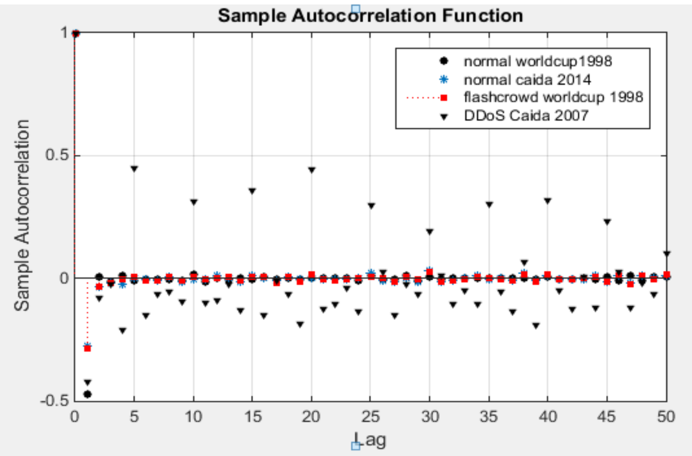

Figure 6 Autocorrelation of rate feature with lag = 1.

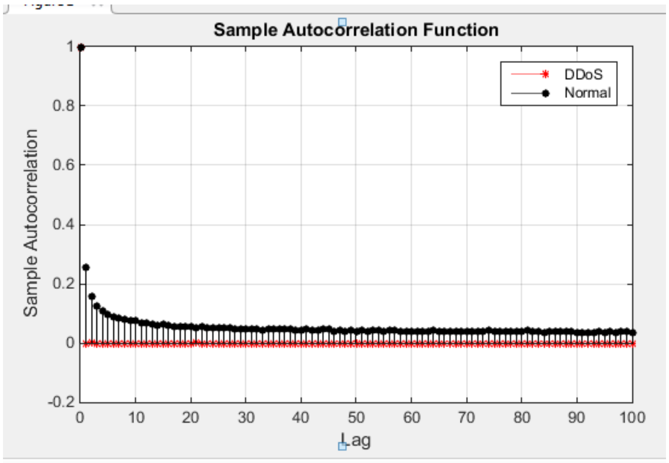

Figure 8 Autocorrelation of delta DDoS and normal

b. Delta

Analysis of the actual delta feature gave the same results with the rate feature as they both relied on the same data that is time. But the delta has already a continuous data packet inter-arrival which was a difference between the arrival time of data packets. Based on delta feature, the more obvious differences in the nature of normal traffic and DDoS shown, as LRD of normal traffic and SRD of DDoS traffic as shown in Figure 8. In this case, the distribution of traffic generated from normal traffic and DDoS very obvious, though difficult to approach with certain distribution, as a result of the extreme value delta DDoS dominant delta value was very small as shown in figure 7. It could be used to identify the presence of anomalous traffic. More results from LRD analysis for DDoS could be seen in a previous report [17].

B. Flow Level Traffic

Extracted features in the KDD Cup 1999 provide 41 features that can be selected to identify the type of traffic anomalies. By making the identification of each feature or relationship between features could provide an opportunity to identify the type of anomalies. Feature number 23 is the packet rate per two seconds of each flow. As shown in feature number 23, every type of DDoS traffic provided different distributions (Figure 9). Likewise, the results of autocorrelation analysis of the data showed normal traffic data indicate the nature of long-range dependence. Anomalies due to some type of DDoS showed different, except smurf attack showed the correlation was almost flat (Figure 10).

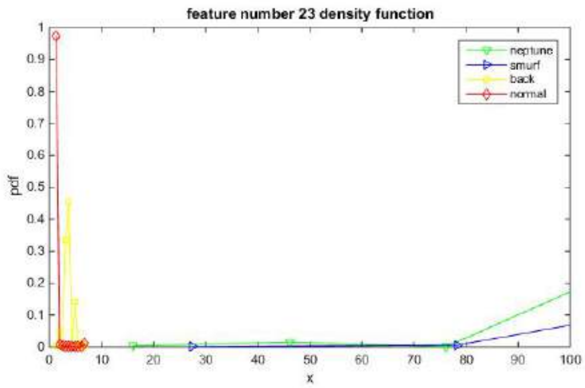

Figure 9 Density function of feature number 23 KDDCup 1999

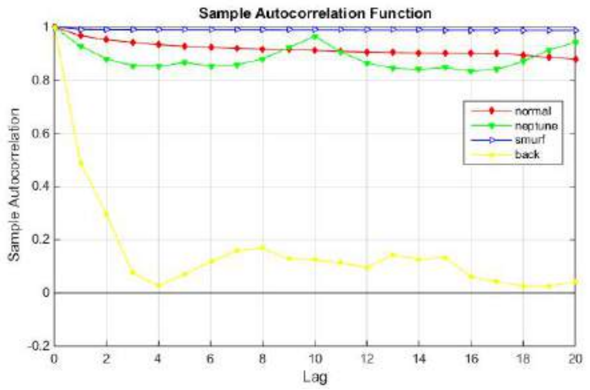

Figure 10 Autocorrelation of feature number 23

Further statistical analysis of traffic using covariance at feature number 23 to 31 from flow based features of KDDCup traffic. The result obtained 31 covariance values. Group analysis done using windowing technique with window size 100 on a set of traffic flows in each type of traffic. In this study, the used of sliding window technique provide slightly different results with the use of batch data. The mean covariance value of every feature on every window were not significantly changed.

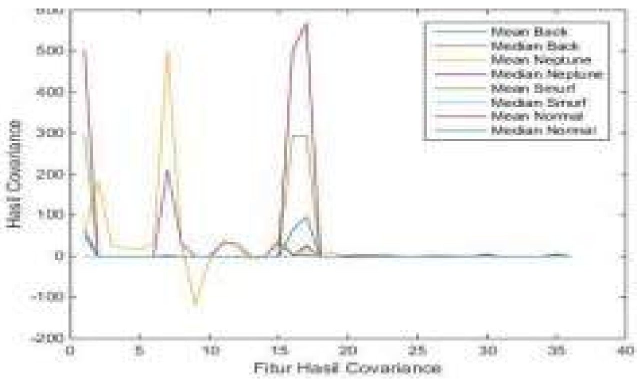

Figure 11 Covariance matrix of feature number 23 to 31 in window size 100

Seen from mean of covariance value of each type of traffic was significantly change in figure 11. The threshold rule could be implemented to differentiate each type of traffic. Whether using classification algorithm such as linear classifier, support vector machine, kernel estimation, decision tree, etc.

One useful method to measure similarity which implement covariance matrix is Mahalanobis distance. In this research, mahalanobis distance measured to identify distance distribution among types of traffic which will be very useful for classifier. Mahalanobis distance was measured for every flow from each type of traffic. For each type of traffic, exponential distribution founded on distance behavior of flow to each type of traffic average as depicted in figure 12. This lead to information on threshold value suitable for every type of traffic.

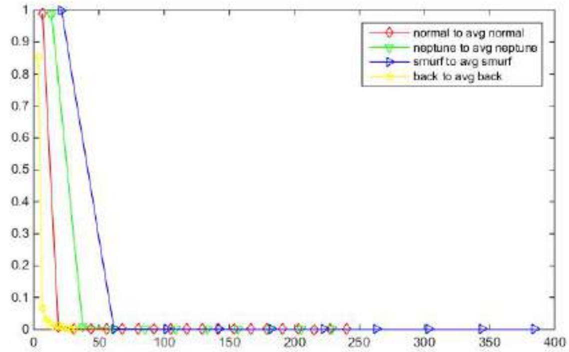

Figure 12 Mahalanobis distance distribution flow to average of each type of traffic

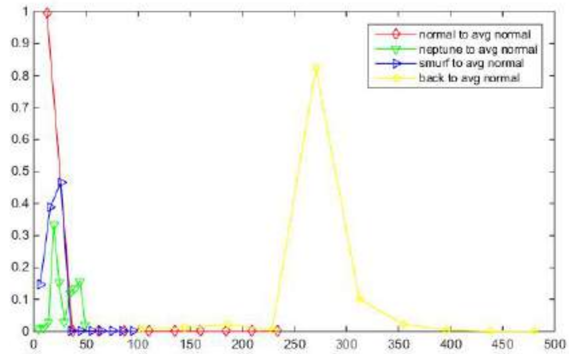

Figure 13 Mahalanobis distance distribution flow to average of normal traffic all features

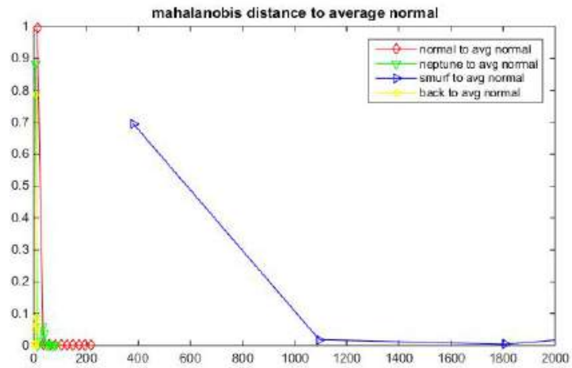

Figure 14 Mahalanobis distance distribution flow to average of normal traffic feature 23 to 31

The used of number of features in detection algorithm was very significant to improve detection accuracy. It shown from figure 13 to figure 14, which provide information of mahalanobis distance distribution vary when implement different number of features. The higher number of feature not significantly complement to higher distance separation for every type of traffic.

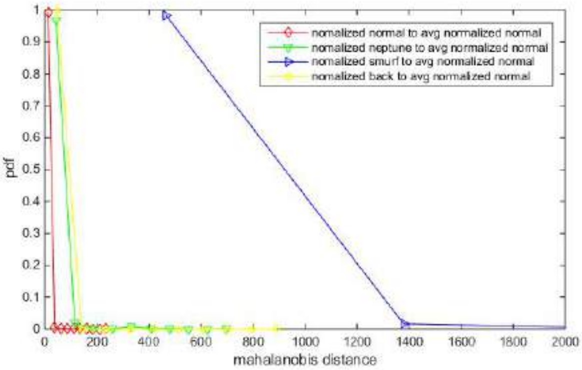

Figure 15 Mahalanobis distance normalized flow to average normalized normal feature 23 to 31

Based on figure 14, distance separation on every type of traffic had not significantly clear, especially to differentiate normal traffic and back attack traffic. Several research did normalization to improve detection accuracy which affected by numbers of data. Normalization done by $(x - \bar{X})/_{\sigma X}$, given data of $X = x_i$, $i = 1 : t$, and $\bar{X} = mean(X)$. The result showed significant improvement in distance separation for each flow to average normalized normal traffic as depicted in figure 15.

V. Conclusion and Future Work

The distribution of data traffic from every feature will vary. Traffic distribution was not only pareto, poisson or gaussian normal but vary at flashcrowd and DDoS. Normal traffic rate feature in almost all datasets could be approximated by a lognormal distribution, while the DDoS and the rising traffic in flashcrowd tend to have extreme value distribution. Based on traffic delta features, normal traffic could be approximated by a generalized Pareto distribution while others tend not to be approached with a certain distribution because it had a dominant high extreme value. By sample autocorrelation of the package rate, appeared that normal traffic and flashcrowd still had the LRD behavior attribute. Each feature of KDDCup 1999 showed different distribution for each type of traffic. Gaussian distribution may only suit for normal flow rate feature, but not in other types of traffic. Mahalanobis distance of each type of traffic also clearly not showed in normal Gaussian, but almost as exponential distribution. It lead to different approached to identify threshold value in mahalanobis distance based anomaly detection.

Data distribution on any traffic features provide justification on the basis of the proposed analysis of traffic anomaly detection method. This information provide justification for future research, especially on threshold identification on each type of traffic. More complete features, longer empirical trace interval and more data will improve the statistical analysis. It seem that mahalanobis distance between types of traffic clearly separated and promising to differentiate types of attacks in our future work.

Referensi

[1] T. Karagiannis, M. Molle and a. M. Faloutsos, "Long-Range Dependence Ten Years of Internet Traffic Modeling," *IEEE Internet Computing,* pp. 2-9, October 2004.

[2] Zafer Sahinoglu and Sirin Tekinay, "On Multimedia Networks: Self-Similar Traffic and Network Performance," *IEEE Communications Magazine,* January 1999.

[3] Ming Li, "An approach to reliably identifying signs of DDOS flood attacks based on LRD traffic pattern recognition," *ELsevier Journal of Computers & Security,* vol. 23, pp. 549-558, 2004.

[4] D. Brignoli, "DDoS Detection Based on Self-Similarity," 2008.

[5] Xiolong Ye, Julong Lan and Wanwei Huang, "Network Traffic Anomaly Detection Based on Self-similarity Using FRFT," Beijing, China, 23-25 May 2013.

[6] Zachary Miller, William Deitrick and Wei Hu, "Anomalous Network Packet Detection using Data Stream Mining," *Journal of Information Security,* vol. 2, pp. 158-168, 2011.

[7] M. Thangavel and Dr. P. Thangaraj, "Cluster Based Statistical Anomaly Intrusion Detection for Varied Attack Intensities," *International Journal of Computer Application ,* vol. 24, pp. 27-33, 2011.

[8] Shui Yu, Wanlei Zhou, Weijia Jia, Song Guo, Yong Xiang and Feilong Tang, "Discriminating DDOS Attack from Flash Crowds Using Flow Correlation Coefficient," *IEEE Transactions on Parallel and Distributed Systems, Vol. 23, No. 6, June 2012,* pp. 1073-1080, 2012.

[9] Therasak Thapngam, Shui Yu, Wanlei Zhou and Gleb Beliakov, "Discrimiinating DDoS attack Traffic from Flash Crowd through Packet Arrival Patterns," 2011.

[10] Shuyuan Jin, Daniel So Yeung and Xizhao Wang, "Network Intrusion Detection in Covariance Feture Space," *Journal of the Pattern Recognition Society 40, Elsevier,* 2007.

[11] Chih-Fong Tsai and Chia-Ying Lin, "A Triangle Area Based Nearest Neighbors Approach to Intrusion Detection," *Pattern Recognition,* vol. 43, pp. 222-229, 2010.

[12] Yudha Purwanto, Kuspriyanto, Hendrawan and Budi Rahardjo, "Traffic Anomaly Detection in DDoS Flooding Attack," in *TSSA*, Bali, 2014.

[13] Zhiyua Tan, Aruna Jamdagni, Xiangjian He, Priyadarsi Nanda and Ren Ping Liu, "A System for Denial of Service Attack Detection Based on Multivariate Correlation Analysis," *IEEE Transaction on Parallel and Distributed Systems,* 2013.

[14] The CAIDA UCSD "DDoS Attack 2007" Dataset, "http://www.caida.org/data/passive/ddos-20070804_dataset.xml".

[15] WorldCup98, "http://ita.ee.lbl.gov/html/contrib/WorldCup.html," 2011.

[16] KDDCup 1999, "http://kdd.ics.uci.edu/databases/kddcup99/kddcup99.html,," Last modified: October 28, 1999.

[17] Hanif Nurohman, Yudha Purwanto and Hafidudin, "Self Similar Traffic," in *ICCEREC*, Bandung, 2015.

Time Based Anomaly Detection Using Residual Polynomial Fitting on Aggregate Traffic Statistic

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***Abstract*—Flashcrowd and Distributed Denial of Service (DDoS) almost has similar symptom across network and server. But security element such Intrusion Detection System (IDS) must handle both differently. If IDS cannot differentiate flashcrowd and DDoS attack, Quality of Service of legal user traffic in flashcrowd will degraded. So it is important for IDS to differentiate between flashcrowd and DDoS. Many earlier comparison method could sense the anomalous event, but not pay much attention to identify which flow was the anomaly. We presented residual calculation between windowed aggregate traffic statistical value combination. With residual calculation among statistical percentile 10th and mean, a high accuracy of flashcrowd and DDoS differentiation of synthetic anomalous event gained. This method could directly identify the anomalous flow and perform visual analysis to detect the start to end of both event.**

Keywords—flashcrowd, DDoS, anomaly detection, residual

I. Introduction

One type of anomaly is flashcrowd, a high rate traffic from increasing number of legal users. As the high rate traffic occurred, anomaly traffic detection in Intrusion Detection System (IDS) with conventional rate threshold often fail to identified flashcrowd as permitted traffic. This will lead to false prevention action which will lead to Quality of Service (QoS) degradation. The intrusion detection must detect the anomaly and identified flashcrowd as permitted traffic as it came from legal users.

DDoS has almost the same symptom as flashcrowd as sensed by a high rate traffic. But high rate traffic in DDoS was caused by illegitimate bots in botnet which send high packet rate traffic. IDS must let the legal traffic flow in flashcrowd to maintain high QoS in flashcrowd event, but IDS has to stop the attacking traffic in DDoS to maintain high QoS of legal users. Although has the same symptom, the anomaly detection method implemented in IDS must have capability to differentiate flashcrowd and DDoS attack. This will make IDS do right prevention action according to certain anomalous event.

Distance and comparison method do comparison among elements, which need minimal two elements to be compared. In case of comparison, most study have done comparison among flows whether between two different flows or with same flow at different time lag. So the system need another condition to decide a condition. And from the comparison, some research could not decide which flow is the anomaly even could detect the anomalous event.

We present residual statistic features combination to differentiate the anomaly traffic caused by flashcrowd and DDoS by its own aggregate traffic statistic accurately. The differentiation use aggregate traffic statistic as it provide faster traffic feature identification than flow level. Our method didn't need any separation, combination or comparison of flows to identified anomaly and differentiate flashcrowd and DDoS. Based from empirical result, very different visual signature of flashcrowd and DDoS attack was detected. The flashcrowd signature visually seen from the decrease time serie of residual value. But the DDoS showed different signature as increasing residual time serie. As we applied threshold obtained from training phase, the anomalous DDoS event detected from the start of the event.

II. Related Work

Flashcrowd and DDoS attack differentiation is a popular research lately. Based on our earlier report in [1], anomaly detection in flashcrowd and DDoS differentiation is a specific capability focus in anomaly detection research. Earlier research in this capability focus implement many method such as statistic, information theory, machine learning, etc.

In similarity and distance concept, flashcrowd and DDoS attack differentiation need two elements to be compared. In [2], pair of upstream routers flow used to calculate probability metric using Bhattacharyya coefficient to distinguish flashcrowd and DDoS attack. Another similarity method to differentiate flashcrowd and DDoS is Pearson correlation for packet arrival pattern used in [3]. This research perfomed three scenario to differentiate normal to DDoS, which were similarity among packet rate in a flow and with different flow. In [4], comparison between two flows from different origin with the same destination was used to calculate similarity metric using proposed flow correlation coefficient to differentiate DDoS and flashcrowd. And flow correlation and information distance used in [5] to differentiated DDoS and flashcorwd in Local Area Network.

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Aggregate traffic analysis usually used for anomaly detection researches. There are several proposed features and methods to detect anomaly traffic. Statistical analysis widely used in anomaly detection research. There are statistical measurement such as traffic volume in each router interface for deviation analysis [6], aggregated traffic statistic for bivariate parametric detection and sequential probability ratio test [7], aggregated traffic for statistical inference and α-stable model [8], and stream clustering aggregated traffic features for non-parametric analysis with t-test statistic [9].

Most of the existing research which implement statistical analysis of aggregate traffic appear in intrusion detection context, but few in flashcrowd and DDoS differentiation. This research used statistical measurement of windowed aggregate traffic to differentiate between flashcrowd and DDoS attack. With the used of statistical features, the system can directly detect the anomalous event in traffic with no need of another element to be compared.

III. FRAMEWORK

In this study, three datasets used for three types of traffic. CAIDA2014 normal dataset used for the representation of normal traffic. Worldcup 98 dataset used for normal traffic and flashcrowd representation. While Caida 2007 dataset used for the representing DDoS traffic. All datasets were divided into two parts for the training and testing phase to perform supervised learning. Training phase was used as based data to identify the signature of normal, flashcrowd and DDoS event. The next phase used to validate and identify similar pattern to the results of each type of traffic signature in the training phase.

Input data gathered from packet counter in fixed size time bin t, as instantaneous packet rate in aggregate traffic (count / t). The t value must be greater than RTT; where RTT package usually for 0,5s in UMTS [10] and 182 ms average in web server [11]; which in this research used 1 s. The use of aggregate traffic features resulting quicker process than using traffic flow features. As a result, the system can be installed in a high intensity environment of network traffic, such as the backbone and edge network.

A. Training Phase

Each three types of input data filtered with three sigma rule to gain more grain data. The three sigma rule from normal traffic type also used as threshold for anomaly identification. Three sigma rule in statistics was to look for values that are around the mean (μ) of the data with a wide range of three standard deviations (σ).

$$Tr = \mu + 3\sigma \tag{1}$$

Landmark windowing utilized to get online temporal statistical value of traffic. Window size was associated with the identification of stationary traffic needs to obtain accurate statistical empirical value. Large window size has the possibility of a more robust statistical value, but has a bad impact for the online process in IDS. It's because a large window size will make slower identification and respond to the attack. In this study, we used the window of 10 t, to measure the value of empirical statistics. The empirical statistics calculated in mean, variance, median, percentile 10th, and percentile 90th are used as traffic features for the anomaly detection. Earlier similar research in [12], use windowed traffic statistic to detect bottleneck in cellular UMTS traffic.

In training step, normal traffic pattern gained by polynomial fit of each traffic features. Each polynomial parameter mined from polynomial equation of each feature from normal traffic training. Fitting was done using two-degree polynomial. Main feature x was obtain from the feature which provide highest information gain.

$$px(x) = p_2x^2 + p_1x + p_0 \tag{2}$$

Algorithm Training
Input Xt, Yt
Output f_p, Tr, $p(f_p)$

Windowing
Statistical features
Information gain
$Max\ (MI)$
return f_p
Mean and standard deviation
$Tr(f_p) = \mu + 3\sigma$
return Tr
Polynomial fit on f_p
return $p(f_p)$

B. Testing Phase

Residual analysis was done by combination of traffic features in features signature. Each traffic features in normal traffic performed statistical polynomial fitting to gain traffic signature. Residual is difference between data observation with variable value which become observation object [13]. Residual analysis was done by substracting x feature value with polynomial fitting output of x value to y polynomial parameter.

$$residual = x - py(x) \tag{3}$$

Algorithm Testing
Input Xt, Yt, f_p, Tr, $p(f_p)$
Output alarm

Windowing
Statistical features
Polynomial fitting of $x(f_p')$ on $p(f_p\ \ )$
return $y(f_p{}')$
$Res = x(f^{'}_{p}) - y(f_{p}{}')$
if $Res > Tr$ **then**
    return alarm on
else
    return alarm off
end if

From residual analysis, we applied three sigma rule as threshold to detect anomaly. The determination of limit threshold using the three sigma rule from mean and standard deviation in training phase. The threshold value from training phase then compared with the data residual testing.

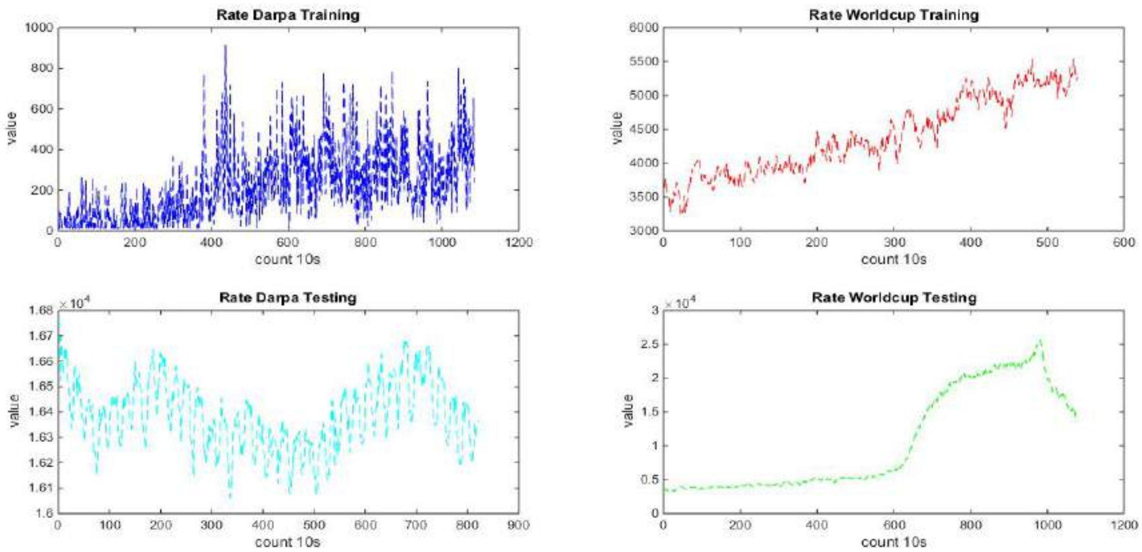

Figure 1. Traffic rate

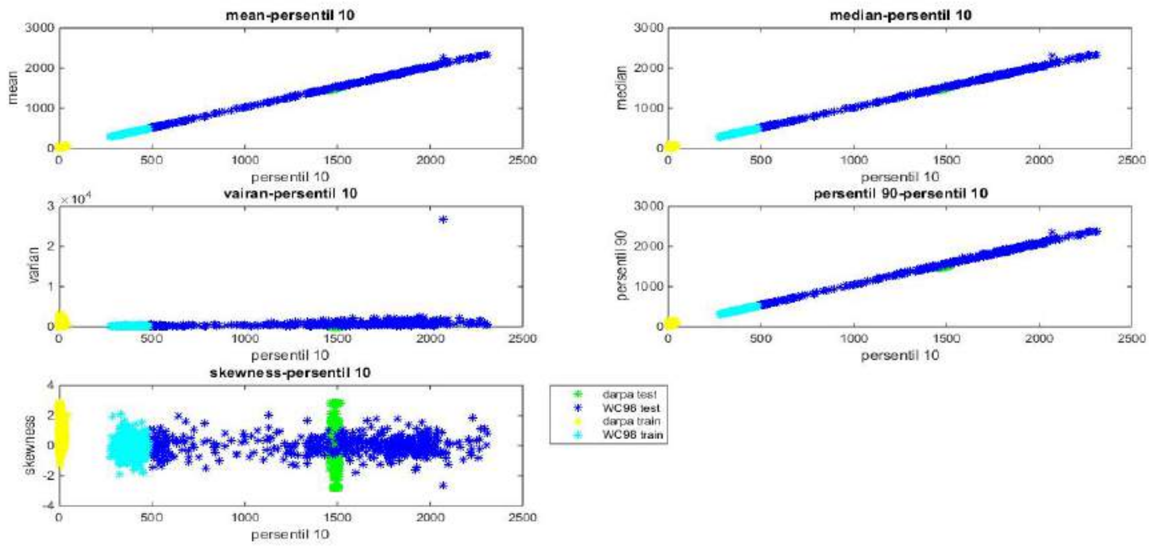

Figure 2. Percentile 10^{th} to X scatter plot

IV. DISCUSSION

We plot the moving average of aggregate traffic rate from each dataset in figure 1, and found a different average baseline in normal dataset. As we used known dataset with different normal baseline, we decided to separate the analysis to two different scenario. First scenario for flashcrowd event detection from worldcup98 dataset, and later for DDoS detection from synthetic normal and DDoS DARPA dataset.

Five traffic features obtained from windowing process. We implement mutual information to obtain most distinguishable feature to differentiate types of traffic. Percentile 10^{th} was the highest information gain among all features, so the combination done based from percentile 10^{th} feature (percentile 10^{th} to X). Visual scatter of percentile 10^{th} to X was depicted in figure 2.

Table 1. Threshold

Data	Threshold
Normal Training Worldcup '98	-15.594444945876807
Normal Training DARPA	-4.539241933300755

From normal training dataset, we determined threshold values as in table 1, and polynomial equation of each normal dataset as in table 2. We found percentile 10th to mean feature as the most distinguishable features combination to detect anomalies. After have done polynomial fitting of mean to percentile 10th, we found residual time serie as depicted in figure 3.

Table 2. Polynomial equation of percentile 10th

Data	Polynomial equation
Training Worldcup'98	3,66459026861813 x^2 + 0,985318861665699 x + 17,8350333273617
Training DARPA	2,61920042913203 x^2 + 0,992377109465127 x + 5,23295271425579

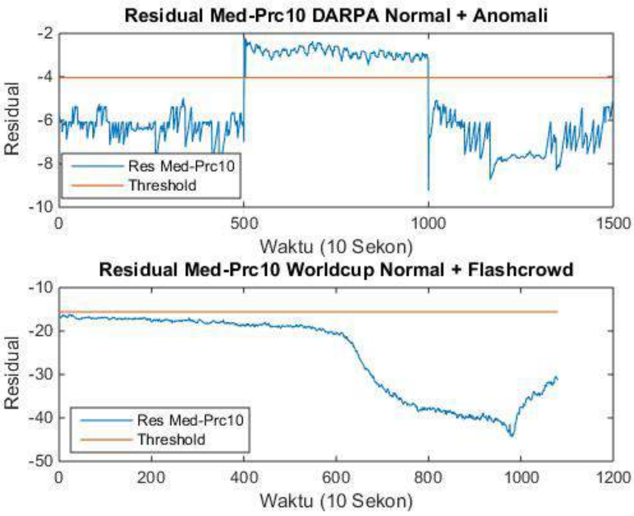

Figure 3. Anomaly detection

High rate flashcrowd traffic came from large normal activity users. Every user generate normal traffic rate, which has low average and high variability user packet per second. High variance rate traffic in flashcrowd mostly influenced by high variance of number of users in a time. This result in high variance of percentile 10th because few different in number of users had big impact for percentile 10th and percentile 90th value. So when combined with mean feature that grew to rise generate decreasing residual values.

Furthermore DDoS traffic came from many bot; far fewer than the normal number of users at flashcrowd [14] ; with a high rate traffic generator. High rate traffic generator from every bot will deliver relatively constant package rate with a low variance. Resulting in percentile 10th and percentile 90th value which not had such a high variation as flashcrowd. This made the residual value increase from combination of relatively fixed mean feature and percentile 10th feature.

Table 4. Detection result

Data	DR	FPR
Data Testing Worldcup 1998 Normal + Flashcrowd	-	0%
Data Testing DARPA Normal + Anomali	99.8%	0%

Research result from testing phase, showed great result. The residual calculation can detect the syntetic DDoS attact as attact and still found flashcrowd as normal traffic. In flashcrowd testing, the residual calculation showed decreasing result, which will not pass the upper threshold. As residual value not pass the upper threshold, the anomaly will not detected so the detection rate remain 0%. But in synthetic attack testing, DDoS attack was detected as the residual value pass the upper threshold. Detection rate reach 99,8% as residual value can detect 99,8% of the windowed DDos traffic, without miss classification (FPR 0%).

High rate flashcrowd traffic come from large normal activity users. Every user generate normal traffic rate, which has low average and high variability user packet per second. High variance rate traffic in flashcrowd mostly influenced by high variance of number of users in a time. This will result in high variance of percentile 10th because few different in number of users will have big impact for percentile 10th and percentile 90th value. So when combined with mean feature that continues to rise will generate decreasing residual values.

Furthermore DDoS traffic came from many bot; far fewer than the normal number of users at flashcrowd; with a high rate traffic generator. High rate traffic generator from every bot will deliver relatively constant package rate with a low variance. Resulting in percentile 10th and percentile 90th value which not have such a high variation as flashcrowd. This makes the residual value increase from combination of relatively fixed mean feature and percentile 10th feature.

V. Conclusion

Residual analysis can be used to detect anomalous traffic event. From empirical scenario, combination of percentile 10th and mean feature, provide high detection accuracy of flashcrowd and DDoS attack differentiation. From residual analysis, 99,8% of windowed DDoS traffic can be detected with zero false positive rate. Flashcrowd event still recognize as normal as the residual value not pass the upper threshold.

We propose to extend this study to online detection, and with the used of another recognized dataset. Online detection shown very important as stream traffic will always enter the detection system, and has to be detected instantaneously. High accuracy of stream detection will provide highly beneficial to security system implementation.

References

[1] Yudha Purwanto, Kuspriyanto, Hendrawan and Budi Rahardjo, "Traffic Anomaly Detection in DDoS Flooding Attack," in *TSSA*, Denpasar, Bali, 2014.

[2] Ke Li, Wanlei Zhou, Ping Li, Jing Hai and Jianwen Liu, "Distinguishing DDoS Attacks from Flash Crowds Using Probability Metrics," in *International Conference on Network and ystem Security*, 2009.

[3] Therasak Thapngam, Shui Yu, Wanlei Zhou and Gleb Beliakov,

"Discrimiinating DDoS attack Traffic from Flash Crowd through Packet Arrival Patterns," 2011.

[4] Shui Yu, Wanlei Zhou, Weijia Jia, Song Guo, Yong Xiang and Feilong Tang, "Discriminating DDOS Attack from Flash Crowds Using FLow Correlation Coefficient," *IEEE Transactions on Parallel and Distributed Systems, Vol. 23, No. 6, June 2012,* pp. 1073-1080, 2012.

[5] Yuan Tao and Shui Yu, "DDoS Attack Detection at Local Area Networks Using Information Theoritical Metrics," in *International Conference on Trust, Security and Privacy in Computing and Communications*, 2013.

[6] Yu Chen, Kai Hwang and Wei-Shin Ku, "Collaborative Detection of DDOS Attack Over Multiple Network Domains," *IEEE Transactions on Parallel and Distributed Systems, vol 18, No 12,* 2007.

[7] Gautam Thatte, Urbashi Mitra and John Heidemann, "Parametric Methods for Anomaly Detection in Aggregate Traffic," *IEEE/ACM Transaksi on Networking Vol 19 No 2 April 2011,* 2011.

[8] Federico SImmross Wattenberg, Juan Ignacio Asensio Perez, Pablo Casaseca de la Higuera, Marcos Martin Fernandez, Ioannis A Dimitridis and Carlos Alberola Lopez, "Anomaly Detection in Network Traffic Based on Statistical Inference and α-Stable Modelling," *IEEE Transaction on Dependable and Secure Computing, Vol 8 No 4 2011,* 2011.

[9] Ming Yu, "An Adaptive Method for Source-end Detection of Pulsing DoS Attacks," vol. 7, no. 5, pp 279-288, 2013.

[10] F. Varica, F. Ricciato and R. Pilz, "Large-scale RTT Measurements form an Operational UMTS/GPRS Network," *Proceedings of First International Conference on Wireless Internet (WICON'05),* 2005.

[11] Phillipa Sessini and Anirban Mahanti, "Observations on Round-Trip Times of TCP Connections," in *ZD Net*, 2006.

[12] F. Ricciato and P. Romirer-Maierhofer, "Traffic Analysis for UMTS Network Validation and Troubleshooting," in *Video and Multimedia Transmission Over Cellular Networks*, M. Rupp, Ed., Austria, John Wiley & Sons, 2009, pp. 229-239.

[13] G. R. Iversen and M. Gergen, Statistic: The Conceptual Approach, New York: Springer Science & Business Media, 1997.

[14] Moheeb Abu Rajab, Jay Zarfoss and Fabian Monrose, "My Botnet is Bigger than Yours (Maybe, Better than Yours) : Why Size Estimates Remain Challenging," in *In First Workshop Hot Topics in Understanding Botnets (HotBots '07)*, 2007.

Performance Analysis of White Box Switch on Software Defined Networking Using Open vSwitch

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Abstract – Complexity and flexibility are the main problems that will be faced in the world of the future network. To be able to answer these problems, a method called Software Defined Network (SDN) is being developed. The SDN concept is to separate the network's controller and forwarding plane of the hardware. In this research, SDN component, white box switch was built. To test its performance, RTT and throughput of some configuration were measured. The result is compared to conventional switch, which is Cisco Catalyst 2950. White Box Switch was created of a computer with 2.50 GHz processor and 32 GBytes memory. The result shows that performance of Cisco Catalyst 2950 is superior to white box switch because of its ASIC that allow it to forward data in hardware. Furthermore minimum specification of white box switch can be calculated, that is 1.225 GHz processor and 1 GBytes memory. To get the best performance, we can use single-board computer to be used as white box switch. Single-board computer contains all of its hardware in a single-board and it will give best performance of a computer.

Keyword : Software Defined Networking (SDN), Open vSwitch (OVS), White Box Switch

I. Introduction

There are two main problems encountered in the world of network at this time, complexity and flexibility. Network must be flexible in order to easily do changes to be able to keep pace with technological developments. In addition, the network also should not be too complex to be done in terms of cost savings. To be able to answer these problems, a method called Software Defined Network (SDN) is being developed.

SDN is an approach taken in the design and management of the network architecture. The simple concept is to separate the network's controller and forwarding plane of the hardware. Network settings is done through software that embedded in SDN Controller. It will give orders to the switch / router that is underneath in order to carry out the task according with to the user. With this, flexibility of a network can be optimized and complexity can be reduced.

Architecture / framework SDN can be divided into 3 parts, namely Northbound (high level of application), SDN Controller, and Southbound (Low level Application). White box is a component of SDN switch located on the Southbound and serves as a component to regulate the flow of data according to SDN controller. Unlike conventional switch, white box switch is just an empty switch ("blank" switch) that do not have brains. To be able to work as it should be, the white-box switch requires a virtual switch software that can be directly embedded or deliberately planted. In addition, the white box switches also must take orders from SDN controller to be able to work. That is the most fundamental difference between the white box switches with conventional switches.

White box this switch and SDN are a perfect solution for the problems to be faced in the future, which is about the complexity and flexibility. No wonder that the emergence of white-box this switch is to get the attention of many large companies. Many companies are starting to take advantage to offer white box switches. From some above statement, it is known that white box switch market is very high. But there is a drawback, the price which can not be exactly cheap. This makes the 'low-budget' researchers have trouble if they want to experiment with white box switches.

So in this research, we try to figure out how to build a white box switch and compare it performance with conventional switch.

II. Related Theory

II.1 Software-Defined Networking

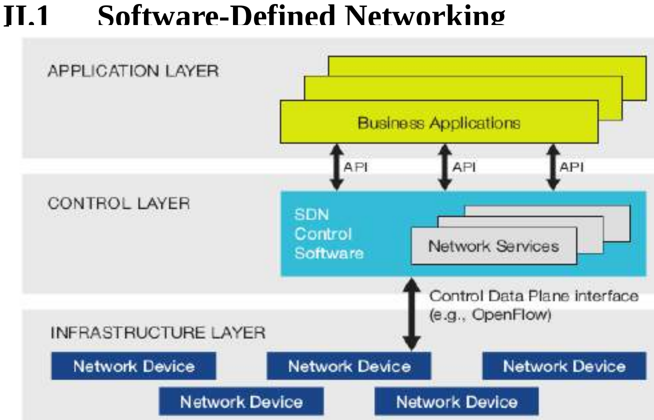


Figure II.1 Software-Defined Networking Architecture [1]

Software Defined Networking (SDN) is an emerging network architecture where network control is decoupled from forwarding and is directly programmable. This migration of control, formerly tightly bound in individual network devices, into accessible computing devices enables the underlying infrastructure to be abstracted for applications and network services, which can treat the network as a logical or virtual entity. [1]

Network intelligence is (logically) centralized in software-based SDN controllers, which maintain a global view of the network. As a result, the network appears to the applications and policy engines as a single, logical switch. With SDN, enterprises and carriers gain vendor-independent control over the entire network from a single logical point, which greatly simplifies the network design and operation. SDN also greatly simplifies the network devices themselves, since they no longer need to understand and process thousands of protocol standards but merely accept instructions from the SDN controllers [1]

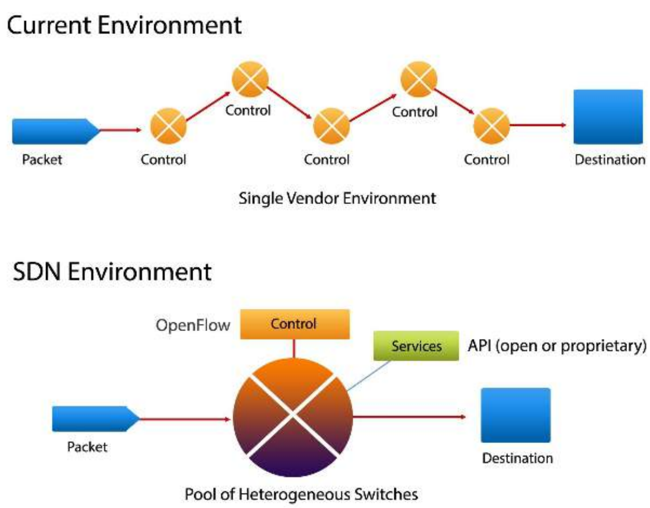


Figure II.2 Comparison between Conventional Network and SDN Network [2]

II.2. White Box Switch

White box switch is a component of SDN located on the Southbound (infrastructure) layer and regulate the flow of data in accordance with the command from SDN controller (control layer). Unlike the conventional switch, white box switch is just an empty switch ("blank" switch) that do not have brains like a conventional switch. All of white box activity is been regulated by SDN controller.

White box switches rely on an operating system (OS). Common operating system for white box switch is Linux-based because of the many open and free Linux tools available that help administrators customize the devices to their needs.[3] To become a switch, white box switch needs a virtual switch software that installed in OS. One of popular virtual switch software is Open vSwitch (OVS).

II.3. Open vSwitch (OVS)

Open vSwitch (OVS) is a production quality, multilayer virtual switch licensed under the open source Apache 2.0 license. It is designed to enable massive network automation through programmatic extension, while still supporting standard management interfaces and protocols. In addition, it is designed to support distribution across multiple physical servers similar to VMware's vNetwork distributed vswitch or Cisco's Nexus 1000V. [3]

OVS is designed to be flexible, portable and to reside within a hypervisor or management domain, to provide connectivity between the virtual machines and the physical interfaces.

It can operate as a basic L2 switch in a standalone configuration, supporting VLAN, SPAN, RSPAN, ACL, QoS policies, port bonding, trunking, GRE and IPsec, tunneling, and per-VM traffic policing. Flow visibility with NetFlow and sFlow is also provided. To support integration into virtual environments, OVS exports interfaces for manipulating the forwarding state and managing configuration state at run-time, allowing the specification on how packets are handled based on their L2, L3, and L4 headers.[4]

III. Application Design and Implementation

III.1 White Box Switch Implementation

White Box Switch is created using a computer with specification of CPU Intel 2.56 GHz, Memory 32 GB, and has 3 ethernet ports. The operating system used is Fedora 20 and in it was installed applications Open vSwitch (OVS) 2.3.1 to create a virtual switch so that the computer can function as a switch.

To make a virtual switch using OVS, we just have to create a virtual bridge and assign computer ethernet port attached to the virtual bridge. All of thing above can be doone using OVS command.

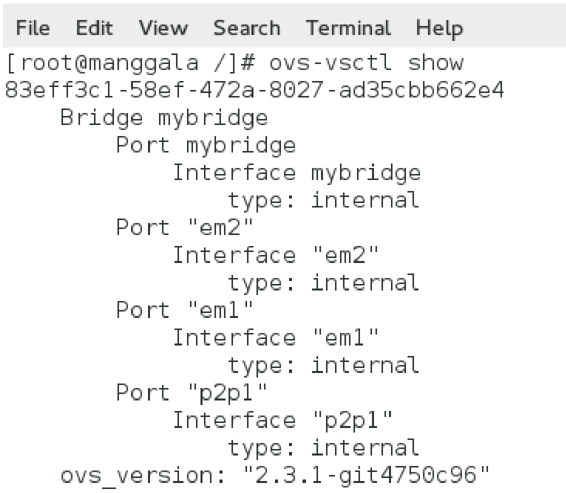
ovs-vsctl add-br mybridge

ovs-vsctl add-port mybridge em1
set interface em1 type=internal

ovs-vsctl add-port mybridge em2
set interface em2 type=internal

ovs-vsctl add-port mybridge p2p1
set interface p2p1 type=internal

Command above will create a virtual bridge named *mybridge* and assign port em1, em2, and p2p1 attached to *mybridge*. To ensure that your virual bridge actually created, you can use command ***ovs-vsctl show*** to check your virtual bridge.



```
File   Edit   View   Search   Terminal   Help
[root@manggala /]# ovs-vsctl show
83eff3c1-58ef-472a-8027-ad35cbb662e4
    Bridge mybridge
        Port mybridge
            Interface mybridge
                type: internal
        Port "em2"
            Interface "em2"
                type: internal
        Port "em1"
            Interface "em1"
                type: internal
        Port "p2p1"
            Interface "p2p1"
                type: internal
    ovs_version: "2.3.1-git4750c96"
```

Figure III.1 Screenshot virtual bridge on OVS

III.2 Test Method

To test the performance of the white box switch, we will measure Round Trip Time (RTT) and throughput (bandwidth) of the white box switch in a network configuration

III.2.1 RTT

Round Trip Time or commonly abbreviated to RTT is the time taken by a pulse signal or a packet of data to travel from point of origin to destination and back again. In the context of the final project iini, place of origin is a client computer while the destination is a computer server. The client computer will initiate a data packet which is then transmitted to the server computer. After receiving the data packet, the server computer will send back the package. So the time required for data packets back to the client computer can be calculated and the time called RTT.
RTT is measured using ping command. It performed by client computer and server computer as destination. We use custom setting on ping command to measure RTT. The command is as shown below.

Ping 192.168.1.10 -c 1000 -i 0.002 -s 1000

-c is command to cchange amount of packet send
-i is the interval between packet send (s)
-s is command to change the packet size

III.2.2 Throughput

In the context of data communication, throughput is the speed of delivery packages that make it through the communication channel. Or it can be said that the throughput is the amount of data that is sent from one place to another within a certain time interval. Throughput is usually measured in bits/second, Kbits/second, Mbit/second. In this research, we measure throughput using benchmark tool IPERF. Using client – server configuration, wa measure the throughput of the switch.

After measure RTT and throughput from the white box switch, then we measure RTT and troughput from conventional switch cisco catalyst 2950 with the same configuration as white box switch. The result will be compared to see how good performance of white box switch.

Throughput is measured using benchmark tool IPERF. Client-Server connection is needed to perform this measurement. Commad for the client side is

\# **iperf -s -u**

and for the client side is

 # iperf -c 192.168.1.20 -u -b 1000m -l 100

-s is command to declare PC to be a server
-c is command to declare PC to be a client
-u is command test using UDP
-b is command to change bandwidth used for measurement
-l is command to change the payload size

III.3 White Box Switch Testing

We consider that will be two testing methods to evaluate white box switch performance. First method is to compare RTT and throughput from white box switch and conventional Cisco switch. Then the second method is compare RTT and throughput from one white box switch with more than 1 virtual bridge in white box switch. In these test, we distribute configuration with two PC as a client and a server which connected via white box switch or conventional cisco switch.

III.3.1 Comparison white box switch and conventional cisco switch catalyst 2950

The network configuration which white box switch implemented is like in figure III.2. There are 2 PC, a server (IP address 192.168.1.10 / 24) and a client (IP address 192.168.1.20 / 24) that connected via white box switch with one virtual bridge.

To see the performance, RTT and throughput will be measure in each configuration. The result of this test will be compared so we can find out how good the performance of white box switch in forwarding data.

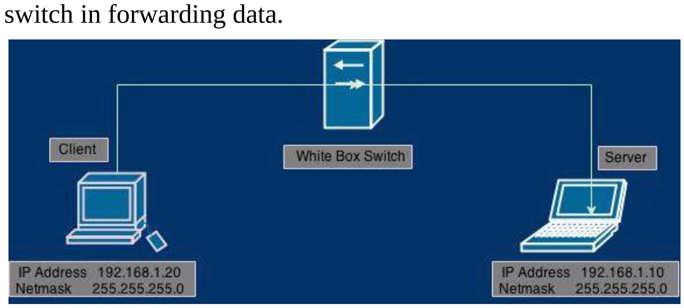


Figure III.2 Configuration using white nox switch

OVS command used to build a white box switch with above configuration is as follows.

ovs-vsctl add-br mybridge

ovs-vsctl add-port mybridge em1
set interface em1 type=internal

ovs-vsctl add-port mybridge em2
set interface em2 type=internal

Command above is to create virtual bridge named mybridge and assign to port em1 and em2 attached to mybridge. Then using Ethernet cable to connect ethernet port between client/server computer to port in mybridge.

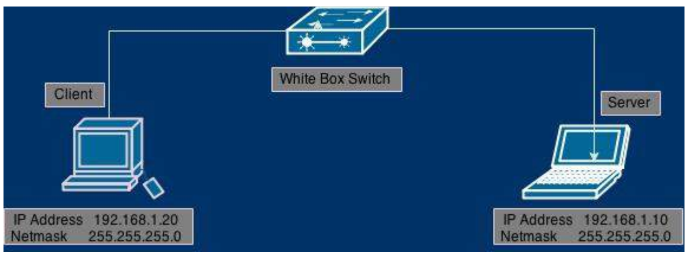


Figure III.3 Configuration using cisco catalyst 2950

Figure III.3 is the configuration to measure performance cisco catalyst 2950. The configuration is the same aswhite box switch configuration, just replace white box switch with cisco catalyst 2950. Because it is actually a switch, we do not need any command for configure the switch.

III.3.2 Specification white box switch performance

OVS can create more than one virtual bridge in a system. But maximal total amount of virtual bridge is five. It means that maximal hop that can be managed by OVS is not more than five. This testbeds consist of four configuration with increased virtual bridge on white box switch.

Goal of this test is to find out whether specification of white box switch may effect performance of white box switch. To see the performance, RTT and throughput will be measure in each configuration. And during the process of measuring throughput, CPU usage of white box switch will be measure too. With this, performance of white box switch can be observed.

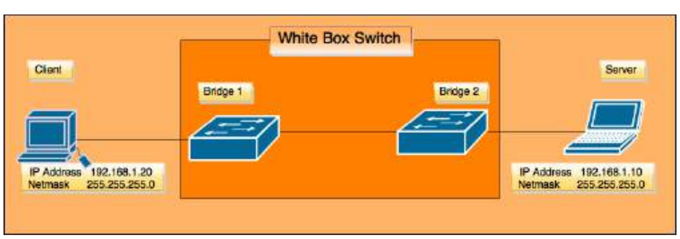


Figure III.4 White box switch with two virtual bridges

To connect two virtual bridges, we can use patch ports. OVS command used to build a white box switch with above configuration is as follows.

ovs-vsctl add-br mybridge

ovs-vsctl add-port mybridge em1
set interface em1 type=internal

ovs-vsctl add-port mybridge em2
set interface em2 type=internal

ovs-vsctl add-port mybridge patch1-2
ovs-vsctl set interface patch1-2 type=patch
ovs-vsctl set interface patch1-2 options:peer=patch2-1

ovs-vsctl add-port mybridge patch2-1
ovs-vsctl set interface patch2-1 type=patch
ovs-vsctl set interface patch2-1 options:peer=patch1-2

Patch port is feature provided by OVS for connecting two bridge. Patch port use a virtual port that we can create in OVS environment. To connect two bridges, we have to repeat the command twice, once for the first bridge (specifying the patch port on the 2nd bridge as the peer), and again for the second bridge (specifying the patch port on the 1st bridge as the peer). So, the two patch interfaces must have reversed name and peer values. [5]

For the other configuration, OVS command would be same. We can just add the patch port corresponding with how much switch we have. The other configuration are as shown below.

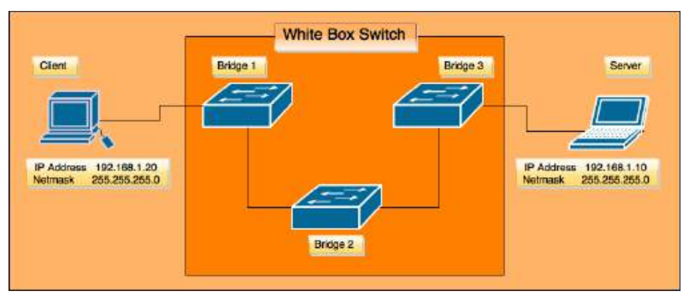


Figure III.5 White box switch with three virtual bridges

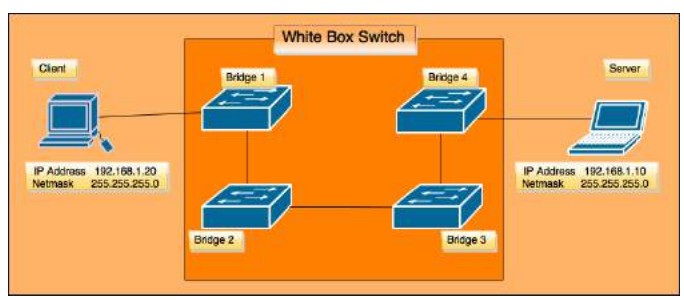


Figure III.6 White box switch with four virtual bridges

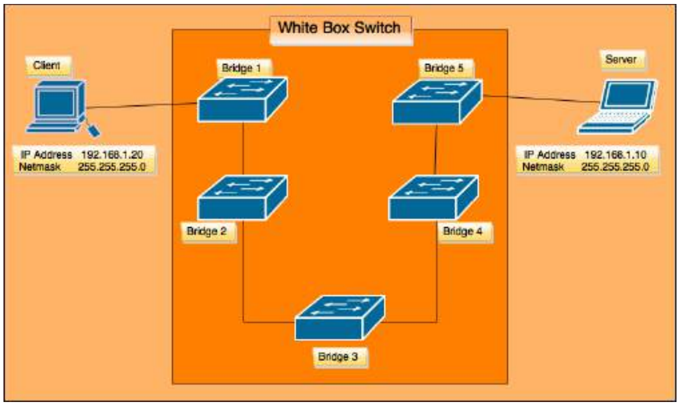


Figure III.7 White box switch with five virtual bridges

IV. Result and Performace Analysis

IV. 1 Comparison betweenWhite Box Switch and Cisco Catalyst 2950

IV.1.1 RTT

The result of the RTT test is as shown in figure IV.1 and table IV.1

Table IV.1 RTT Comparison

Packet Size (Kbytes)	White Box Switch		Cisco Catalyst 2950	
	Average (ms)	Deviation (ms)	Average (ms)	Deviation (ms)
0.1	0.45	0.064	0.287	0.029
1	0.465	0.064	0.332	0.023
5	0.628	0.082	0.399	0.078
10	0.815	0.095	0.421	0.056
15	0.884	0.086	0.512	0.036
20	0.962	0.067	0.542	0.039
25	1.077	0.099	0.685	0.06
30	1.168	0.103	0.789	0.071
35	1.273	0.05	0.879	0.052
40	1.308	0.066	0.924	0.061
45	1.409	0.076	1.04	0.067
50	1.524	0.075	1.17	0.078
55	1.614	0.088	1.224	0.07
60	1.688	0.09	1.288	0.071

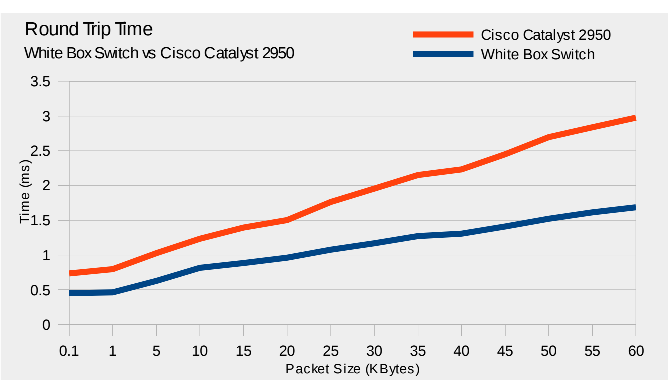


Figure IV.1 RTT Comparison

We varied the packet size from 100 to 60000 bytes to see the performance of both switch. As can be observed, the Cisco Catalyst 2950 showing better performance than white box switch for all packet sizes. The different is less than 0.2 ms in small packet sizes but it's getting bigger in the bigger packet sizes.

IV.1.2 Throughput

The result of the Throughput test is as shown in figure IV.2 and table IV.2

Table IV.2 Throughput Comparison

Payload Size (Bytes)	White Box Switch					Cisco Catalyst 2950				
	Bandwidth (Mbit/second)	Jitter (ms)	Lost	Total	Paket Loss (%)	Bandwidth (Mbit/second)	Jitter (ms)	Lost	Total	Paket Loss (%)
100	55.1	0.011	1085	689232	0.16	55.4	0.005	1851	691893	0.27
500	276	0.015	4250	689831	0.62	277	0.039	3701	692200	0.53
1000	552	0.019	2877	689627	0.42	554	0.017	3833	692263	0.55
1500	785	0.056	2387	654027	0.36	786	0.017	2627	655008	0.4
2000	887	0.03	3088	554515	0.56	895	0.028	2470	559218	0.44
2500	946	0.038	1483	472990	0.31	944	0.043	1852	472121	0.39
3000	891	0.035	1066	371156	0.45	907	0.029	1576	377759	0.42
3100	903	0.036	1724	363994	0.47	912	0.034	1395	367706	0.38
3200	908	0.039	1637	354590	0.4	920	0.033	1599	359195	0.45
6400	954	0.072	833	186232	0.45	952	0.073	789	185881	0.42
12800	960	0.073	294	92771	0.42	960	0.152	371	93713	0.4
25600	960	0.178	168	46880	0.36	960	0.208	189	46893	0.4
51200	961	0.438	72	23452	0.31	962	0.401	66	23478	0.2

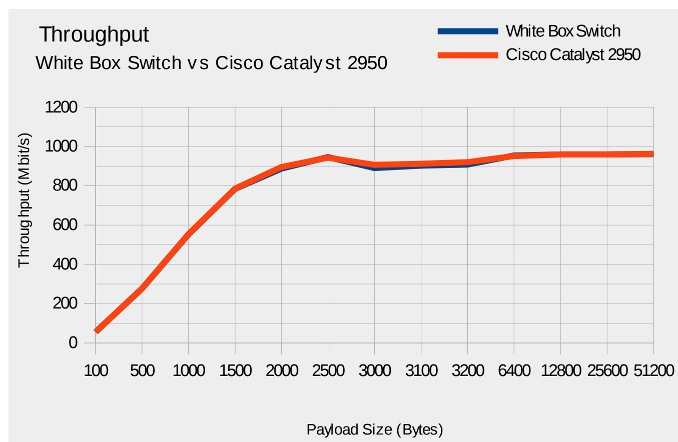


Figure IV.2 Throughput Comparison

We varied the payload size from 100 to 51200 bytes to see the performance of both switch. As can be observed from the table IV.2 and figure IV.2, white box switch and Cisco Catalyst 2950 almost showed the same result on the throughput performance. Both of switch also have a similar behavior for all payload sizes. But, rable IV.2 show that Cisco Catalyst 2950 still has better performance than white box switch although no more than 12 Mbit/second. In the end, the maximum throughput that can be reach by both switch is almost the same, it is around 960 Mbits/second..

IV.2.3 Analysis

From the above result, both RTT and throughput shown that Cisco Catalyst 2950 is more superior than white box switch.

This means that the performance of software-based switch is still can not match to the performance of hardware-based switch. Easily, it can be said that OVS is a software, and to be running properly, OVS need a computer with memory and processor.. Software-switching forward the packets using help by main CPU.
It is another case for Cisco Catalyst 2950 which have ASIC (Application Specific Integrated Circuits) that allow them to forward packets in hardware. ASIC will makes forwarding packets faster than software-switched. It is because with ASIC, forwarding packets will just be handled by PFC Card, and it's not the general purpose processors such as the main CPU. So, software-switched have to reach main CPU to be able to forward packets while hardware-switched have not.

IV.2 Performance of White Box Switch

IV.2.1 RTT

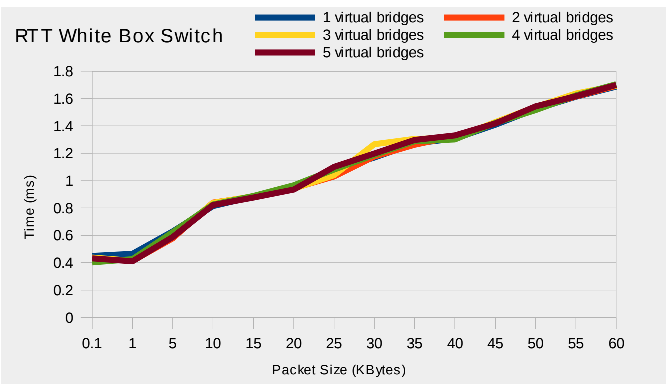


Figure IV.3 RTT of White Box switch's Variationts

As shown in figure IV.3, all of white boxx switch almost have same result for all packet sizes. And beside that, all of them have a similar behavior for all payload sizes. In the end, it can be said that adding a virtual switch on a white box switch will not decrease the white box switch performance.

IV.2.2 Throughput

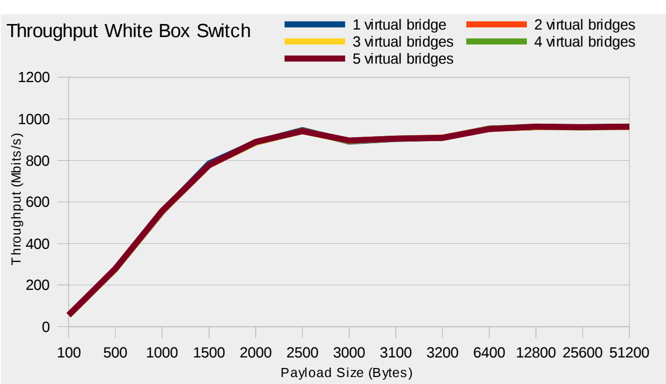


Figure IV.4 Throughput of White Box switch's Variations

As shown in figure IV.4, all of white boxx switch almost have same throughput result for all packet sizes. And beside that, all of them have a similar behavior for all payload sizes. In the end, it can be said that adding a virtual switch on a white box switch will not decrease the white box switch throughput.

IV.2.3 Analysis

The result of this test is really interesting because it fit with the initial hypothesis but the result is unexpected. It is expected that with the increase of virtual bridges, performance of white box switch (RTT and throughput) will decrease but it is not happened. The performance is steady although we increase the number of virtual bridges until five. This is only means that specification of white box switch is still above the minimum requirements of OVS to be able to work optimally. To prove that statement, we measure the CPU and Memory Usage when white box switch work optimally to measure the throughput at the maximum payload size. It is shown in the table IV.3 below.

Table IV.3 CPU & Memory Usage of White Box Switch

White Box Switch	CPU Usage	Memory Used (Mbytes)
1 virtual bridges	< 32 %	779.8
2 virtual bridges	< 35%	785.3
3 virtual bridges	< 40 %	804.7
4 virtual bridges	< 45 %	811.7
5 virtual bridges	< 49 %	819.5

The white box switch have processor with specification Intel ® Xeon ® CPU E5-2609 v2 @ 2.50GHz and Memory 32 GB. As we can see above, the specification needed to run maximum virtual bridge is still below the specification of white box switch. This makes the white box switch can work optimally without decrease the performance of switching. Based on the data in table IV.3 too, we can croughly calculate the minimum specification of white box switch to be able to work optimally.
For the processor, let's make it used 49% of total clock. So we basically we can calculate

$$\mathbf{49\% \times 2.50GHz = 1.225GHz}$$

For the memory let's make it used 1GB (rounding off value).
So the minimum specification of the white box switch to be able to work optimally as a switch processor with minimum clock rate 1.224 GHz and minimum RAM 1 GBytes. We don't running other software application, so that specification is actually running the OS (Fedora 20) and OVS.

V. Conclusion

The conclusions that can be drawn from this research are as follows.

1. Forwarding data performance of white box switch is still inferior to Cisco Catalyst 2950. This is because Cisco switch using ASIC that allow them to forward packets in hardware (hardware-switched) while white box switch based on (software-switched)

2. Based on this research, the recommendation specifications of white box switch are 1.225 GHz clock rate processor and 1 Gbytes memory. Another hardware specification can be customized based on needs.

References

[1] "*Software-Defined Networking*: *The New Norm for Networks*." Open Networking Foundation, Palo Alto, CA, 2012

[2] Laliberte, Bob. "Market Landscape Report: The Emergence of Software-defined Network Architectures." Enterprise Strategy Group, Milford, MA, 2013

[3] Open vSwitch. http://www.openvswitch.org, accessed on November, 25th 2014, 19.00 GMT +7

[4] F. Sans and E. Gamess, "Analytical Performance Evaluation od Different Switch Solution," Journal of Computer Networks and Communications, vol. 2013, Article ID 953797, 11 pages, 2013.

[5] Connecting OVS Bridges with Patch Ports. http://wwwblog.scottlowe.org/2012/11/27/connecting-ovs-bridges-with-patch-ports/ accessed on April, 25th 2015 17.00 GMT +7

Performance Analysis for Uniform and Binomial Distribution on Contention Window using Different Hop Distance

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***Abstract*—In this paper, we present a mathematical analysis on the performance of uniformly distributed and non-uniformly distributed backoff timer based on binomial algorithm for contention window. We evaluates these protocols under mathematical modelling in order to analyze the average throughput, end-to-end delay and collision probability performance by using different value of hop distance during data transmission process by varying number of nodes from 50 to 300 vehicles. The simulation results show that binomial distribution outperforms the conventional uniform distribution in terms of throughput, delay and collision probability in all scenarios. We also can observed that by using longer hop distance, the better performances have been achieved in both distributions, particularly by using binomial distribution. Thus we can say that the binomial probability distribution that is specifically designed to replace the conventional uniform probability distribution in order to differentiate the selection probability during the backoff process of selecting the transmission slot can be implemented in MAC protocol for vehicular network.**

Keywords—Uniform and non-uniformly distibuted backoff timer, binomial algorithm, MAC protocol

I. INTRODUCTION

The MAC protocol for vehicular network is known as Distributed Coordination Function or DCF, which the standard is based on CSMA/CA. The DCF should be implemented in all stations, where it is used in ad-hoc and infrastructure network configurations [1]. As in Figure 1, for a node to do the transmitting process, it shall sense the medium to be idle. If the medium is idle, then transmission process can be continued [2]. The CSMA/CA distributed algorithm will command an interval of minimum specified time that exist between transmissible frame sequences. A transmitting station must make sure that the medium is idle for this required duration before attempt to do the transmitting process. If the medium is busy, then that station shall defer until the end of the current transmission, in which the medium is sensed idle for a DIFS (Distributed Inter Frame Space) period. After defer, or before attempting to resend after a successful transmission, the station will choose a random backoff interval and would decrease the backoff interval counter while the medium is idle.

At the beginning process of backoff procedure in which a collision happen, a station will chooses its backoff stage to 0 and arms the backoff timer by executing a uniformly distributed random time backoff from the initial contention window which representing the number of minimum contention window size. If the medium is idle, the backoff timer will be decremented. Otherwise, the process will be frozen when there is a medium activity. The decrementing process of backoff timer is resume if the medium is sensed idle again for a DIFS time. Here, the station will transmits its frame only if the backoff timer reaches zero. Let say the transmitted frame has not received any ACK frame, a collision is detected and the station will retries to transmit the frame by switching to the next backoff stage where the contention window size will be doubled [3].

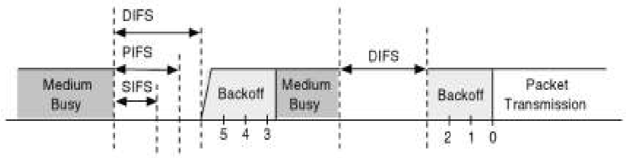

Fig. 1. Basic access method of DCF

After the maximum backoff stage, the transmission attempts will no longer affect the contention window, in which this latter remains constant value with the maximum number of contention window. If a maximum number of retransmissions is reached, the frame will be dropped. During the backoff procedure, when an ACK frame is received, the station will resets the contention window at its initial backoff range, so that the backoff procedure shall be invoked for a node to transmit the next frame. This process is done after it reached the maximum backoff stage too, no matter the transmission is successful or not. Noted that, a Short InterFrame Space (SIFS) time is used for ACK frames. If a DATA frame is correctly received, the receiver station will wait for a SIFS time before transmitting the ACK frame. If the ACK frame is not received within an ACK timeout interval, it is assumed that a collision has been occurred.

Nevertheless, the main characteristics in vehicular network where nodes are typically move on a road with higher mobility and frequent network topology change dynamically than those in other ad-hoc network scenarios lead to several challenges in network design, especially on MAC layer [4]. Aiming to improve the MAC performance, we propose a mathematical analysis by implementing a non-uniformly distributed backoff timer based on binomial distribution for contention window, so that we can increase the performance of throughput while reduce the delay and collision probability.

II. RELATED WORKS

The usual method of avoiding collision is to adjust a contention window size to the current number of competing nodes. From previous studies, there a several methods of tuning a number of contention slots in terms of varying workload of the network that have been proposed and successfully used, where most of them focused on contention window extension after a collision and a contention window size reduce in case of a successful packet transmission. As in [5], they proposed the MAC layer of Tiny Operating System which called as B-MAC. It is a type of fixed-window CSMA protocol where it selects contention slots uniformly at random. B-MAC is designed based on a fixed size of contention window because of the reasonably well performance that has been sustained in an actual environment. On the contrary, there is a limitation in the network scalability where it has low intermittently in a high-load state. While in Multiple Access with Collision Avoidance for Wireless (MACAW), the researchers exploits the Binary Exponential Backoff method by not sharing the channel state information since it does not make use of carrier sense [6]. By using this protocol, it will restarts contention for the upcoming transmission because of the contention window size has been initialized to the minimum number only if the previous node succeeds in transmitting a packet. Nevertheless, an overhead is suffered when the medium is accessed, since the nodes competing to access the medium are centered in a certain interval and the contention window size changes substantially. Thus MACAW resolves this limitation by using a learning method that does not newly reset the contention window size, instead decreases the contention window size that has been used in the previous contention as a size for the next contention, only after a packet is successfully transmitted.

In 802.11 specification, it resolves the fairness problem of service by using a memory technique [7,8]. As in conventional procedure, a node that participates in a contention mode, in which one of the slots in the contention window is randomly picked by implementing a uniform probability distribution, will cause the value of selected slot is set in a countdown backoff timer. The countdown backoff timer is stopped when node sensed that the medium is busy, and it will resumes countdown when the medium is idle. Let say the number of countdown timer becomes 0, the corresponding node starts to transmit. When the transmission process is completed, the contention window size is initialized to the predetermined minimum value. Consequently, the bandwidth will dissipates since the node always requires to determine a sufficient contention window size. In order to solve such a limitation, a Differential Probability of Selection MAC (DPSMAC) protocol has been proposed [9]. It exploits the fixed contention window size and random slot selection method with non-uniform probability by implementing a geometric distribution function. In their analysis, this protocol has minimizes the collisions between nodes, reduces the delay time, and maintains the fairness of service relatively and constantly, although the simplicity of the protocols structure as compared to conventional 802.11 MAC protocols. As for the limitation, this protocol has some difficulty of reconfiguring the protocols after execution, thus lacking in flexibility.

The alternative technique proposed recently consists in introducing a non-uniform distribution using geometric probability of choosing slots, such as in Dynamical Adjustment on Contention Window (DACW) that specially designed for vehicular ad hoc networks by considering vehicle's travel information such as location, speed and direction [10]. From their analysis, we can see that this protocol has decreases the possibility of continuous collisions among the competing nodes, together with increases the average throughput for either symmetrically or asymmetrically bidirectional traffic condition. However, compared to the conventional Enhanced Distributed Channel Access (EDCA) mechanism, it can be concluded that the effectiveness of this protocol only works in bidirectional highway scenario.

In a nutshell, we can conclude that size of contention window is the major factor affecting the MAC performance in vehicular network. Thus in this research, we propose a modified backoff algorithm that can decrease the possibility of continuous collisions among the competing vehicles by generating a non-uniformly distributed backoff timer based on binomial distribution.

III. THE MODIFICATION OF MAC PROTOCOL

Our proposed protocol is based on a non-uniform probability distribution, where the selection of transmission slot differs most notable from that in conventional CSMA based wireless MAC protocols, so that it can reduces the overlapped selection rate of a slot by essentially choosing a slot according to the differentiated probability. As in Figure 2, if a random slot succeeds in transmitting in the contention window used for backoff, slots in the remaining locations can be selected for the next process, except for the slot that has been used before. In other words, the probability distribution function must be designed such that those remaining slots that are located except the slot that has successfully transmitted can be deliberately chose. The probability distribution function of such a property can be derived by multiplying the probability with which remaining slots can be selected by that with which succeeding slots can be selected based on a random slot for all slots.

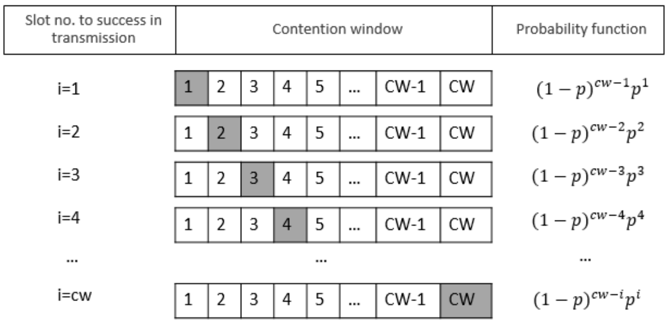

Fig. 2. Probability of selecting a slot in order to reduce the contention between nodes

Looking towards at the designation of probability distribution function, it shows that the best performance can be determined if the first slot is selected when no other contention happens in the current contention window, which is at i= 1. If there is a collision again, it might be better to choose the earlier slot again even though the first slot has already been selected before. So, the slot is selected in order to transmit data without a collision for the minimum delay time. In order to sustain such an optimum selection method, if the probability distribution function is derived using the probability with which the remaining slots can be selected, it could be said that the probability (i) with which each node chooses the i^{th} slot within the range of the contention window follows a binomial distribution with a parameter p, thus the probability mass function can be given as follows:

$$f(x:p,n) = \binom{n}{x} p^x (1-p)^{n-x} \quad (1)$$

where n is number of trial, x is number of success in n trial and p is probability of success in single trial [11].

IV. Mathematical Analysis

By implementing binomial distribution function, we have $P_i = (1-p)^n$ as probability that the slot is in idle state, $P_s = (np)(1-p)^{n-1}$ as the probability that a transmission occurring on the channel is successful, and $P_c = 1 - [(1-p+np)(1-p)^{n-1}]$ as the probability of collision transmission. The transmission probability, p can be calculated with minimum value of contention window as $p = \frac{2}{CW_{min}+1}$. From P_i, P_s and P_c, the expected value of collision number before a successful transmission can be determined as:

$$E(N_c) = \frac{P_c}{P_s} \quad (2)$$

Thus it becomes:

$$E(N_c) = \frac{1-((1-p+(\alpha.d)p)(1-p)^{(\alpha.d)-1})}{(\alpha.d)p(1-p)^{(\alpha.d)-1}} \quad (3)$$

If there is a collision of transmission, the time taken will consists the message transmission time and DIFS time. So, the total collision time before a successful transmission can be obtained as:

$$T_{col} = \sigma.\frac{1-(1-p+(\alpha.d.p))(1-p)^{(\alpha.d)-1})}{(\alpha.d.p)(1-p)^{(\alpha.d)-1}} \quad (4)$$

If the collision is happen between 2 idle periods, the expected number of idle period will be:

$$E(N_i) = \frac{(1-p+(\alpha.d)p)(1-p)^{(\alpha.d)-1})}{(\alpha.d)p(1-p)^{(\alpha.d)-1}} \quad (5)$$

The number of time slots in each idle period is obtained by p and contending vehicle number, which is αd. The expected value of this situation will be:

$$E(T_i) = \sigma\left[1-(1-p^{(\alpha.d)}\right]\sum_{i=0}^{\infty} i.(1-p)^{(\alpha.d)i} \quad (6)$$

Then it becomes:

$$E(T_i) = \frac{\sigma.(1-p)^{(\alpha.d)}}{1-(1-p)^{(\alpha.d)}} \quad (7)$$

Next, the total collision time before a successful transmission occur can be obtained as:

$$T_{idle} = \frac{(1-p+(\alpha.d.p))(1-p)^{(\alpha.d)-1})}{(\alpha.d.p)(1-p)^{(\alpha.d)-1}} . \frac{\sigma.(1-p)^{(\alpha.d)}}{1-(1-p)^{(\alpha.d)}} \quad (8)$$

While the equation for a successful transmission time will be:

$$T_{trans} = \sigma m + \sigma D \quad (9)$$

Since single hop transmission supposed to consist the summation of collision time, idle time and successful transmission time, then the equation will be:

$$T_{hop} = T_{col} + T_{idle} + T_{trans} \quad (10)$$

From source to destination, the end-to-end multi hop flow is resulting due to L/d relay hops [12]. So, end-to-end delay can be determined as:

$$\text{Delay} = \left(\frac{L}{d}\right) \times T_{hop} \quad (11)$$

While the average throughput [13] can be expressed as

$$\text{Throughput} = \frac{P_s \times payload}{[P_i \times t_{slot}] + [P_s \times t_{success}] + [P_c \times t_{col}]} \quad (12)$$

V. Result and Analysis

We validate our analysis using MathCad Prime 3.0 Software. We choose 1 km road segment that composed of 4 lanes. We set the message size, m of 1000 Bytes with 1 Mbps of wireless transmission rate which corresponds to the transmission time of 32 time slots. As shown in Table 1, we presented the road traffic parameters and MAC protocol settings. Noted that the vehicle densities are set as 50, 100, 150, 200, 250 and 300 vel/km/lane, thus the corresponding values of vehicle density α are 0.2, 0.4, 0.6, 0.8, 1.0 and 1.2 vel/m, respectively. We also vary the hop distance d at 50, 75 and 100 m in order to see the differences of using different value of hop distance during data transmission process.

TABLE 1. Simulation parameter

Parameter and setting	**Value**
Vehicle density, α	50, 100, 150, 200, 250, 300 vel/km/lane
Road length, L	1000 m
Number of lanes	4
Time slot, σ	20µs
DIFS time, σD	50µs
Message transmission time, σm	8ms
Hop distance, d	50, 75, 100 m
Message size	1000 bytes
Wireless transmission rate, r	1 Mbps
CWmin	31 slots
CWmax	1023 slots
Payload	8184 bits

We determine the throughput for uniform and binomial distribution for hop distance of 50 m while the number of nodes is set at 50, 100, 150, 200, 250 and 300 vehicles. As in Figure 3, we can see that when the number of vehicles increases in all scenario, in which the hop distance is vary from 50, 75 and 100 m, the throughput reduces. We can conclude that, a protocol that using binomial distribution achieve better result than uniform distribution. Noted that from the calculation using our mathematical analysis, the uniform distribution has higher collision probability than binomial distribution due to inability of the nodes to select its own backoff integer. Thus the throughput using uniform distribution for each node will be lower. We also can conclude that the time required for backoff procedure is comparatively increase because of the slot selected in the contention window is relatively located toward the rear side. In a multi-hop ad hoc network, if all flows select to use short hop distances to transmit the packets, it will cause more channel contention. This is because the node does not adjust its transmission power down when transmitting to its close neighbors. The reasons why short hop routing is not beneficial as it seems to be is because of short hop transmission does not help spectrum reuse and achieves less bit-distance compare to long hop transmission. On the contrary, if all flows select to use long hop transmission, then it will cause more hidden node interference. This argument indicates that there exists an optimal transmission range for maximizing throughput [15]. In a nutshell, it can be concluded that better throughput should be achieved with longer hop distance [16, 17].

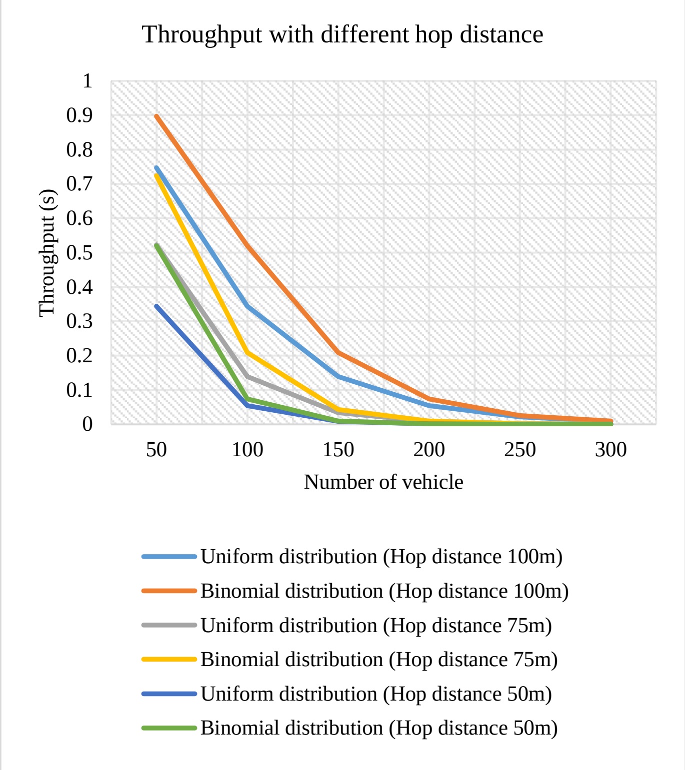

Fig. 3: Throughput with different number of vehicles using hop distance of 50, 75 and 100m

As in Figure 4, we change the hop distance d from 50, 75 and 100 m and investigate the delay for uniform and binomial distribution for the number of vehicles at 50, 100, 150, 200, 250 and 300 vehicles. We can conclude in all scenarios, a protocol that using binomial distribution achieve slightly better result than uniform distribution, in which for a hop distance for 50m, 75m and 100m, the difference between both distributions is only 0.67%, 2.06% and 6.41%, respectively. From our developed mathematical model for end-to-end delay analysis, we can say that the delay is sensitive to the vehicle density. We can see that in a situation where the vehicle density is large, for an instance at 300 vehicles, the delay always increases with decreasing range of hop distance. This is because, when the contention is high, the data dissemination will be decelerated, thus resulting in high message collision ratio and long contention delay [13].

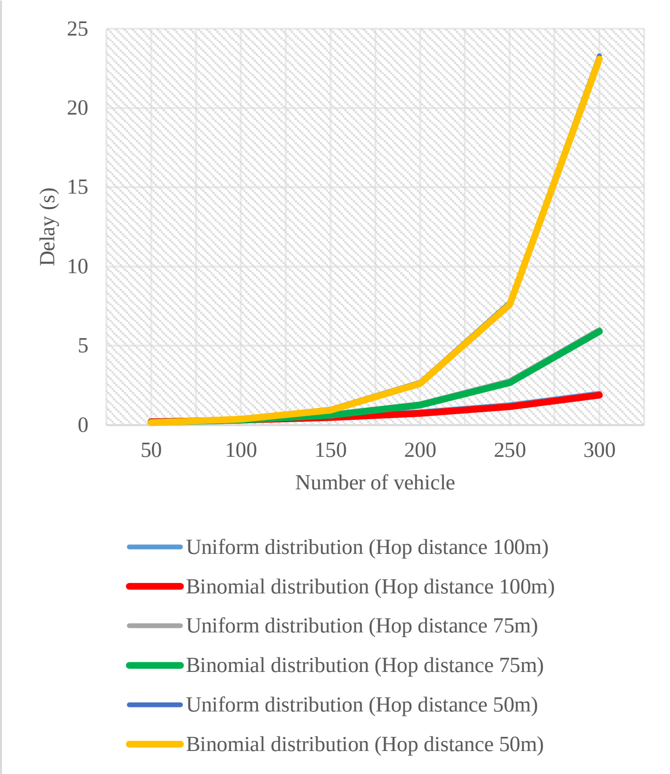

Fig. 4: Delay with different number of vehicles using hop distance of 50, 75 and 100m

From Figure 5, we can infer that when the number of vehicles are increase from 50, 100, 150, 200, 250 and 300, the collision probability will also increase. Noticeably the collision probability will always grow up with the increasing in number of data flows. In all scenario where the hop distance being changed from 50, 75 and 100 m, we can see that the collision probability of binomial distribution is less than uniform distribution. This is because, nodes can select their backoff integer in order to avoid on choosing the same backoff integer as the other nodes do. It means, each node has enter different channels in order to reduce the possibility of collision. In a nutshell, we can say that in a situation where the vehicle density is large, the collision probability will always increases, as well as the shorter range of hop distance.

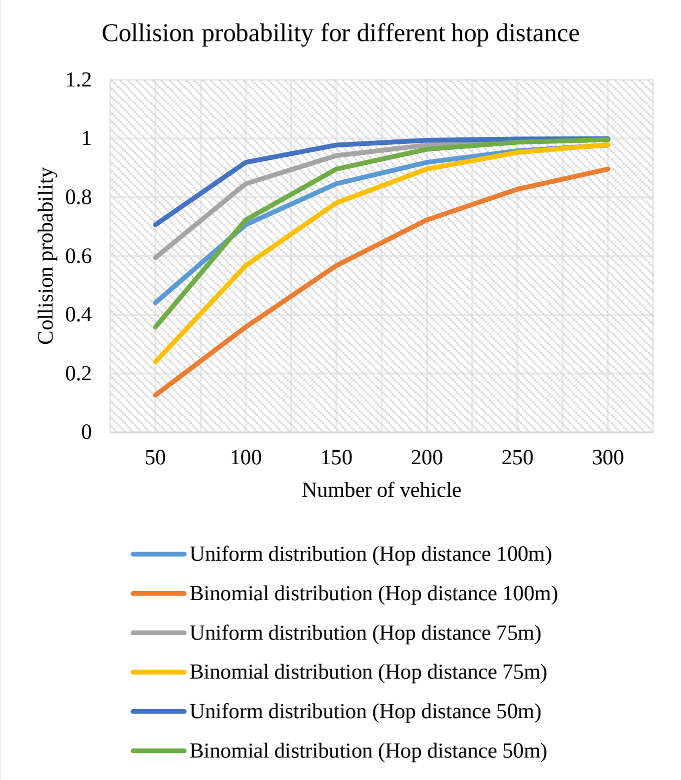

Fig. 5: Collision probability with different number of vehicles using hop distance of 50, 75 and 100m

VI. Result and Analysis

This paper presented the analysis of two different distribution which is uniform distribution and non-uniformly distributed backoff timer based on binomial algorithm for contention window. We determine the throughput, delay and collision probability for uniform and binomial distribution for different hop distance of 50 m, 75 m and 100 m in order to see the differences of using different value of hop distance during data transmission process, while the number of nodes are set at 50, 100, 150, 200, 250 and 300 vehicles. In a nutshell, the simulation results show that binomial distribution outperforms the conventional uniform distribution in terms of throughput, delay and collision probability in all scenarios. We can observed that by using longer hop distance, the better performance of throughput, delay and collision probability have been achieved in both distributions, especially by using binomial distribution. Thus we can say that the binomial probability distribution that is specifically designed to replace the conventional uniform probability distribution in order to differentiate the selection probability during the backoff process of selecting the transmission slot can be implemented in the future. For future work, we will evaluate and study more on the behaviour of binomial backoff algorithm, in addition we can test the protocol performance with real large scale of test bed experiment using NS-2.

References

[1] C.Y. Kuo, Y.H. Kuang, and K.C. Lin, "Performance Enhancement of IEEE 802.11 DCF using Novel Backoff Algorithm," Springer EURASIP Journal on Wireless Communications and Networking, 2012.

[2] G. Shanti, S.M. Kumar, P. Sucharita, J.K. Kumar, and B.N. Bhandari, "Performance Analysis of IEEE 802.11e under Binomial Backoff Algorithm," IEEE International Advance Computing Conference (IACC), pp. 267-270, 2014.

[3] V. Tiwari, "Performance Enhancement in VANET with Admission Control and Contention Window Adjustment," Master of Applied Science Thesis, University of Victoria, Canada, 2012.

[4] W. Alasmary and W. Zhuang, "The Mobility Impact in IEEE 802.11p Infrastructure less Vehicular Networks," Elsevier Ad Hoc Networks Journal, vol. 10, no. 2, pp. 222-230, 2012.

[5] W. Ye, J. Heidemann, and D. Estrin, "Medium Access Control with Coordinated Adaptive Sleeping for Wireless Sensor Networks," IEEE/ACM Transactions on Networking, vol. 12, no. 3, pp. 493–506, 2004.

[6] B. J. Park, H. S. Yoo, and S. H. Kim, "Performance Analysis of Enhanced MAC Protocol to Improve Utilization in Multimedia Service," The Journal of the Institute of Webcasting, Internet and Telecommunication, vol. 9, no. 5, pp. 15–21, 2009.

[7] M. Gast, "802. 11Wireless Networks: The Definitive Guide, "O'Reilly Media", Sebastopol California USA, Second Edition, 2005.

[8] J. Polastre, J. Hill, and D. Culler, "Versatile Low Power Media Access for Wireless Sensor Networks," Proceedings of the Second International Conference on Embedded Networked Sensor Systems, pp. 95–107, 2004.

[9] L. Myungsub and H.P. Chang, "A Differential Probability Selection MAC Protocol Considering Energy Consumption in Wireless Sensor Networks," "International Journal of Distributed Sensor Networks, vol. 2013.

[10] Y.J. Wu, T.H. Yen, and L.J Chen, "A Modified Backoff Algorithm with Dynamical Adjustment on Contention Window in VANETs", Recent Advances in Telecommunications and Circuit Design, 2013.

[11] J. Wroughton and T. Cole, "Distinguishing between Binomial, Hypergeometric and Negative Binomial Distributions," Journal of Statistics Education, vol. 21, no. 1, 2013.

[12] H. Jiawei, H. Yi, and W. Jianxin, "Vehicle Density Based Forwarding Protocol for Safety Message Broadcast in VANET," The Scientific World Journal", vol. 2014.

[13] H. Jiawei, H. Yi, and W. Jianxin, "Vehicle Density Based Forwarding Protocol for Safety Message Broadcast in VANET," The Scientific World Journal", vol. 2014.

[14] G. Bianchi, "Performance Analysis of the IEEE 802.11 Distributed Coordination Function," Journal on Selected Areas in Communications, pp. 535–547, 2000.

[15] J. Deng, S. Yunghsiang, P.N. Chen and P.K. Varshney, "Optimal Transmission Range for Wireless Ad Hoc Networks Based on Energy Efficiency", IEEE Transactions on Communications, vol. 55, no. 9, 2007.

[16] Y. Gou, D.M. Chiu and J.C. Lui, "The Fundamental Role of Hop Distance in IEEE802.11 Multi-Hop Ad Hoc Networks," 13th IEEE International Conference on Network Protocols, 2005.

[17] H.J. Nardelli, H. Alyes, H.M. Carlos, M. Latva, "Throughput Maximization in Multi-Hop Wireless Networks under Secrecy Constraint," CSIT, 2015.

Network Packet Data Online Processing for Intrusion Detection System

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***Abstract*— The existence of intrusion detection systems in computer network as a part of network security tool is very important, with this tool a computer system can detect the intrusion action before it makes more damage. Traditional intrusion detection system using the rule that created by expert for detecting the intrusion, but because of the increasing of internet activity the data to be analyzed in order to establish that rules become large and create the possibility the new intrusion technique cannot detect. Using data mining techniques to find intrusion pattern from network packet data was success to detect intrusion in offline environment, but the effective intrusion detection system must able to detect the intrusion in online environment. Therefore needed a method that can be used to perform online processing of network packets data. This paper discusses the data processing network packets to establish the connection records are complete or incomplete in an effort to enabling the intrusion detection system detecting the intrusion online and based on the test result this method was success to detect the intrusion in online environment.**

Keywords—intrusion detection; network-packet data processing; online processing

I. INTRODUCTION (*HEADING 1*)

Intrusion detection system is a system used to detect the attack or intrusion on a network or computer system. Generally, intrusion detection can be done by matching the pattern or network traffic with the known attack patterns (misuse), or by looking for abnormal traffic patterns (anomaly). To determine the patterns is intrusion or normal activity patterns requires intuition and experience from an expert to make rules in accordance with an intrusion. This causes the intrusion detection system has limitations to detect new types of attacks [1].

The increase of internet activity and attacks against computer systems make the volume of network packets data that must be analyzed become large, this causes new problems for analysts [2]. Applying data mining for finding intrusion or normal activity patterns can be the solution for the problem in processing large volume data. Data mining has the ability to process large amounts of data and by using data mining can help to integrate both of intrusion detection techniques. [3].

Many research has been done and promoting the intrusion detection technique using data mining, many of that research was success to apply in offline environment, where the network data packet was collect first by using network monitoring tools and then analyzed with data mining technique. The problem with this approach is in terms of performance. If using this approach there is the possibility the computer system had been damaged first before intrusion detected. A good intrusion detection system should be able to detect the intrusion before further damage occurs, and this is only possible if the intrusion detection system can detect intrusion in the online environment. Thus the main problem in the implementation of data mining for intrusion detection systems are not only the problem of how to extract features from the network packets data but also how to process the network packet data online. If the network packet data can be processed online then there is a possibility of intrusion detection can also be work in online environment.

II. PREVIOUS RESEARCH

W. Lee [1] in his research introduces how to build temporal features and statistics to detect intrusion. These features can be categorized into two categories: time-based features and host-based features. Time-based features consist of same-host features and same-service features. Features same-host feature is used to check the connection in the last 2 seconds that have the same destination host while feature same-service used to check connections in the last 2 seconds that have the same service. Host-based features are used to detect these types of attacks that cannot be detected in a span of 2 seconds, for example slow probing attack. Host-based features built similar to time-based features but the feature does not use a host-based span of 2 seconds, but using a range of 100 connections. The development of these features produce 41 feature applied in the dataset DARPA KDD'98[4] that construct dataset DARPA KDD'99. This research succeeded detect intrusion that exist in the offline data, but when tested in an online environment found several issues that cause the approach taken in the offline environment cannot be applied to the online environment. The first problem is the connection will not be checked if it is incomplete or not given a termination signal causes a delay in intrusion detection for connection with a long duration, this often happens in the TCP protocol. The second problem was it

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took a long time to process network data packets from the network with high traffic, often occurs when the network data packets containing denial of service attacks. So we need a way to process the network data packets online in order to detect the intrusion quickly.

In some of the results showed that of the 41 feature dataset DARPA KDD'99 not all of this feature are related [5] [6], so it necessary to do the features selection which will certainly reduce the dimensions of the dataset and improving computing performance. K. S. Shrivastava and P. Jain [5] preprocess the DARPA KDD'99 dataset using rough set theory to select the related feature and then classify that data using support vector machine (SVM), the result of this research is 6 features that give the best performance in classification. The feature are: src_bytes, dst_bytes, count, srv_count, dst_srv_count, dst_host_same_src_port_rate.

S. Benferhat, Sedki K., and K. Tabia [7] proposed a method for processing the network packet data for enabling the intrusion detection system can work in online or real-time environment. In their research the audit data or connection record for one connections is made complete or incomplete. The network packet data are processed within a specific time, when the overall time to process the data is less than the specified time then the data connection will be made complete and if in excess of the specified time then the data connection will be made incomplete. If the connection record made incomplete, the rest of the data that has not been processed are ignored.

III. Contribution

This research focuses on the processing of data for TCP, UDP, and ICMP, for detect intrusion with category probe and denial of service (Dos). Data processing network packets created by implementing the function to build complete or incomplete connection records to establish the features that found in K. S. Shrivastava and P. Jain [5] research.

IV. Methodology

Method for processing data packets in this research are shown in figure 1. Network monitoring tool captures data packets that passing through the network and stores it in the form of packet capture (pcap) files. Each of pcap file contain network packet data information with duration *t* seconds.

The next process is extracting header information. This process aims to extract header information from network packet data that contained in pcap file. Header information that is extracted is as follows:

- *Time*, the arrival time of data packets.
- *Source IP address*, the source host IP address
- *Destination IP address*, the destination host IP address
- *IP Protocol*, Indicates the type of protocol used (UDP, TCP, ICMP)
- *Source port*, the port number of the source host sends the related TCP or UDP segment.
- *Destination port,* the destination host port number that receives the related TCP or UDP segment.
- *length*, indicates the TCP,UDP or ICMP protocol segment length that are sent
- *TCP stream*, indicates the number of stream or connection of TCP protocol.
- *ICMP sequence*, sequence number of the ICMP packet is sent
- *ICMP type,* indicates the type of ICMP that are transmitted.

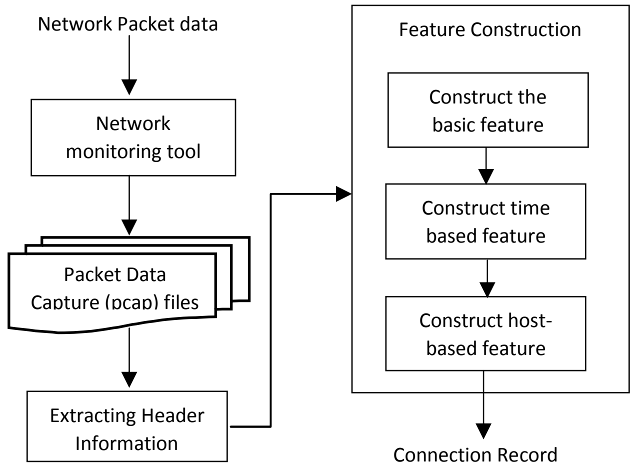

Fig. 1. Network packet data processing

After the header information of the data packets is obtained, the next step is forming connection records and their features. To declare a connection on TCP data packets can be based on information of TCP stream number. The same TCP stream numbers mean the same connection. For data packets of UDP and ICMP protocol because they are a connectionless protocol, each packet of data can be treated as a connection, unless it has a response such as DNS requests and ICMP echo request then the request and response or reply must be treated as one connection instead of two connection. To declare a connection on UDP protocol can be based on information destination host, source host, destination port and source port and for ICMP protocol can be based on ICMP sequence number, the same sequence number is expressed as one connection. The steps to construct the connection record features is as follows:

1) Construct the basic feature (src_bytes and dst_bytes). This process perform summation size of data packets that sent and received by the host. For TCP Protocol obtained by summing the TCP length in one connection. UDP protocol obtained by summing the UDP length and for ICMP protocol by summing the ICMP frame length.

2) Construct time-based features (count, srv_count). Feature count obtained by counting the number of connections that occur on a source host and destination host within last 2 seconds and features srv_count is counting connections from a source host to a destination host with the same service port number within last 2 seconds.

3) Construct host-based features (dst_host_srv_count, dst_host_same_src_port_rate). Features dst_host_srv_count obtained by counting the number of connections to a destination host with the same service port number in the range 100 last connection and to feature dst_host_same_src_port_rate obtained by calculating the percentage of connections to a destination host with the same source port number.

To enable online processing then performed time limitation for header information extraction process and features of the construction process. This will cause the connection record will be formed complete and incomplete. Connection will be formed complete if the overall connection information from its inception until the termination is in one pcap file and can be extracted and processed less than or equal with specified time limit, if these conditions are not met then the connection record will be formed incomplete.

After feature construction process done the next step depends on the purpose of connection records construction, if the purpose is to create training data then the next step is the process of data labeling and data cleansing from duplicate data, if it is intended for the detection of intrusion then the next step is the classification of data based on a model obtained from the training process into 3 class: normal, probe, DoS.

V. EXPERIMENTS AND RESULTS

In the implementation, Tshark was used as a network monitoring tool and Memory storage engine from MySQL was used to store data during the process of features construction. Tshark is a network protocol analyzer, with this tool we can capture packet data from live network and save it in pcap format, or read packets from capture file. Storage engine Memory storage allows us to store data in memory, so it can improve the performance of the system in the feature construction process.

Tshark capture network data packets and store them in the form of pcap files with a duration of 2 seconds. pcap file number that can be stored in one moment is limited to 5 files, this performed to prevent the buildup of pcap file that must be extracted and processed which resulted in delays in the intrusion detection. Figure 2 shows the commands used to run tshark.

```
tshark  -b duration:2 -b files:5 -i $INTERFACE\
-w ./svm.pcap 2> $LOG_DIR/tshark.log &
```

Fig. 2. Tshark command

The time limit for extracting and building features in this test is set to 5 seconds. Before the data connection records processed for training or classified, first performed normalization process. Normalization method used for in this test is the min-max normalization. Support vector machine is used to as a method to classify the connection record in three classes: normal, DoS, and probe. For the implementation of SVM, in this test is using libsvm [8].

The establishment of dataset for train the system performed by online processing of network packet data for normal activity in accessing services in a computer system and intrusion acts to a target host. For intrusion category probe the dataset was construct by capturing and processing network packet data that generated by Zenmap. For DoS attack network packet data was generated by following tools: ping, hping, letdown, udp.pl, slowloris.pl. The dataset that formed in this test amounted to 90224 connection record consisting of: 21058 normal class, 63690 probe class, and 5476 DoS class. For data test that used in the offline testing amounted to 17535 connection records consisting of 14217 normal class, 533 probe class, and the 2785 DoS class.

There are three measure that used to measure the performance of intrusion detection systems, namely: attack detection rate (ADR), false positive rate (FPR), and the accuracy rate (ACC) [6].

$$ADR = \frac{\text{Total number of attacks}}{\text{Total number of detected attacks}} \quad 100\% \tag{1}$$

$$FPR = \frac{\text{Total number of misclassified processes}}{\text{Total number of normal processes}} \quad 100\% \tag{2}$$

$$ACC = \frac{\text{Total number of correct classified processes}}{\text{Total number of processes}} \quad 100\% \tag{3}$$

The first step in this test is offline testing. Offline testing is intended to find the best SVM model to be implemented into the online environment. Determination of SVM model parameters performed manually by changing the kernel type and value of the parameter C. For measuring the classifier model performance is done in two ways: k-fold cross validation and use data test. From the offline test results obtained two models which provide a high level of performance, namely: a model with polynomial kernel of type C = 100 (M1) and model with kernel type linear and C parameter = 1 (M2). Performance comparison of that two models can be seen in table 1.

TABLE I. OFFLINE TEST RESULT

Model	Kernel type	C Param.	ACC (%)	ADR (%)	FPR (%)
M1	polynomial	100	98,46	98,16	0,56
M2	linear	1	90,14	66,60	3,48

Both models from offline test results are then applied the online system for online testing. The performance of both models in online test are shown in table 2.

TABLE II. MODEL PERFOMANCE IN ONLINE TEST

Model	ACC (%)	ADR (%)	FPR (%)
M1	86,38 %	82,05	7,17
M2	89,68 %	88,37	8,63

The ability of each model to detect intrusion action can be seen in Table 3. Based on this result we can see that M1 have a bad performance in detecting intense scan plus UDP and slow comprehensive scan attack but have a good performance in detecting another attack than M2, so based on this fact the model selection is important for getting a good performance. Model with good accuracy rate not guarantee that will give good attack detection for each attack.

TABLE III. COMPARISON OF THE ABILITY TO DETECT INTRUSION

Intrusion	**Intrusion type**	**Record count**	**Attack Detection Rate (%)**	
			M1	M2
Intense Scan	Probe	545	99,27	97,80
Intense Scan Plus UDP		1945	32,08	78,66
Intense Scan All TCP Ports		3534	98,50	98,56
Intense Scan No Ping		391	98,98	96,93
Regular Scan		457	97,81	96,72
Quick Scan		227	95,15	92,95
Quick Scan Plus		284	90,49	80,63
Slow Comprehensive Scan		2645	17,13	70,66
Ping -l 7500	DoS	3030	92,64	93,53
Hping		2896	86,60	85,70
Letdown		3368	85,39	84,56
UDP.pl		3852	95,43	96,81
Slowloris		2003	77,18	75,24

VI. CONCLUSION AND FUTURE RESEARCH

Based on the test results online can be said the implementation of complete / incomplete connection record successfully made intrusion detection systems running on-line environment. The establishment of incomplete connection record causes not all the existing network packet data in a pcap file is extracted and processed, this raises the possibility of undetected intrusion, especially on a network that has high traffic. In addition the amount of data successfully extracted and processed will depend on the computer system performance.

For the next research should be considered an effective way to improve the extraction and processing of data in the online environment for improving the ability of intrusion detection. Based on the online test result we can see that model that we use to classify the connection record will give the different performance, so it is important to try another model or method.

REFERENCES

[1] W. Lee,"A Data Mining Framework for Constructing Features and Models for Intrusion Detection Systems", Thesis, School of Arts and Sciences Columbia University, New York, 1999

[2] E. Bloedorn, D. Christiansen, W. Hill, C. Skorupka, L. Talbot, and J. Tivel, "Data Mining for Network Intrusion Detection: How to Get Started", MITRE Technical Report, 2001

[3] A. Agathou and T. Tzouramanis,2008, "The Role Of Data mining in Intrusion Detection Technology", Garson, G.D. dan Khosrow-Pour,M (ed.), Handbook of Research on Public Information Technology, Vol.1, Information Science Reference, Hershey,New York, 2008

[4] DARPA Intrusion Detection Data Sets, http://www.ll.mit.edu/ideval/data

[5] K. S. Shrivastava and P. Jain, "Effective Anomaly Based Intrusion Detection using Rough Set Theory and Support Vector Machine", International Journal of Computer Applications, Vol.18-No 3, 2011, pp. 35-41

[6] Rung-Ching Chen, Kai-Fan Cheng and Chia-Fen Hsieh, "Using Rough Set And Support Vector Machine for Network Intrusion Detection", International Journal of Network Security & Its Applications (IJNSA),Vol 1, 2009.

[7] S.Benferhat, K. Sedki, and K. Tabia, "Preprocessing Rough Network Traffic For Intrusion Detection Purposes", IADIS International Telecommunications, Networks and Systems, 2007, pp.105-109,

[8] LIBSVM -- A Library for Support Vector Machines, www.csie.ntu.edu.tw/~cjlin/libsvm/

Optimizing The Utilization of Container Truck Transportation

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***Abstract*— The crowdedness of transportation of containers made multi-depot and multi-terminal necessary in the system design. To optimize the route of container truck, Ant Colony Optimization (ACO) is proposed in this paper. In managing route of the truck, ACO works to decrease the number of container from 10 to 4 truck which imply to reduce the consumption of gasoline. There are several algorithm of ACO that are used to find the optimal routing of the container which are Ant system (AS), Elitist Ant System (EAS), Max-Min Ant System (MMAS), Rank-Based Ant System and Ant Colony System (ACS). The results of simulation show that Rank-Based Ant System has the best time over all methods for this transportation problem.**

Keywords—container truck; Ant Colony Optimization; optimal routing

I. Introduction

Jakarta International Container Terminal (JICT) covers a total of 100 hectares and is the largest container terminal in Indonesia. To optimize service and support the growth of the national economy, JICT has embarked on a number of expansion projects since 2008, including the addition of quay and yard equipment, the implementation of a sophisticated terminal operating system and the first Auto gate system in Indonesia Port. According to the data of Unit Angkutan Khusus Pelabuhan (Anguspel) DKI Jakarta, there were 9,900 unit container truck during 2012 and increased about 9% percent in 2013. A survey of LPEM-FEUI in 2008 shows that transportation cost in Indonesia reaches the second position in the total cost of shipping. In order to develop a better system that can solve the problem in container terminal, there is need to optimize the model of the transport container truck. While in this point, optimal routing determination of the container trucks can be a solution. The goal of this optimization is to minimize the total operation of container truck.

Ant Colony Optimization (ACO) as an approachment to stochastic optimization methods is investigated in this paper. Ant Colony Optimization (ACO) is used to determine route of truck container. Ant Colony Optimization (ACO) is based on the foraging patterns of ants. Where ACO use the ability of the ant to always find the shortest path between their nest and a food source[2,3].

II. System Modeling

A. Model Identification

The crowdedness of transportation of containers made multi-depot and multi-terminal necessary in the system design. Depot is a place to store empty containers and to park the trucks. Generally, there are four types of containers, namely, inbound full (IF), inbound empty (IE), outbound full (OF), and outbound empty (OE). Container IF is a container that needs to be picked up in the terminal, loaded onto the truck which then unloaded at the recipient's place, the designated location. After being unloaded in the designated location, the container became an empty container which needs to be returned to the depot. Meanwhile, container OF is a container that needs to be transported to the customer. An empty container needs to be transported to the customer for loading purpose and be brought back to the terminal. Container IE needs to be picked up in the terminal and transported to depot. On the other hand, container OE is a container that needs to be transported to the terminal. Inbound and outbound activities are two regular activities in container transportation. Terms "origin" and destination are used to replace terminal and customer place to easily illustrate the problem. The destination of the container is the recipient and the origin of the container OF is the sender. The origin and the destination of these four types of containers are summarized in Table 1.

All terminals and customer's locations, whether the customer is a sender or a recipient, have time window for each container, depot, on the other hand, does not have time window. Consequently, all loaded inbound and outbound containers have two time windows, namely, time window in the origin and the destination. Meanwhile, empty inbound and outbound containers have one time window.

A set of activities that are related to the origin of container has to begin within the time window of the origin. For the container IF, the set of activities consists of the storing of the container in the terminal and the transportation to the recipient. For container OF, the set of activities in the origin consists of the transportation of empty container, the loading of materials into the container and the transportation of the container from the sender to the terminal. On the other hand,

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the container IE, activity in the origin is uplifting the empty container.

Table 1. Source and destination of the container truck

	IF	***OF***	***IE***	***OE***
Source Destination	Terminal Receiver	Sender Terminal	Terminal Depot	Depot Terminal

Likewise, a set of activities related to the destination of the container needs to begin within the time window of the destination. For container IF, this set of activity consists of materials unloading from the container located in the sender and the transportation of the empty container. For container OF, a set of activities in the destination is to store the container in the terminal. For container OE, the activity in the destination is to store the container in the terminal.

The system design focuses on the optimal routing design of the trucks of container in a particular period and on the order, timing and which truck is needed to execute the container's transportation. The objective is to minimize the total operation time and idle time in the sender/recipient location.

B. Directed Graph

The optimal routing design of the trucks of container is mathematically shown by a graph, G={V,A}. A set of points is $V=V_D \cup V_C$. In this case V_D is a set of starting point and returning point is shown by point 1 and 2. V_C is container points, namely, at point 3, 4 and 5. Meanwhile, a set of arc is A= {(i,j)| i ∈ V_D ,j ∈ V_C; atau i ∈ $V_D \cup V_C$ }. This is illustrated by Fig 1. The graph is formulated based on the set of activities. In formulating the graph, process of finding the solution involves the finding of the optimal routing (arc). This arc connects one container point to another, or one starting/returning point to one container point.

Every starting/returning point, $i \in V_D$ of depot i, where *i*=1,...,m. The attribute of the starting/returning point is the amount of trucks located in the depot. A container point, $i \in V_C$, is a set of activity of the associated container, where *i*=m+1,...,m+n. Details of this can be seen in Table 2. The associated index *i* is used for depot (*i*=1,...,m) and container (*i*=m+1,...,m) since either depot or container is associated to onepoint.

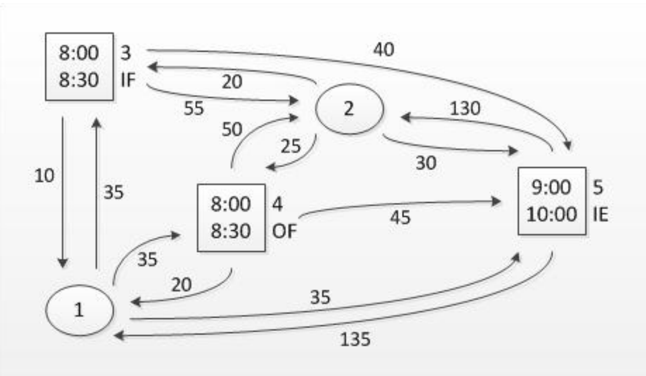

Figure 1. Directed Graph

Table 2. Serial process in point of the container

Type	Set of activity
IF	Picking up container in the terminal From terminal to recipient Unloading material at the recipient
OF	Loading materials at the sender From sender to terminal Storing container in the terminal
IE	Picking up container in the terminal
OE	Storing container in the terminal

Two time window at the point i need to be combined with considering the service time in the source and destination. Time window $[a_i,b_i]$ is a combination between time window at the source $[a_{Oi},b_{Oi}]$ and time window at the destination $[a_{Di},b_{Di}]$. time window and time process is get from (1),(2) and (3). While the transfer time is formulated by (4).

$$a_i = \begin{cases} \min[\max(a_{Oi}, a_{Di} - t_{Oi} - t(O_i, D_i)), b_{Oi}] & \text{jika } P_i \in \{'IF', 'OF'\} \\ a_{Oi} & \text{jika } P_i \in \{'IE'\} \\ a_{Di} & \text{jika } P_i \in \{'OE'\} \end{cases} \quad (1)$$

$$b_i = \begin{cases} \min(b_{Oi}, b_{Di} - t_{Oi} - t(O_i, D_i)) & \text{jika } P_i \in \{'IF', 'OF'\} \\ b_{Oi} & \text{jika } P_i \in \{'IE'\} \\ b_{Di} & \text{jika } P_i \in \{'OE'\} \end{cases} \quad (2)$$

$$\tau_i = \begin{cases} \max([a_{Di} - b_{Oi}], [t_{Oi} + t(O_i, D_i)]) + t_{Di} & \text{jika } P_i \in \{'IF', 'OF'\} \\ t_{Oi} & \text{jika } P_i \in \{'IE'\} \\ t_{Di} & \text{jika } P_i \in \{'OE'\} \end{cases} \quad (3)$$

$$T_{ij} = \begin{cases} \tau_{ij}, & i \in V_D, j \in V_C \\ \tau_{ij} + \tau_i, & i \in V_C, j \in V_D \cup V_C \end{cases} \quad (4)$$

In comparing the system with and without optimization, number of truck that need to deliver the container and the composite time is very important. here, by using optimization number of the truck can be reduced from 10 to 4 truck.

III. Ant Colony Optimization

Informally, an ACO algorithm can be imagined as the interplay of three procedures : ConstructAntsSolutions, UpdatePheromones, and DaemonActions. These procedures is described in pseudo-code as seen below [1]:

```
procedure ACOMetaheuristic
    ScheduleActivities
        ConstructAntsSolutions
        UpdatePheromones
        DaemonActions % optional
    end-ScheduleActivities
end-procedure
```

There are several algorithm of ACO that is used to find the optimal routing of the container which are Ant system (AS), Elitist Ant System (EAS), Max-Min Ant System (MMAS), Rank-Based Ant System and Ant Colony System (ACS).

A. *Ant system (AS)*

Two phase in AS algorithm are construct solution and update pheromones. in the begining ants are placed at in each points where the position is selected randomly. next the ant k-th does random proportional action to select next point to visit. probability the ant move from point i to poin t :

$$p_{ij}^{k} = \frac{[\tau_{ij}]^{\alpha}[\eta_{ij}]^{\beta}}{\sum_{l\in\mathcal{N}_i^k}[\tau_{il}]^{\alpha}[\eta_{il}]^{\beta}},\ \text{jika}\ j \in \mathcal{N}_i^k \quad (5)$$

The pheromones are updated in two ways: firstly, the pheromones are subject to an evaporation factor (ρ), which allows the ants to forget their past and avoid being trapped in a local minima (6) and secondly, they are updated in relation to the quality of their tour (7,8).

$$\tau_{ij} \leftarrow (1-\rho)\tau_{ij}, \quad \forall(i,j) \in L \quad (6)$$

$$\tau_{ij} \leftarrow \tau_{ij} + \sum_{k=1}^{m}\Delta\tau_{ij}^{k}, \qquad \forall(i,j) \in L \quad (7)$$

$$\Delta\tau_{ij}^{k} = \begin{cases} 1/c^{k} & \text{if arc (i,j) belongs to } T^{k} \\ 0 & \text{otherwise} \end{cases} \quad (8)$$

*S*imulation using computation program shows minimum operation time for Ant System can be seen in Fig. 2. For all algorithms, model system uses 100 containers and 1000 iterations. Using 10 agent ant, the best time is 22,651 minutes with 49 trucks.

B. *Elitist Ant System (EAS)*

There is an extra gain on the solution T^{bs}, which can be able by adding e/C^{bs} to its route. in this algorithm, the formula above (7) is changed to :

$$\tau_{ij} \leftarrow \tau_{ij} + \sum_{k=1}^{m}\Delta\tau_{ij}^{k} + e\Delta\tau_{ij}^{bs}, \quad (9)$$

so $\Delta\tau_{ij}^{bs}$ can be find with next equation.

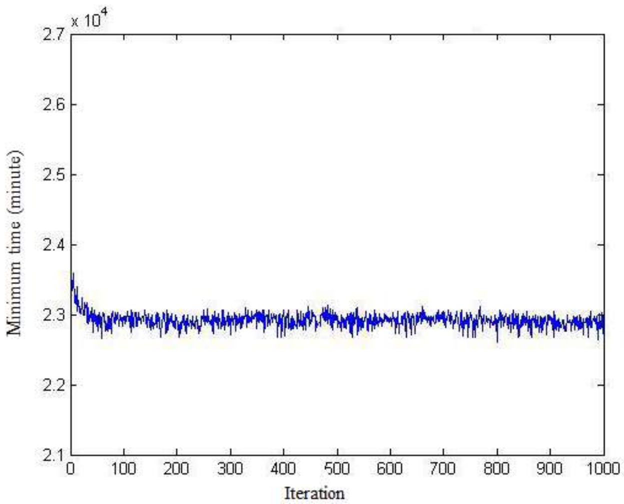

Figure 2. Minimum time using AS

Simulation using computation program shows minimum operation time for EAS can be seen in Fig. 3. Where the minimum time is 22,194 minutes with 44 trucks.

$$\Delta\tau_{ij}^{bs} = \begin{cases} 1/C^{bs} & \forall(i,j) \in T^{bs} \\ 0 & \text{otherwise} \end{cases} \quad (10)$$

C. *Rank-Based Ant System*

There is a step before updating pheromones where the ants are sorted based on the quality of solution and number of extra pheromones of ants with weighting factor according to the rank of the ant. In every iteration, only (w-1) ant with best rank dan best solution so far that can add pheromones. Updating pheromones using this algorithm is shown by (11)

$$\tau_{ij} \leftarrow \tau_{ij} + \sum_{r=1}^{w-1}(w-r)\Delta\tau_{ij}^{r} + w\Delta\tau_{ij}^{bs} \quad (11)$$

Simulation using computation program shows minimum operation time for Rank-Based Ant System can be seen in Fig. 4. Using 10 agent ants, the best time is 22,067 minutes with 44 trucks.

D. *Max-Min Ant System (MMAS)*

The key to achieve best performance of ACO algorithms is to combine an improved exploitation of the best solutions found during the search with an effective mechanism for avoiding early search stagnation. MMAS, which has been specifically developed to meet these requirements, differs in three key aspects from AS: to exploit the best solutions found during an iteration or during the run of the algorithm; to avoid stagnation of the search the range of possible pheromone trails on each solution component is limited to an interval; additionally, to initialize he pheromone trails to τ_{max}[5].

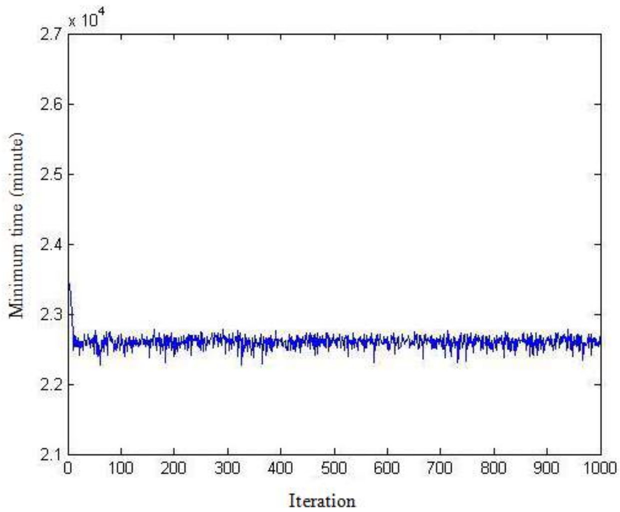

Figure 3. Minimum time using EAS

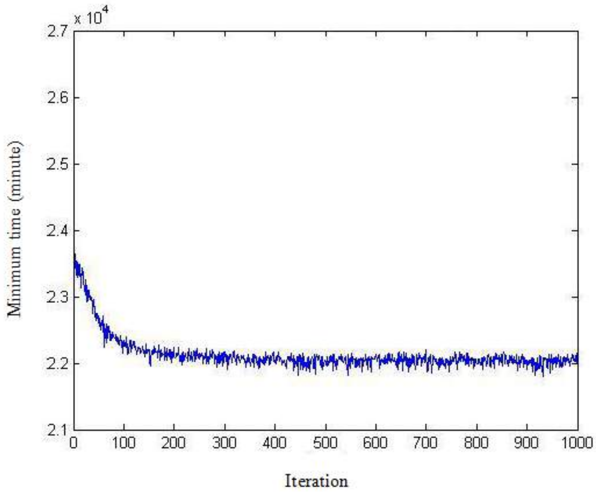

Figure 4. Minimum time using Rank-Based Ant System

Updating pheromones for this algorithm is :

$$\tau_{ij} \leftarrow \tau_{ij} + \Delta\tau_{ij}^{best} \tag{12}$$

Simulation using computation program shows minimum operation time for MMAS can be seen in Fig. 5. Using 10 agent ants, the best time is 22,395 minutes with 44 trucks.

E. Ant Colony System (ACS)

There are theree main points in ACS that make it different than others. First, ACS uses the accumulation experiences of ants. Second, evaporation pheromones and extra pheromones only to the best.Third, everytime an ant uses move from i toj, the ant diminishes the amount of pheromone on visited edges, making them less desirable for future ants) allows for the search of new. So there only one ant, the best ant, can add pheromone every iteration.

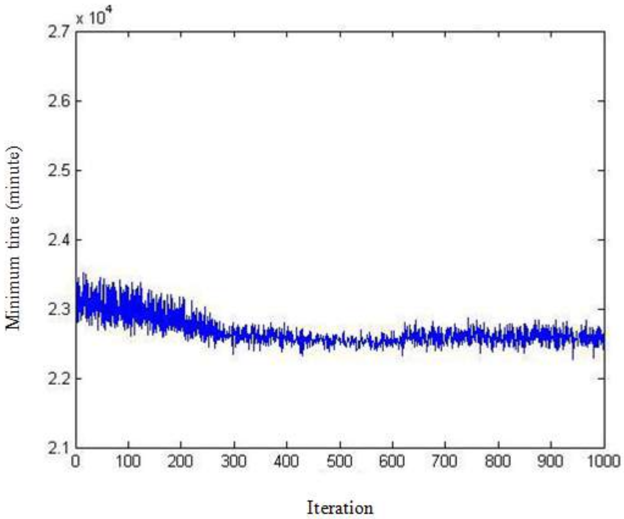

Figure 5. Minimum time using Max-Min Ant System

Updating pheromone is derivated from :

$$\tau_{ij} \leftarrow (1-\rho)\tau_{ij} + \rho\Delta\tau_{ij}^{bs}, \quad \forall(i,j) \in T^{bs} \tag{13}$$

Simulation using computation program shows minimum operation time for MMAS can be seen in Fig. 6. Using 10 agent ants, the best time is 22,714 minutes with 48 trucks.

While Fig.2 - Fig. 6 shows for 100 container, the greater number of container, 500 unit is shown in Fig.7. The minimum time is 95,075 minutes with 196 trucks. this best time is reaced while the iteration is 5,000.

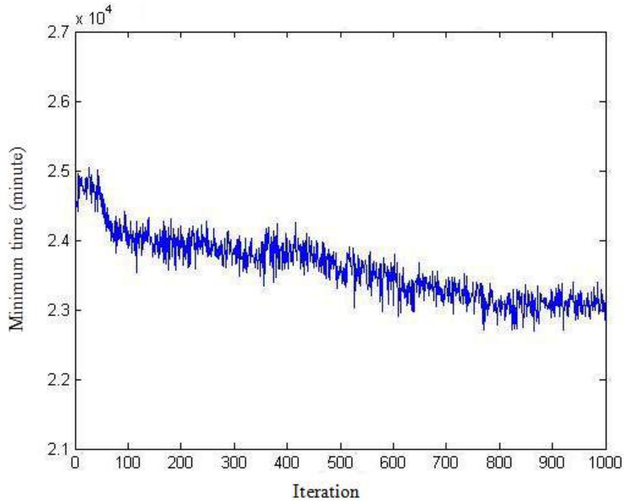

Figure 6. Minimum time using Ant Colony System

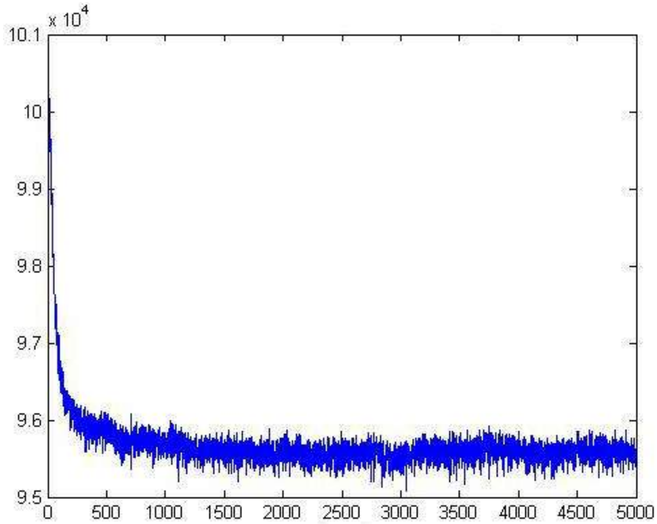

Figure 7. Minimum time with 5,000 iteration

IV. CONCLUSION

Ant Colony Optimization is a potensial algorithm in transportation system especially truck container. Optimazation in route of the truck gives postive result, where the system shows 4 trucks can deliver 10 containers which can save the of gasoline of 6 trucks. In simulating the five algorithm of ACO, it shows that Rank-Based Ant System has the best time among others. There is tendency in raising the iteration, the result of minimum time is getting better.

REFERENCES

[1] Dorigo, M. & Stutzel, T. 2004. *Ant Colony Optimization*. A Bradford Book – The MIT Press. Cambridge, Massachusetts

[2] Dorigo, M., dkk. 2006. *Ant Colony Optimization : Artificial Ants as a Computational Intelligence Technique*, dalam IEEE Computational Intelligence Magazine. IEEE.

[3] Karl . Jones, Bouffet A. Comparison of Ant Colony Optimisation and Differential Evolution. International Conference on Computer Systems and Technologies. 2007.

[4] Zhang, dkk. 2010. *Heuristic-based truck scheduling for inland container transportation*, at OR Spectrum (2010) 32:787-808. Springer – Verlag.

[5] Stutzle, T. Max-Min Ant System. Elsevier : Future Generation Computer Systems 16. 2000.

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The Design Requirements for Libyan Imaging Mini-Satellite (LibyaSat-1)

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***Abstract*— In this paper we present the conceptual design of Libyan remote sensing satellite (LibyaSat-1) and its sub-systems requirements. LibyaSat-1 is a 300 kg mini satellite, which will be used to support high resolution multi-spectral earth imaging camera to fulfill the civilian needs. This satellite will operate at LEO of 775 km and will provide a resolution of 2.5 m for the panchromatic band and 10 m for the VIS/NIR bands with 30 km swath. We have presented the mission overview, mission operation concept and mission requirements. Moreover, the System Tool Kit (STK) simulation is used to show the ground trucks of LibyaSat-1 for three days and to find the contact numbers between LibyaSat-1 and both Murezeq and Tripoli stations. We have also presented the design of telemetry and command subsystem, code and data handling subsystem, electrical power subsystem, altitude orbit control subsystem, and structure subsystem.**

Keywords—mini-satellite; large satellie; sub-systems; Low Earth Orbit (LEO); payload.

I. INTRODUCTION

Large (conventional) polar orbiting satellites have a wet mass larger than 1000 kg and operate at altitudes of about 1000 km. These satellites are sun-synchronous, which means that any object on the orbit or the movement of the satellite will appear in the same position when viewed from the sun [1, 2]. This is important as it provides a constant amount of sunlight to solar cells on the satellite. Conventional satellites are mainly used for high resolution images. One example is the Quick-Bird satellite, which high spatial resolution, i.e., panchromatic is 61 cm GSD at Nadir and Multispectral is 2.4 meter GSD at Nadir [3]. As set out in Table 1, these large satellites are really costly, heavy and consume large amount of power.

Mini Satellites are small, have low power consumption and cheap as compared to conventional satellites. They have a mass ranging from 100 to 500 kg and consume power of about 53.2 W [4, 5]. They play a very important role in the field of remote sensing and they perform similar tasks to conventional satellites. The applications of remote sensing include land use, agriculture, environment monitoring, disaster assessment, research, and national security. An example of mini satellite is FORMOSAT-2, which is a Taiwanese imagery satellite with a mass of 693 kg (without original propellant) and operate in the Low Earth Orbit (LEO) at altitude of 891 km [3]. Another example is SPOT 5, which is a French remote sensing mini satellite. It is a sun-synchronous imaginary mini satellite that orbits at altitude 832 km. This satellite provides high resolution images for Libyan Centre for Remote Sensing and Space Science (LCRSSS) as part of commercial operations between France and Libya [6].

LCRSSS established in 1989 and is a governmental research organization, which belongs to a government space organization dedicated to the researches in remote sensing, space science, and earthquakes. This organization does not have its own remote sensing satellite and all images of Libya are taken by French SPOT satellites and then send to Libyan ground station for image processing. Fig. 1 shows an image of Tripoli that was captured by SPOT 6. In this paper, we present the design requirements for the most important sub-systems of the first Libyan mini satellite (LibyaSat-1). We will present conceptual design of LibyaSat-1 and the design requirements for the most important sub-systems. We will also list the designed and achieved budget for each sub-system.

Fig. 1. Image of Tripoli taken by SPOT 6

TABLE 1 COMPARISON BETWEEN SATELLITES

Types	Mass (kg)	Cost (US $)	Power Consumption
Conventional	>1000	0.1-2 B	~ 1000 W
Medium	500-1000	50-100 M	~ 800 W
Mini	100-500	10-50 M	53.2W
Micro	10-100	2-10 M	35 W

II. MISSION OVERVIEW

The main mission of the LibyaSat-1 satellite under the remote sensing program is to acquire and monitor terrestrial, marine environment, desertification and resources throughout Libyan area and its surrounding waters, and possibly over other regions of the world for international cooperation, via satellite imaging data to fulfill the civilian needs.

- The remote sensing satellite program is one of the major satellite programs to be executed by Libyan center for remote sensing and space science for the first one phase space program of Libya.
- The remote sensing instrument is a push-broom electrical-optical type of sensor with spectral bands in the visible (VIS) and near infrared (NIR). The spectral bands include panchromatic and four

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Land sat type multi-spectral bands. The ground sampling distances (GSD) are 2.5 m for the panchromatic band and 10 m for the VIS/NIR bands respectively.

- The swath is 30 km for all selected bands.

Table 2 shows the simulation results after testing LibyaSat-1 conceptual design at different altitudes. We can see that LibyaSat-1 is assumed to be sun-synchronous with an altitude of 775 km which can communicate with the ground station using RF links.

TABLE 2 LIBYASAT-1 CHARACTERISTTICS

Repeat (day)	561Km (SSO)	**775Km (SSO)**	888Km (SSO)	1000Km (SSO)
Repeat (day)	1	**3**	1	???
Rev/day	15	**14 1/3**	14	13.661
Circular Velocity (Km/Sec)	7.579	**7.465**	7.406	7.35
Ground Velocity (Km/Sec)		**6.750**		
Period (Min)	95.877	**100.347**	102.734213	105.1186
Eccentricity	0	**0**	0	0
Inclination (deg)	97.635	**98.497**	98.98	99.478
Width (Km) (FOR:+- 45 Deg)	1178	**1662.9**	1928.8	2200.1

III. Mission Requirments and Satellite Characteristics

Table 3 shows the key mission requirements and the systems specifications of LibyaSat-1. LibyaSat-1 satellite is expected to operate in sun synchronous LEO at altitude of 775 km with a total mass of no more than 300 kg and data storage volume of no less than 80 Gbits. The minimum expected life time of LibyaSat-1 is 5 years.

1) *Agility*

Roll and Pitch: The spacecraft bus shall be capable of providing a cross-track viewing tasking rate better than 24 degrees within 60 seconds regardless initial point.

Yaw: The spacecraft bus shall be capable of providing a Yaw viewing tasking rate better than 7 degrees within 60 seconds regardless of the initial point.

2) *Satellite Reliability*

0.6 at the end of the 5-year mission.

3) *Remote Sensing Insturment*

Panchromatic (PAN) with 4 Visible/NIR bands.

TABLE 3 MISSION REQUIREMENTS SPECIFICATIONS

Parameter	**Specification**	
RSI Mass (kg)	95	
Swath (km)	30	
RSI Power (watts) (Orbit Average)	75	
Download Rate (Mbits/s)	150	
	PAN (Panchromatic)	MS (Multispectral)
GSD (M) (Ground Sampling Distance)	2.5	10
Pixel (numbers)	12000	3000
A/D (Bits)	12	12
Mass For Satellite lift-off weight	No more than 300kg total weight (MAX)	
Lifetime	5 years	
Agility	Roll and Pitch: 24 deg/60s Yaw: 7 deg/60s	
Launcher	Falcon 1e	
Data storage volume (Gbits)	80	
FOR	+-45 deg	
Downlink Rate (X Band)	150 Mbps	
Downlink Rate (S Band)	2.0 Mbps	
Local Time	10:00 am ~ 10:30 am	
Reliability	0.6	

IV. System Engeering

A. *Operational Concept*

The libyansat-1 operations are to be conducted from Tripoli Ground System (TGS). The TGS is composed of the Mission Operations Center (MOC), the Telemetry, Tracking, and Command (TT&C) stations, a Flight Dynamics Facility (FDF), a Mission Control Center (MCC), a Science Control Center (SCC), a Ground Communications Network (GCN), an X band Data Receiving Only Station, and an Image Processing Center (MOC) personnel, which will monitor and command the spacecraft using real-time health and status data. Command uplink files will be generated by MOC personnel, forwarded to the TT&C station and transmitted on an uplink data stream. Fig. 2 shows the location of TGS and MGS.

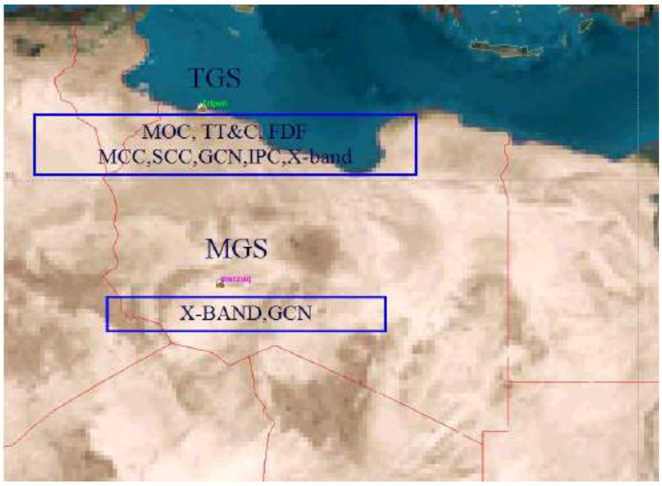

Fig. 2. The geographical locations of TGS and MGS.

There will be 2~4 contacts of the spacecraft with the TT&C stations on Libya per day. The remote sensing instrument will operate mainly over the region of Libya. Moreover, the equatorial descending time is around 10:00 am to 10:30 am local time. The baseline operation mode is simultaneous panchromatic and multi-spectral imaging, with real-time downlink data at a rate up to 150 Mbps via a dedicated X band channel. The STK simulation [7] is used to test LibyaSat-1 in the orbit.

Fig. 3, 4, 5 and 6 show the ground tracks for 3 days. LibyaSat-1 traces out a path on the earth surface, called its ground track, as it moves across the sky. As the earth below is rotating, the satellite traces out a different path on the ground in each subsequent cycle. We see that after three days the satellite traces out many different paths and covers most of the area.

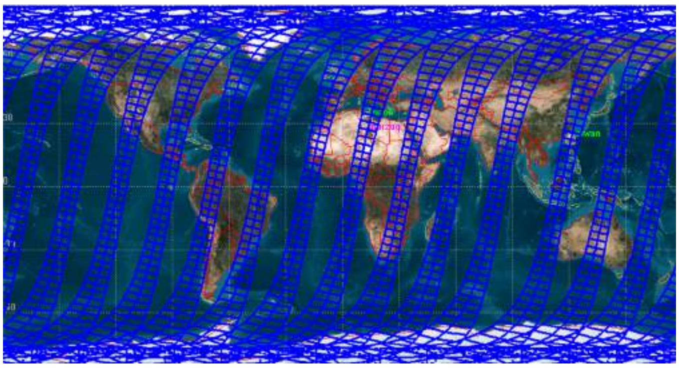

Fig. 3. Ground tracks of the first day

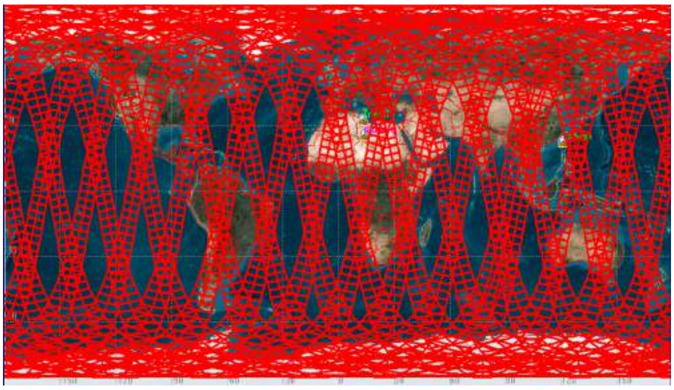

Fig. 4. Ground tracks of the second day

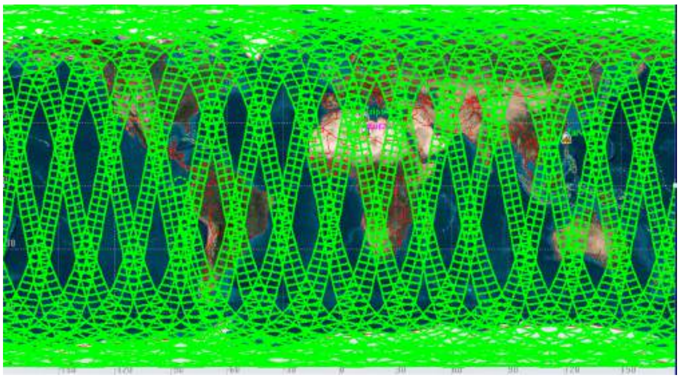

Fig. 5. Ground tracks of the third day

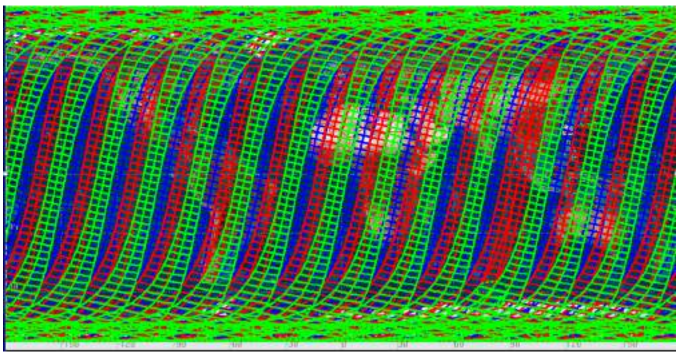

Fig. 6. Ground tracks (repeat period,3 days)

Fig. 7 and 8 show the contact number and time of LibyaSat-1 with Murezeq and Tripoli stations. We see that there are two contacts in the first and the third days between LibyaSat-1 and Tripoli and Murezeq stations at two different times. In the second day there is four contacts with both stations, which is a most double as compared to the first and the third days

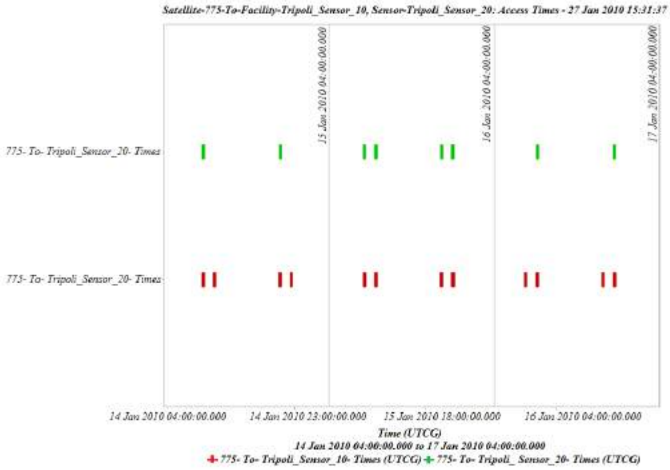

Fig. 7. The contact numbers between LibyaSat-1 at 775 km orbit and both Murezeq and Tripoli stations

775-To-Tripoli_Sensor_10

	Access	Start Time (UTCG)	Stop Time (UTCG)	Duration (sec)
	1	14 Jan 2010 09:37:17.751	14 Jan 2010 09:47:35.004	617.253
	2	14 Jan 2010 11:20:04.752	14 Jan 2010 11:21:51.770	107.018
	3	14 Jan 2010 20:48:58.915	14 Jan 2010 20:59:12.024	613.109
	4	14 Jan 2010 22:33:03.964	14 Jan 2010 22:34:20.195	76.232
	5	15 Jan 2010 09:04:39.166	15 Jan 2010 09:13:57.074	557.908
	6	15 Jan 2010 10:44:20.699	15 Jan 2010 10:52:14.526	473.827
	7	15 Jan 2010 20:16:40.001	15 Jan 2010 20:25:40.952	540.951
	8	15 Jan 2010 21:56:15.126	15 Jan 2010 22:04:28.490	493.364
	9	16 Jan 2010 08:33:01.662	16 Jan 2010 08:39:01.652	359.991
	10	16 Jan 2010 10:10:31.201	16 Jan 2010 10:20:22.505	591.304
	11	16 Jan 2010 19:45:32.739	16 Jan 2010 19:51:17.419	344.680
	12	16 Jan 2010 21:22:08.836	16 Jan 2010 21:32:10.814	601.978
Global Statistics				
Min Duration	4	14 Jan 2010 22:33:03.964	14 Jan 2010 22:34:20.195	76.232
Max Duration	1	14 Jan 2010 09:37:17.751	14 Jan 2010 09:47:35.004	617.253
Mean Duration				448.135
Total Duration				5377.615

775-To-Tripoli_Sensor_20

	Access	Start Time (UTCG)	Stop Time (UTCG)	Duration (sec)
	1	14 Jan 2010 09:38:50.293	14 Jan 2010 09:46:03.943	433.650
	2	14 Jan 2010 20:50:30.470	14 Jan 2010 20:57:39.118	428.647
	3	15 Jan 2010 09:06:29.176	15 Jan 2010 09:12:08.841	339.665
	4	15 Jan 2010 10:46:48.744	15 Jan 2010 10:49:47.483	178.739
	5	15 Jan 2010 20:18:31.374	15 Jan 2010 20:23:48.470	317.096
	6	15 Jan 2010 21:58:34.222	15 Jan 2010 22:02:07.476	213.254
	7	16 Jan 2010 10:12:08.895	16 Jan 2010 10:18:46.037	397.142
	8	16 Jan 2010 21:23:43.869	16 Jan 2010 21:30:34.167	410.298
Global Statistics				
Min Duration	4	15 Jan 2010 10:46:48.744	15 Jan 2010 10:49:47.483	178.739
Max Duration	1	14 Jan 2010 09:38:50.293	14 Jan 2010 09:46:03.943	433.650
Mean Duration				339.811
Total Duration				2718.492

Fig. 8. Contact time of LibyaSat-1 at 775 km with Murezeq and Tripoli stations

B. Telecom, Telemetry and Command Subsystem

The TT&C subsystem comprises of onboard s-band (2.039 GHz) receiver and s-band transmitter (2.215 GHz), transmitter for uplink and down link respectively. A s-band receiver is used to receive command signal from Murezeq ground station while s-band transmitter antenna is used to send a set of signals that show the status of the LibyaSat-1 resources and health, i.e., temperature, voltage and currents. All these signals are sent to the control center via ground station using RF system; see Fig. 9.

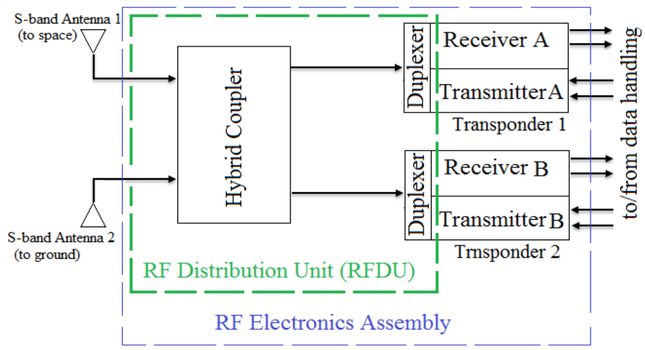

Fig. 9. TT&C subsystem block diagramof LibyaSat-1

The x-band subsystem is used for image data downlink. The 5.4 m x-band dish antenna and its pointing mechanism are used at Murezeq ground station. The link budgets for s-band up and down links for LibyaSat-1 are calculated and presented in [6]. Table 4 shows the up and down link budgets for s-band antennas.

TABLE 4 S-BAND UP AND DOWN LINK DUDGETS

Feature	Uplink	Downlink
Frequency	2.039 GHz	2.215 GHz
Satellite Altitude	775 km	775 km
Elevation Angel (deg)	10^0	10^0
Transmitter Power(Pt)	20 dBW	7 dBW
Ground Sation Antenna Gain (Gt)	32.3 dB	6.4 dB
EIRP Power	51.3 dBW	13.4 dBW
Space Loss	-165.93 dB	-166.63 dB
Atmospheric Losses	-0.5 dB	-0.5 dB
Rain Loss	-0.5 dB	-0.5 dB
Polarization Loss	-0.3 dB	-0.3 dB
Pointing Loss	-0.2 dB	-0.2 dB
On-Board Losses	-10 dB	-4.5 dB
System Noise Temperature(Ts)	27.9 dB/k	24.6 dB/k
Satellite Antenna Gain	5 dB	5 dB
Figure of Merit (G/T)	-22.88 dB/k	-18.2 dB/k
Bite rates	32 Kbps	2000 Kbps
C/No Received	79.57 dB-Hz	77.07 dB-Hz
Eb /No Received	34.52 dB	14 dB
Bit Error Rate (BER)	10^{-5}	10^{-5}
Required Eb /No	11.8 dB	9.6 dB
Implementation Loss	-2.5 dB	-1 dB
Margin	20.72 dB	3.4 dB

C. Code and Data Handling Subsystem

The Command and Data Subsystem is the "brain" of mini satellite. It is used to process and store information and data from all satellite sub-systems. It also sends commands to all sub-systems and equipment. The function of this sub-system is as following:

- ➢ Receives, validates, decodes, and distributes commands to other spacecraft subsystems.
- ➢ Gather, processes and formats spacecraft housekeeping and mission data for downlink or to be used by an on-board computer.
- ➢ The ideal C&DH system is one which has previously been proven on another spacecraft and which requires no or little modification for the mission under development (i.e. Heritage).

1) C&DH Requirements

a) Computer system

- *Processor:* up to 20 MIPS, with fault tolerance and supports real-time operating systems, i.e., eCOS, VxWorks.
- *SRAM / SDRAM:* 128 MBytes, protected with EDAC (single error correction, double error detection).
- *EEPROM:* 4 MBytes, storage of application code.
- *PROM / EEPROM:* 64 KBytes bootloader, SSR 50Gbits for SHO (Status of health), and file management raw image data.
- *Real Time Clock:* 1-Hz with synchronization to GPS clock.
- *FDIR:* Supervisory circuit / Watchdog timer.
- *Redundancy:* Full or partial implementation (TBD).

b) I/O Interfaces

- *Attitude Control Subsystem:* 2 x Sun sensor, 2 x Magnetometer, 4 x Gyroscope, 2 x Star camera, 2 x GPS receiver, 3 x Magnet torque, and 4 x Reaction wheel.
- *Thermal Control: up to 80 thermistors.*

c) Telemetry and Control

- *S- Band* :

  1- CCSDS (Consultative Committee for Space Data Systems) communication protocol with uplink of 32 Kbps and downlink of 2 Mbps.

  2- Telecommand decoder, with local memory for command storage, authentication and configuration.

  3- Compliant with packet telemetry and channel coding standards CCSD102.0-B-5 and CCSDS101.0-B-5.

- *X-Band (Payload):* 150 Mbps (Downlink) – shown here for completeness.

d) Power Specification

- *28V unregulated voltage input*
- *Over-current and under-voltage protection*
- *DCDC switch-mode converters and linear regulators:* 5V, 3.3V and possibly 1.5V (for on-board electronics) and +/-12V for Analog.
- *Power consumption:* 45 Watts, in continuous mode (exact figure is to be determined) and about 1.8 A maximum input current (continuous mode).

e) Miscellaneous:

- *Mass:* 20kg (approx.).
- *Operating Condition:* military (i.e. -55°C to 125°C).
- *Component Selection:* space or military-grade

- *Radiation-Hardened:* up to 50KRad, depending on mission requirements (e.g. low-earth orbit, mission life).

D. Electrical Power Subsystem

Electronic Power Sub-system (EPS) is designed to provide a sufficient power to all subsystems including payload to ensure a reliable operations of all equipment and components during the mission. EPS consists of batteries, solar panel, Battery Charge Regulator (BCR), Power Conditioning Module (PCM), Power distribution Unit (PDU), and Battery Monitoring (BM); see Fig. 10. The multi-junction GalnP/GaAa solar cells are chosen to be used as a power generator for LibyaSat-1 and the battery will be Lithium-ion. Table 5 and 6 show different types of solar cells and batteries.

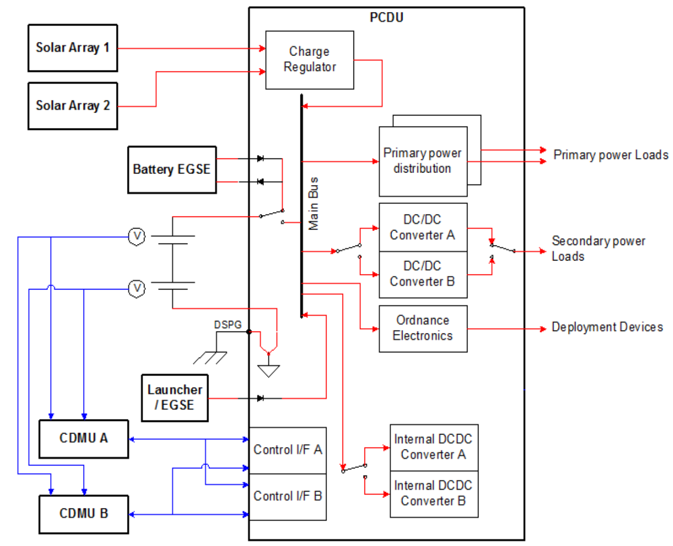

Fig. 10. EPS Architecture

TABLE 5 DIFFERENT TYPES OF SOLAR CELLS

Cell Type	Silicon	Gallium Arsenide	Multi-junction Galnp	Indium phosphide
Efficiency	14%	18%	27%	19%
Radiation degradation	High	Medium	Medium	Low
Cost	Low	High	High	High

TABLE 6 DIFFERENT BATTERY TYPES

Secondary Battery Couple	Specific energy density (W. hr/kg)	Status
Nickel-cadmium	15-30	Space-qualified, extensive database
Nickel-hydrogen (individual pressure vessel)	35-43	Space-qualified, good database
Nickel-hydrogen (common pressure vessel)	40-56	Space-qualified for GEO and planetary
Nickel-hydrogen (common pressure vessel)	43-57	Space-qualified
Lithium-ion (LiS02, LiSF, LiS0CI2)	70-110	Space-qualified
Sodium-sulfur	140-210	Under development

1) EPS requirements

- 3% margin from the whole power budget for every orbit.
- Primary power bus voltage: unregulated power of 28v (+5.6 V/-4 V); see Fig. 11.
- Secondary power bus voltage: using converters to reduce 28 V to:

  a) +4.8v ~ +5.6 V

  b) +14.25 ~ +15.75 V

  c) -15.75v ~ -14.25 V

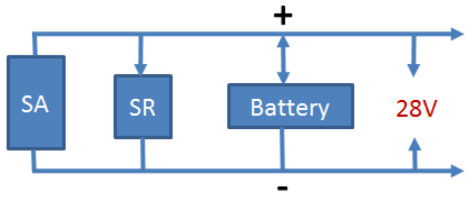

Fig. 11. DET System

- Battery low voltage protection: to prevent S/C from recovery until solar arrays get sufficient power and charge batteries.
- Battery high voltage protection: battery voltage > 34 V, terminate all charge regulators.
- Battery charge management: the operating system for batteries charging must be done automatically by special flight software without a ground uplink command.
- Power distribution short circuit protection: all equipment and payload instruments must be over- current protected using Solid-state switch (MOSFET).
- Deployment control using NEA.
- Redundancy: PCDU shall be cold redundant except for power distribution and charge regulator.

For LEO we expect the battery's DOD to be 40-60% for NiH2 technology compared to 10-20 % for NiCd. We base these expectations on the average DOD over 24 hours and assume the batteries are fully recharged at least once during this period. The number of batteries, N, may be equal to one for this calculation if you simply require a battery capacity .Two to five batteries are typical.

We must have at least two (unless the battery uses redundant cells) because the spacecraft needs redundant operation if one unit failed. But more than five batteries require complex components for recharging.

2) Power Profile

Table 7 shows the mass and power consumption for EPS. These numbers were calculated taking in to account different assumptions such as when we do not operate the payload and x-band transmitter.

Ps/c basic power consumption when we do not operate the payload and the transmitters.

$\mathbf{P_{SRI}}$ remote sensing instrument for power consumption = 50 watt.

Transmitter:

- Power consumption for X-band transmitter 55 watt & APM 5 watt.
- Power consumption for S-band transmitter 35 watt.

- S-band receiver power consumption 8 watt is included in Ps/c as this receiver will be always on.

Fig 12. below gives a good idea about the operation of the power profile.

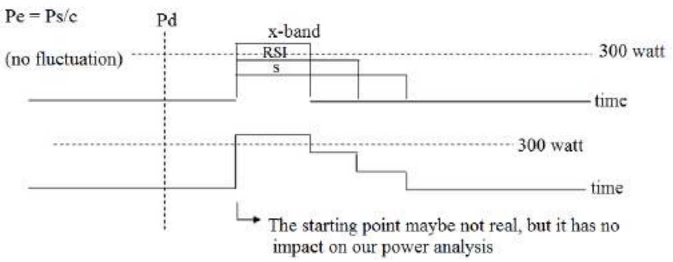

Fig. 12. The operation of power profile for s/c

TABLE 7 MASS AND POWER CONSUMPTION FOR EPS COMPONENTS

Component	Mass	Consumption
Battery	7 kg	0 watt (passive device)
S/A	16 kg	0 watt (generator device)
PCDU	15 kg	17 watt
Harness	22 kg	3 watt(dissipation power)

E. Altitude Orbit Control Subsystem

The aim of Attitude and Orbit Control Sub-systems (AOCS) is to control the attitude of the micro satellite depends on the operation mood in the mission orbit by using different sensors and actuators. As set out in Table 8, AOCS provides an altitude pointing accuracy and control stability of better than 0.20 (3δ) for allaxis and pointing knowledge less than 0.036^0.

TABLE 8 AOCS DESIGN PARAMETERS

Pointing accuracy	< 0.2 deg -- 3δ ,3D
Pointing knowledge	< 0.036 deg
Positioning knowledge	< 100 m
Jitter	< 0.33 arc sec/ 373.7 µsec
Agility	Cross track maneuver rate should be better than 30 degrees within 60 seconds
	Along track maneuver rate should be better than 30 degrees within 60 seconds
	Yaw maneuver rate should be 10 degrees within 60 seconds

F. Structure Subsystem

This Subsystem provides mechanical support and alignment for all equipment and components on the mini satellite. It is also used to provide a thermal conductivity and its body is used as a ground for electrical devices. Fig. 13 shows the configuration of different components on LibyaSat-1. Moreover, Tables 9 and 10 show the mass budget and selected material respectively.

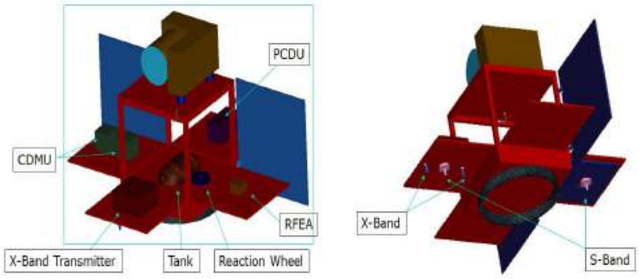

Fig. 13. The proposed components layout of LibyaSat-1

TABLE 9 MASS BUDGET

System	Current mass (kg)	LV Adapter	10
		Bus Total	234
AOCS	34		
C&DH	20	RAL CAM1	50
TTC	13	Total	284
EPS	60	Balance	-
RCS	30	Satellite Total	284
SMS	55	Launch Max.	415
TCS	12	Margins(%)	31.56%

TABLE 10 MATERIAL SELECTION

Item	description
Top panel	Honeycomb Panel
Top frame	Milled Al Alloy Ring
Bottom cone panel	Honeycomb Panel
Tank support panel	Milled Al Alloy Plate
Wall panel	Honeycomb Panel
longeron	Milled Al Alloy Plate

G. Satellite Chararcterstics

Table 11 and 12 show the assigned and obtained budgets for mass and power respectively. We can see that the total mass and power of all designed subsystems is within the budget but the margin is less than the expected. Compared to all sub-systems, the EPS has the higher mass because of the batteries and solar cells. In terms of power consumption, the TTC sub-system has the highest power consumption as compared to all subsystems on LibyaSat-1. This is because of the number of antennas used and their constant state of activeness to keep tracking, download images and communicate with ground station.

TABLE 11 MASS BUDGET FOR ALL SUBSYSTEMS

Sub-system	Mass (kg) System flow-down	Design Mass (kg)
C&DH	42.5(17%)	20
TT&C	22.5(9%)	13
EPS	45(18%)	60
Thermal	10(4%)	12
Structure	37.5(15%)	55
Propulsion	17.5(7%)	30
AOCS	25(10%)	34
FSW	0	0
RSI	50(20%)	50
TOTAL Mass	250	**274**
Max Mass	300	300
Budget	50	**26**

TABLE 12 POWER BUDGET FOR ALL SUBSYSTEMS

Sub-system	Power (W) System flow-down	Design power (W)
C&DH	77.5(31%)	45
TT&C	27.5(11%)	104.3
EPS	17.5(7%)	20
Thermal	27.5(11%)	10
Structure	0	0
Propulsion	12.5(5%)	13.8
AOCS	37.5(15%)	47
FSW	0	0
RSI	50(20%)	50
TOTAL power	250	290.1
Av power	300	300
Budget	50	10.1

V. CONCLUSION

We have presented a conceptual design of LibyaSat-1 and the design requirements of its important subsystem. We have simulated the model of LibyaSat-1 at different LEO altitudes including the 775 km. We found out that the most suitable altitude is 775 km. We have also measured the contact time and number of LibyaSat-1 with both Tripoli and Murezeq stations. The total designed mass and power for all sub-systems were within the assigned budget. The total mass of LibyaSat-1 is 74 kg and total required average power is 190.1 W.

VI. ACKNOWLEDGEMENT

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REFERENCES

[1] C. Underwood, V. Lappas, A. D. S. Curiel, M. Unwin, A. Baker, and M. Sweeting, "Science Mission Scenarios Using "PALMSAT" Pico-Satellite Technology" 18th Annual AIAA/USU Conference on Small Satellites, Logan, Utah, 2004.

[2] F. Em. M. Tubbal, R. Raad, and K-W. Chin, "A Survey and Study on the Suitability of Planar Antennas for Pico Satellite Communications " unpublished.

[3] D. Lu, S. Hetrick, and E. Moran, "Impervious surface mapping with Quickbird imagery," International journal of remote sensing, vol. 32, no.9, pp.2519-2533, 2011.

[4] R. Sandau, "Status and trends of small satellite missions for Earth observation," Acta Astronautica, vol. 66, no.1, pp.1-12, 2010.

[5] F. Em. Tubbal, R. Raad, K-W. Chin, and B. Butters, "S-band Planar Antennas for a CubeSat," International Journal on Electrical Engineering and Informatics, vol. 7, no.4, December 2015.

[6] F. Em. Tubbal, A. Alkaseh, and A. Elarabi, "Telemetry, Tracking and Command Subsystem for LibyaSat-1," IEEE 9th International Conference on Telecommunication System, Services and Application, Bandung, Indonesia, pp. 1-4, November, 2015

[7] Systems Tool Kit (STK) [online] available:http: //www.agi.com/products/stk/.

Calculation WRAN Base Station with Okumura Hata in Banten

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***Abstract*—This paper tried to design the implementation of WRAN 802.22 as a new proposed and interesting standard to provide broadband service in tv white space (unused spectrum digital television), specially in hard to reach and low density population area. With the standard parameter WRAN that have been drafted, we calculate the total amount of site needed in Banten, one of the province in Indonesia. In this paper, we use frequency 650 MHz, 30 dB transmit power, 10 m receivers height and 30 meter of transmitter height. By using okumura hata propagation, Banten needs 245 site to cover all the area.**

***Index Terms*—WRAN, 802.22, coverage, BS, okumura hata, TVWS**

I. Introduction

TODAY, many country in the world in digital switch over process. The migration from analogue to digital technology in tv broadcasting makes some spectrum savings, which we called as digital dividend. While this saving spectrum compete over by many companies (like provider broadband celuller to run Wimax or LTE), WRAN operates just using the spectrum tv that vacant at that time as we called as TV white space (TVWS). In telecommunications, white spaces refer to frequencies allocated to a broadcasting service but not used locally and efficiently.

II. Overview

A. IEEE 802.22 WRAN

802.22 is the first Wireless regional area network (WRAN) standard by IEEE that is expected to be widely used in the future specially in rural areas.. Its designs to work in UHF and VHF band, in range frequency 54-862 MHz, where theres white space in tv bands. The novelty about this standard is, coexistence mechanism between licensed TV band, unlicensed broadband network, and radio TVWS[1]. This new technology also robust to large delay spread, doppler spread, and moreover it has simple and light specs that make it cheaper 10 times than broadband celuller nowadays.

To prevent interference with incumbent users (such as digital tv broadcasting and wireless microphone), WRAN using cognitive radio, an intelligent wireless system that has ability to detecting and using the unused/free available spectrums of the radio. CR effectively reuses the finite frequency resource using spectrum sensing.

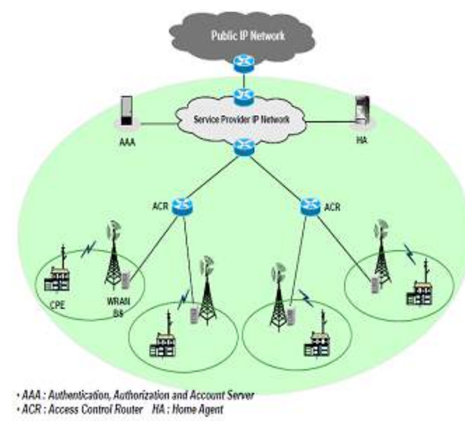

Fig. 1. WRAN Hierarchy [4]

B. WRAN Hierarchy

IEEE 802.22 is point to multipoint system, where the system is similar with the current broadband cellular like 3G or Wimax. Base Station link to end user or Customer premise equipment (CPE) through line of sight (LOS) and non light of sight propagation. The significant differences between these technology are where WRAN has wider cell coverage and coexistence with primary users which have the priority right in the spectrum.

WRAN base station and the CPE both will detect the channel wether its vacant or busy by incumbent user. But all the action and decision only in Base Station side. WRAN operates supported by many technologies, such as: Cognitif radio, Geo location and also incumbent database.

TABLE I
System Parameters of WRAN[3]

Parameters	Spesification	Remarks
Frequency range	54-862 (MHz)	
Channel Bandwith	6,7,8 (MHz)	regulatory domain
Modulation	QPSK 16QAM 64 QAM	
Transmit EIRP	4W max for CPE 4W max for BSs in the USA regulatory domain	Max EIRP for BSs may vary in other regulatory domains
Duplex	TDD	
Multiple Access	OFDMA	

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C. Propagation Model

Okumura Hata is one of the most common propagation model that used for predicting path loss in different area. This model is applicable for frequency between 150-1500 MHz and height of base station around 30-100 m. Its model dividing the area into 4 forms:

- Dense urban: central city/business district, very high building and very high population.
- Urban: high building and characterized by high population density. It may be a city or town, and usually developed by the process of urbanisation.
- Sub urban: its a residential area or inhabited districts located either inside a town or city's outer rim or just outside its official limits and usually has a large community.
- Rural area: a village, open and spread out area, small population (mostly sparse) and has high trees.

Okumura Hata Formula:

$$PL = A + B + C \tag{1}$$

Where A, B and C depends on the frequency, height transmitter and receiver used.

$$A = 69.55 + 26.16logf - 13.82logh_t - a(h_r) \tag{2}$$

$$B = 44.9 - 6.55logh_r \tag{3}$$

where
f in Mhz
For Dense urban:

$$a(h_r) = 3.2(log(11.75h_r)^2 - 4.97 \tag{4}$$

Small and medium cities :

$$a(h_r) = (1.1log(f) - 0.7)h_t - (1.56log(f) - 0.8) \tag{5}$$

C = 0
Sub urban environment:

$$C = -2(log(f/28)^2 - 5.4 \tag{6}$$

Rural area:

$$C = -4.78[log(f)]^2 + 18.33logf - 40.98 \tag{7}$$

D. Radius Cell

Radius of the cell could be found with this formula:

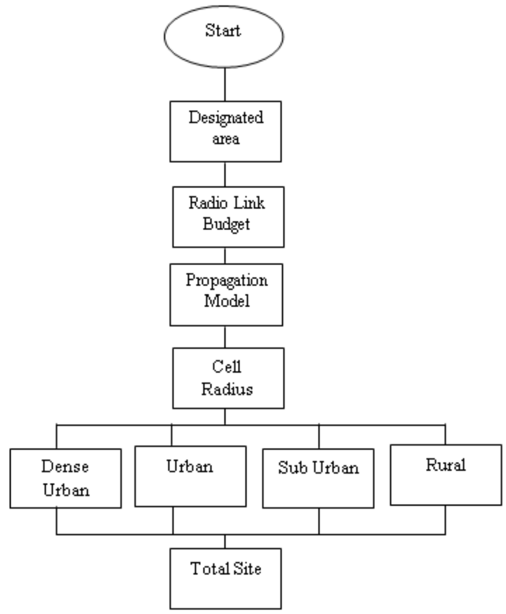
$$A = 2.6R^2 \tag{8}$$

where
A = area
R = radius of the cell

III. DESIGNS AND CALCULATION

A. Flowchart to calculate the number of base station based on coverage

First step, choose area to be analyzed and collecting the data, Determine the radio link budget, match the area with suited propagation model, found radius, and coverage area.



B. Designated Area

Area that choosed to calculate the amount of base station needed to cover awhole area, is Banten Province.

Fig. 2. Map of Banten

Banten is located between 57'50" and 71'11" south latitude and 1051'11" and 1067'12" east longitude. The province has an area of 9,662.92 km.

TABLE II
CITY/REGENCY OF BANTEN PROVINCE

No	City/Regency	Village/Urban Community	Area Land
1	Lebak Regency	340 Villages	3.044,72
2	Pandeglang Regency	335 Villages	2.746,90
3	Serang Regency	246 Villages	1.467,39
4	Tangerang Regency	320 Villages	956,9
5	Cilegon	43 communities	175
6	Serang	66 communities	266,71
7	Tangerang	104 communities	164.54
8	South Tangerang	54 communities	147.19

Banten Province is subdivided into four regencies and four autonomous cities with each regency subdivide into some small vilages, and each city subdivide into some urban community. From the Table 2, we can conclude that the regencies has larger areas than the cities.

C. Radio Link Budget

This design is using frequency 650 MHz, with 30 m base station transmitter, 10 m antenna receiver and transmit power

TABLE III
LINK BUDGET PARAMETERS

Parameter	Value
Frequency	650 MHZ
hTX	30 m
hRX	10 m
Transmit power (dBm)	30
EIRP (dBm)	36
Receiver Sensitivity (dBm)	-93.2
GTX (dB)	9
GRX (dB)	11
Maximum Allowable Path Loss/MAPL (dB)	116.3

30 dBm. Then we get maximum allowable path loss (MAPL) 116.3 dB. To have radius of the cell, it can be done by input the value of maximum allowable path loss into the okumura hata in (1).

Radius cell Base station of cities and regency calculated in Banten could be seen below:

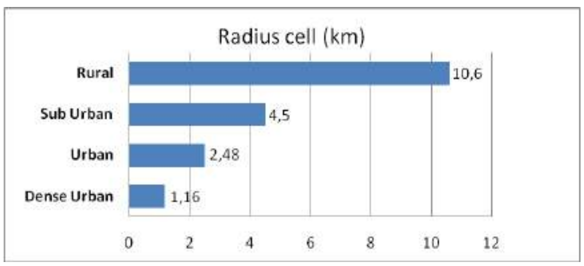


Fig. 3. Radius cell

Dense Urban with 1.16 km has the lowest radius cell out of this 4 type of propagation area by okumura hata. Sub urban and urban area have radius cell about 4.5 km and 2.48 km. While, Rural area surely be the largest with the value of the radius cell till 10.6 km. As we know, rural area and suburban area relatively have smaller population density and theres no much obstacle blocking the transmitter to the CPES receiver, which make its two have cell radius larger. Vice versa, in high rise building city, cpes antenna receive large number of refflected and scattered signal.

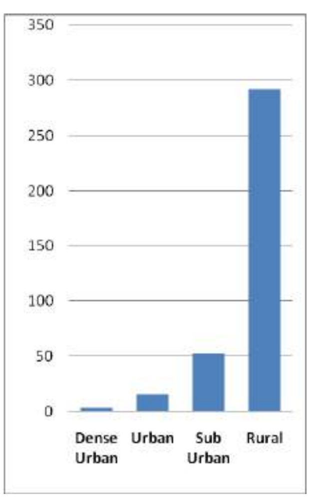


Fig. 4. cell area

The signal come to the receiver by many path. This signal sometimes may be in phase so the signal strength increase, but mostly out of phase and intefere the main signal. It explains why the dense urban has the smallest radius cell.

Using equation (8) we get cell area coverage (km2) as shown in figure 4. In Result, the base station can cover 3.5 km2 in dense urban area, 15.95 km2 in urban, 52.65 in sub urban, and almost 300 km2 in rural area (292 to be exact).

IV. RESULT

The eight city and regency divided into two type of propagation model area. Lebak, Pandeglang, Serang, and Tangerang as Sub Urban area, and the rest as Urban area. By dividing total land area to cell area (with the suite type area propagation), we get the amount of site needed.

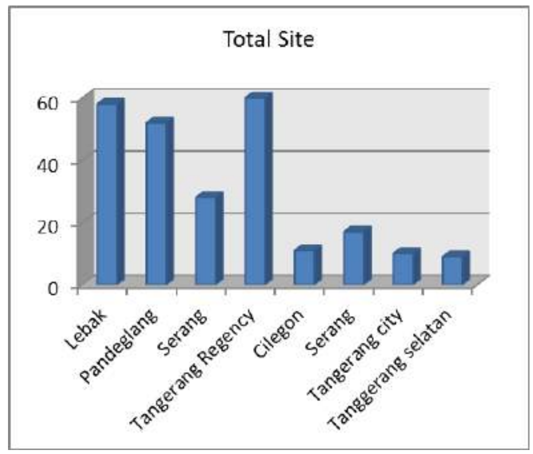


Fig. 5. Total Site

From the calculation, Lebak, Pandeglang, serang and tangerang need 58, 53, 28 and 60 site to cover whole land area. Meanwhile, Cilegon needs 11 site, Serang 17 site, Tangerang and Tangerang Selatan need each 10 and 9 base stations. All of this calculation affected by the value of the path loss. The parameters like power transmit, height of transmitter, height of receiver and any other parameters in link budget could be adjusted to get a better coverage.

From the calculation above we could take conclusion that WRAN could be a new alternatif technology to people in low density population place(rural areas)to access internet and any other application with low cost. Furthermore provider service may not need buy the licenced of the spectrum.

But, beside based on coverage, to run WRAN 802.22 well and properly, we need to calculate the traffic and capacity of base station to get reliable quality of service.

REFERENCES

[1] A.Mishra and D. L W. Johnson,*White Space Communication, Advanced, Development and Engineering Challenge*, Springer, 2015.

[2] G. Choinard, *The IEEE 802.22 WRAN Standard and Its Interface to The White Space Database*. IETFS PAWS 2011.

[3] A.N Mody and G. Choinard, *Enabling Rural Broadband Wireless Access using cognitif Radio Technology*. Doc IEEE 802.22-10/0073r03. 2010.

[4] Y. H. Kim , *IEEE 802.22 WRAN overview*. Kyung Hee University

[5] O.E. Herrera, A. Guttierrez, A. M Ospina.*WRAN and LTE Comparison in Rural Environtments*. Communication Conference (COLCOM). 2012. IEEE Colombian

[6] Z. Lei, and S.H. Hwang.*IEEE 802.22 PHY overview*. Doc IEEE 802.22-10/0106r0. 2010

[7] Z. Xin, et.al.*Preliminary Link Budget Analisys for 802.22b*.Doc.:IEEE 802.22-12-0055-00-000b.2012

[8] S. Rahman, et.al.*Comparative Studi Between Wireless Regional Area Network (IEEE Standard 802.22) and WiMax and Coverage Planningof a Wireless Regional Area Network Using Cognitive Radio Technology*.International Journal of Recent Technology and Engineering. Vol.1, Issue-6, January 2013.

Performance Evaluation of SIMO Least Squares Channel Tracking on Rician Distributed Channel

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Abstract— **Wireless channel with the presence of Line of Sight (LOS) is often modeled as Rician distribution. In order to track the channel for mobile users, exponentially weighted Least Squares (LS) Single Input Multiple Output (SIMO) channel estimation is employed. By incorporating the optimum forgetting, the system performance in terms of tracking Mean Square Error (MSE) and Bit Error Rate (BER) is evaluated. It is shown that the optimum forgetting factor as well as the system performance, are elevation angle-dependent due to the rapid movement of the user in the coverage area.**

Keywords— **Least-Squares (LS), channel estimation, forgetting factor, Rician fading, elevation angle**

I. INTRODUCTION

Communications in a wireless medium creates multipath propagation such that the users experience fading. If LOS is present, the fading channel is modelled as Rician distribution. Moreover, when the user is moving, the K factor as well as the Doppler shift becomes elevation angle-dependent [1][2].

In order to achieve coherent detection, the channel is firstly estimated. There are numerous channel fading compensation techniques available in literatures. Specifically, the Multiple Input Multiple Output (MIMO) channel is tracked by means of Kalman filter for Rician fading with constant LOS in [3]. With the LS approach, tracking the MIMO Rayleigh fading channel is presented in [4] assuming the channel dynamics as i.i.d. first order Markov process, and extension to Rician fading is found in [5]. Considering Rician channel with dominant LOS (K factor 10-25 dB) and the possibility of specular non-LOS (NLOS) component presence, adaptive multiuser MIMO beamforming is developed in [6].

In this paper, we employ the LS SIMO channel estimation assuming the user traverses mainly in rural area such as in High Speed Train (HST) communications. Accordingly, the channel is modelled as flat fading Rician distribution [1][7] with the relation between K factor and elevation angle follows [2]. Based on the analytical tracking MSE [5], the optimum forgetting factor for the interested scenarios is calculated. The system performance in terms of the minimum tracking MSE as a function of Signal to Noise Ratio (SNR) are evaluated. Furthermore, Monte Carlo simulation is performed to determine the BER performance.

This paper is organized as follows. The system model is given in Section II. In Section III, the SIMO LS channel tracking algorithm is presented, which is then followed by the system simulation and simulation results in Section IV. The paper is concluded in section V.

II. SYSTEM MODEL

The downlink communication between the transmitter with $M = 1$ antenna and receiver with $N = 5$ antennas is assumed here. The SIMO system employed is based on the fact that only a single user is served. Therefore, it is not required to produce multiple beams, and accordingly multiple antennas, at the transmitter. In the Rician fading context, the wireless platform is terrestrial system or High Altitutde Platform (HAP). The rest of this section consists of the geometry model, channel model, and transmission system.

A. Geometry Model

The geometry model of the downlink transmission in the presence of LOS is illustrated in Fig. 1. It is assumed that the user motion with speed υ follows a straight line AOB. The distance d denotes the minimum distance from the transmitter to the determined user path AOB. The Direction of Arrival (DOA) of the LOS link is defined as $\theta \in (-90^\circ, 90^\circ)$, whereas the angle between the LOS link and the velocity vector υ is denoted as the elevation angle $\psi \in (0^\circ, 180^\circ)$. Thus, $\theta + \psi = 90^\circ$.

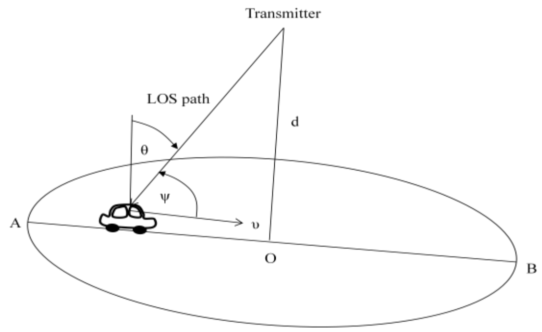

Fig. 1 The downlink geometry model in the presence of LOS

B. Channel Model

On a SIMO transmission structure, the discrete time Rician flat fading channel at time index n is modelled as

$$\boldsymbol{h}[n] = \sqrt{\frac{K}{K+1}}\boldsymbol{h}_{\mathrm{LOS}}[n] + \sqrt{\frac{1}{K+1}}\boldsymbol{h}_{\mathrm{NLOS}}[n] \qquad (1)$$

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where $\boldsymbol{h}_{\text{LOS}} \in \mathbb{C}^{N,1}$ and $\boldsymbol{h}_{\text{NLOS}} \in \mathbb{C}^{N,1}$ denote the LOS channel vector and the NLOS channel vector respectively. In addition, the Rician K factor describes the relative power of the LOS channel w.r.t. the NLOS one.

Assuming a static HAP, the LOS channel vector becomes [7]

$$\boldsymbol{h}_{LOS}[n] = \boldsymbol{a}(\theta)e^{j2\pi F_D T_s \cos\psi\, n} \tag{2}$$

where $\boldsymbol{a}(\theta)$, F_D and T_s are the steering vector, maximum doppler frequency, and symbol period respectively. Additionally, $F_D\cos(\psi)$ is the Doppler shift.

The NLOS channel vector are i.i.d. Rayleigh fading, where their second order statistics are described in [7]. Therein, the second order statistical autocorrelation matrix is independent between the time and spatial domain

$$\boldsymbol{R}[m] = R_{\text{TD}}[m]\boldsymbol{R}_{\text{SD}} \in \mathbb{C}^{N,N} \tag{3}$$

where $\boldsymbol{R}_{\text{SD}} \in \mathbb{C}^{N,N}$ express the spatial domain autocorrelation matrix. Additionally, $R_{\text{TD}}[m]$ defines the second order time domain autocorrelation at time difference m.

The spatial domain autocorrelation matrix depends on the scattering characteristics of the atmosphere. This means that depending on the behavior of the scattered waves, in general, the channel is spatially correlated. However, we assume the atmospheric effects introduces significant spreading on the propagating waves such that the fading is considered independent and identically distributed (i.i.d.), which yields

$$\boldsymbol{R}_{\text{SD}} = \boldsymbol{I} \tag{4}$$

where $\boldsymbol{I} \in \mathbb{C}^{N,N}$ is an identity matrix.

The time domain autocorrelation function follows the Clarks model

$$R_{\text{TD}}[m] = J_0(2\pi F_D T_s|m|) \tag{5}$$

where J_0 is the zero order Bessel function of the first kind.

For analytical tractability as given in [4][5], the Clarks autocorrelation function is approximated by the first order Auto Regressive (AR) process. Therefore, for i.i.d. Rayleigh elements

$$\boldsymbol{h}_{\text{NLOS}}[n] = \rho^{n-l}\boldsymbol{h}_{\text{NLOS}}[l] + \textstyle\sum_{k=0}^{n-l-1}\rho^k \boldsymbol{r}[n-k] \tag{6}$$

where $\boldsymbol{r} \in \mathbb{C}^{N,1}$ is zero mean i.i.d. Gaussian random process with variance σ_r^2. Furthermore, ρ yields the first order AR coefficient, which in this case

$$\rho = J_0(2\pi F_D T_s) \tag{7}$$

C. Tranmission System

The discrete-time baseband received signal at time n is expressed as

$$\boldsymbol{y}[n] = x[n]\boldsymbol{h}[n] + \boldsymbol{v}[n] \in \mathbb{C}^{N,1} \tag{8}$$

where $x \in \mathbb{C}^{1,1}$ is the transmitted symbol. Furthermore, $\boldsymbol{v}[n] \in \mathbb{C}^{N,1}$ is Additive White Gaussian Noise (AWGN) modelled as i.i.d. zero mean complex Gaussian process with variance σ_v^2.

The transmitted symbol sequence is transmitted on a frame-by-frame basis. Each frame constitutes P training symbols followed by D data symbols. Therefore, the symbol $x[n]$ is either a training or data, which rely on the value of time index n.

At the receiver, channel tracking is carried out by switching between Data Aided (DA) and (Decision Directed Mode). In the DA mode, the P training symbols are elaborated in order to achieve a satisfactory low MSE value. Then, the DD mode is executed based on the remaining D received symbols in the frame. During DD mode, the data symbols are detected and serve as reference signal for the adaptive channel estimator. The channel estimator is implemented with the RLS algorithm.

III. SIMO LS Channel Tracking Algorithm

The SIMO LS channel tracking algorithm is the special case of the MIMO one discussed in [4]. Although [4] focuses on Rayleigh channel, the algorithm can be implemented on Rician channel in a straightforward manner. This is because the LS approach is independent of the underlying channel model.

A. Algorithm

On SIMO system, the LS channel estimator minimizes the following cost function [4]

$$\textstyle\sum_{i=1}^{n}\gamma^{n-i}\|\boldsymbol{y}[i] - \boldsymbol{h}[n]x[i]\|^2 \tag{9}$$

where $0 < \gamma < 1$ is the forgetting factor.

The channel estimate is expressed as

$$\widehat{\boldsymbol{h}}[n] = (\textstyle\sum_{i=1}^{n}\gamma^{n-i}\boldsymbol{y}[i]x^*[i])(\sum_{i=1}^{n}\gamma^{n-i}x[i]x^*[i])^{-1} \tag{10}$$

The algorithm of the RLS SIMO channel estimation is given in the following steps [4]:

Initialization:

$\widehat{\boldsymbol{h}}[0] = \boldsymbol{0}$, $P[n] = \delta^{-1}$

For $n = 1, 2, 3, \dots,$ end, do

$$q[n] = P[n-1]x[n]$$
$$k[n] = \frac{1}{\gamma + x^*[n]P[n-1]x[n]}q[n]$$
$$\boldsymbol{e}[n] = \boldsymbol{y}[n] - \widehat{\boldsymbol{h}}[n-1]x[n]$$
$$\widehat{\boldsymbol{h}}[n] = \widehat{\boldsymbol{h}}[n-1] + \boldsymbol{e}[n]k(n)$$
$$P[n] = \frac{1}{\gamma}(P[n-1] - k[n]q^*[n])$$

The block diagram of the RLS SIMO channel estimation is depicted in Fig. 2. It is seen that the channel estimate $\widehat{\boldsymbol{h}}[n]$ is used for symbol detection during DD mode. The symbol detector is based on the LS approach and involves hard detection. For detection purpose, it is defined

$$x[n] = \begin{cases} x[n], & \text{DA mode}_t \\ \hat{\boldsymbol{h}}^H[n-1]\boldsymbol{y}[n], & \text{DD mode} \end{cases} \quad (11)$$

It is also evident that $\boldsymbol{y}[n]$ acts as the observation signal, whereas $x[n]$ works as the reference signal.

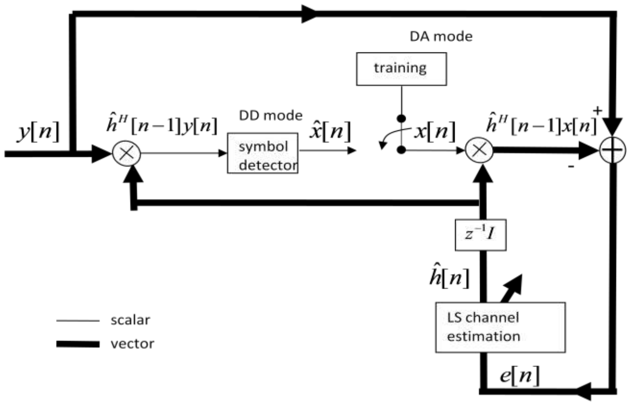

Fig. 2 Block diagram of RLS SIMO channel estimation

IV. System Simulation

The main goal of the system simulation is to obtain the BER performance of the RLS SIMO channel estimation in Rician fading. Since the RLS algorithm depends greatly on the assumed forgetting factor, we firstly determine the optimum forgetting factor γ_{opt} based on the analytical tracking MSE given in [5, eq. 14]. For simulation purpose, the application assumed is HST in Ka-band with frequency of operation 28 GHz, train speed 385 kmph, real time MPEG video streaming at 2 Mbps, and Quadrature Phase Shift Keying (QPSK) modulation. This results in $F_D T_s = 0.01$. The dependency between elevation angle ψ and K factor is taken from [2], which are $\{\psi, K\} = \{10^{\circ}, 0\text{ dB}\}, \{\psi, K\} = \{60^{\circ}, 10\text{ dB}\}$, and $\{\psi, K\} = \{90^{\circ}, 20\text{ dB}\}$.

A. Optimum Forgetting Factor and Tracking MSE

The BER is calculated based on the output of the symbol detector, which means only the DD mode is involved. Therefore, before the algorithm switches to DD mode, the tracking MSE should be acceptably low at $n = P$. From the observed MSE versus SNR, the value $n = P = 100$ seems a good choice. By using the first derivative of the analytical tracking MSE [5, eq. 14]. w.r.t. forgetting factor, the optimum forgetting factor versus SNR are obtained. The optimum forgetting factor curves and their respective tracking MSE are plotted in Fig. 3 and Fig. 4 respectively.

For all scenarios of elevation angle and K factor, it is found that γ_{opt} decreases with increasing SNR. Increament of SNR means that the current received observation is less noisy. Thus, γ_{opt} is reduced to put less weight on old received snapshots.

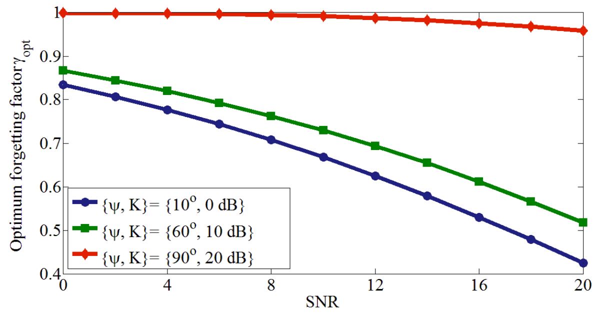

Fig. 3 Optimum forgetting factor, $M = 1$, $N = 5$, $F_D T_s = 0.01$, $n = P = 100$

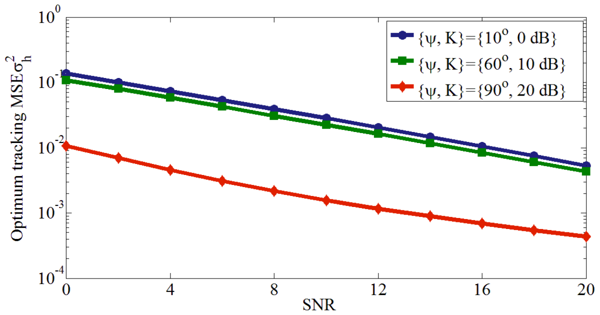

Fig. 4 Optimum tracking MSE, $M = 1$, $N = 5$, $F_D T_s = 0.01$, $n = P = 100$

It is also worth mentioning that lower elevation angle yields higher Doppler shift, consequently, higher time variation of the LOS component. This means that the old received snapshots becomes less correlated with current ones, which requires a reduced γ_{opt}. Moreover, in the $\{\psi, K\} = \{90^{\circ}, 20\text{ dB}\}$ scenario, the channel is practically constant where the disturbance mainly originates from AWGN. The value of γ_{opt} stays close to 1 since it is required to average the noise power from the received observations.

The behavior of the optimum tracking MSE curves, seen in Fig. 4, are similar to their respective optimum forgetting factor ones. However, it is important to note that the existence of Doppler shift increases the tracking MSE by one order of magnitude. This is justified by the two top-most curves as the tracking MSE for $\psi = 60^{\circ}$ is approximately equal to the case $\psi = 10^{\circ}$, although the Doppler shift of the former is 50% less.

B. Bit Error Rate

Lastly established in Fig. 5 is the BER performance of the LS SIMO channel estimation, where the chosen γ_{opt} are taken from Fig. 3. It is assumed that the frame length is $P + D = 500$. The simulation is averaged over 15000 independent channel realizations.

The BER of the constant channel scenario $\{\psi, K\} = \{90^{\circ}, 20\text{ dB}\}$ resembles that of AWGN, as this is also verified from our previous description concerning $\gamma_{\text{opt}} \approx 1$. The interesting view is for the remaining two scenarios, i.e. $\psi = 60^{\circ}$ and $\psi = 10^{\circ}$. At BER 10^{-4}, it is observed that the loss is approximately 7 dB, although it is previously found that the tracking MSE for those scenarios are indistinguishable.

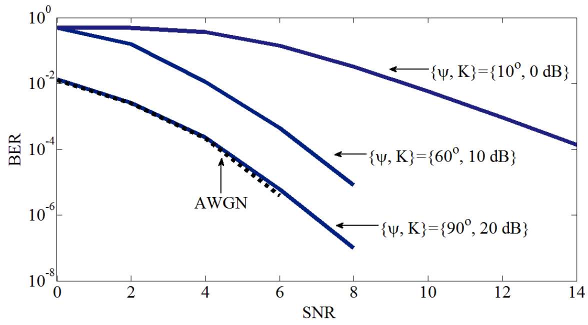

Fig. 5 Bit Error Rate, $M = 1$, $N = 5$, $F_D T_s = 0.01$, $P = 100$, $P + D = 500$

The reason is because the channel estimator is working in the DD mode, such that erroneous detected symbol propagates to the next one. For $\{\psi, K\} = \{10^{\circ}, 0\text{ dB}\}$, the channel exhibits higher time variation of the LOS component as well as higher fading fluctuations due to a lower K factor. By keeping the frame length fixed ($P + D = 500$ symbols), the RLS channel estimator is less robust to the erroneous detected symbols.

V. Conclusion

The system performance of the LS SIMO channel tracking in Rician fading is evaluated in terms of tracking Mean Square Error (MSE) and Bit Error Rate (BER). Across the wireless coverage, it is found that the tracking MSE is increased by one order of magnitude due to the elevation angle-dependent Doppler shift. The BER performance is as well position-dependent, but the frame length also influence the robustness of the channel estimation to propagation error.

References

[1] L. Liu, C. Tao, J. Qiu, H. Chen, L. Yu, W. Dong and Y. Yuan, "Position-Based Modeling for Wireless Channel on High-Speed Railway under a Viaduct at 2.35 GHz," *IEEE Journal on Selected Areas in Commun.*, vol. 30, no. 4, pp. 834-845, May 2012.

[2] Iskandar and S. Shimamoto, "Channel Characterization and Performance Evaluation of Mobile Communication Employing Stratospheric Platforms", *IEICE Transactions on Communications*, vol. E89-B, no. 3, pp. 937-944, March 2006.

[3] C. Komninakis, C. Fragouli, A. Sayed, and R. Wesel, "Multi-Input Multi-Output Fading Channel Tracking and Equalization Using Kalman Estimation", *IEEE Transactions on Signal Processing*, vol. 50, no. 5, pp. 10651076, May 2002.

[4] E. Karami, "Tracking Performance of Least Squares MIMO Channel Estimation Algorithm," *IEEE Transactions on Commun.*, vol. 55, no. 11, pp. 2201-2208, Nov. 2007.

[5] I. Zakia, S. Tjondronegoro, Iskandar, A. Kurniawan, "On the tracking performance of least squares MIMO channel estimation in rician fading", *IEICE Commun. Express*, vol. 3, no. 1, pp. 27-32, Jan. 2014.

[6] G. White, E. Falletti, Z. Xu, D. Borio, F. Sellone, Y. Zakharov, L. Presti, and F. Daneshgaran, *Adaptive beamforming algorithms for advanced antenna types for aerial platform and ground terminals*, *FP6 CAPANINA Project, Doc Ref: CAP-D17-WP3.3-UOY-PUB-01*, http://www.capanina.org/documents/CAP-D17-WP3.3-UOY-PUB-01.pdf, 2006.

[7] E. Falletti, F. Sellone, C. Spillard, and D. Grace, "A Transmit and Receive Multi-Antenna Channel Model and Simulator for Communications from High Altitude Platforms," *International Journal of Wireless Information Networks*, vol. 13, no. 1, pp. 59-75, Jan. 2006.

On the Design of FM Broadcasting Remote Monitoring System

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Abstract-In a country like Indonesia which still uphold eastern culture and religion norm, the content of FM broadcasting stations considered important for mentality development of the society. A committee on broadcasting in Indonesia has been established which has duty, among others, to manage complains on negative broadcasting content (actually not only FM broadcasting). In every province, there is similar committee. The local province committee responsibility is mostly covers a wide geographical area. Currently, the commissioners must visit the FM broadcasting studios in regents reside in a province to compile evidence when complains from society arise. In this paper, we propose a system to remotely monitor the content of FM broadcasting stations. By using this system, commissioners do not have to visit the stations complained by the society. They can monitor the broadcasting content from headquarter. Thus the proposed system offers cheaper solution than existing one.

Keyword: broadcast, FM, remote, monitoring, system

I. INTRODUCTION

In a country like Indonesia which still uphold eastern culture and religion norm, the content of FM broadcasting stations considered important for mentality development of the society. Negative contents are believed will make people behavior tend to also negative, especially for youngster who can be easily influenced by things that surrounded them.

In Indonesia, to manage FM broadcasting content, a committee on broadcasting (not only FM broadcasting actually) called Komite Penyiaran Indonesia (KPI) has been established with national coverage responsibility. In every province, there is similar committee. One of their duty is to manage complains towards the content of FM broadcasting stations. Reports followed-up by collecting evidences usually by visiting the FM broadcasting studios.

The local province committee responsibility is mostly covers a wide geographical area. So, visits will be time and budget consuming. In this paper we propose a system to remotely monitor the content of FM broadcasting stations. By using this system, commissioners do not have to visit the stations while collecting evidences. They can monitor the broadcasting content from headquarter.

The remainder of this paper is organized as follows. Section II briefly discusses related works. Section III presents system design. Section IV presents preliminary results. We conclude our paper and present the future works in Section V.

II. RELATED WORKS

There are small numbers of FM monitoring products in the market. Three of them are Audemat FM Monitor[1], Deva Broadcast [2] and davicom[3]. But all of them mainly only offers the quality of signal monitoring. None of them offers the monitoring of the content of the FM broadcasting stations.

The closest product to our proposed system is Audemat FM Monitor. It can only stream the FM broadcast audio, but doesn't store the broadcasting content. In our system, in addition to streaming, the audio can be stored locally to be played back remotely later.

III. SYSTEM DESIGN

In this section we describe system requirements and design. Before we proceed to system design, we begin our discussion with the system requirements. Some identified requirements are presented below.

- It can connect to the Internet
- It can be tuned remotely (via Internet) to any FM broadcasting frequency
- It can store the audio of monitored broadcasting station content to local storage
- The audio rate can be varied to suite storage capacity
- Stored audio can be downloaded via Internet
- The system must provide chunked recorded audio in order to divide a large audio files size into several smaller file sizes. Unreliable Internet connection can be a problem when downloading a large files, chunked audio files will make download more reliable
- Stored audio can be played back remotely via Internet-streaming
- In addition to stored-audio streaming, system must also provide live streaming

To satisfy the above requirements, we propose our system architecture as depicted in Figure 1.

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A programmable digital FM tuner, called *FM receiver* from now on, placed in each of monitored geographical area (the coverage size usually follows regents size, but not always). The connection between FM receivers to headquarter is via Internet.

When complain arise, the commissioner in headquarter send the command to the FM receiver to tune to the frequency of FM broadcasting station in discussion. The FM receiver will then change its tuning frequency accordingly. The broadcasting content will then be recorded to local storage. The recorded content can be downloaded or played back (via streaming) later. If preferred, real time audio streaming can be used by the commissioners to monitor broadcasting content. The stored audio content can be used as evidences for further legal action.

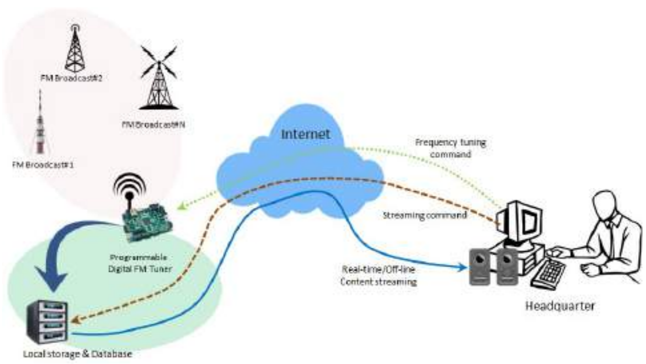

Figure 1 Proposed FM Broadcasting Remote Monitoring System Architecture

To further describe the system in details, Figure 2 show the more details on FM receiver built.

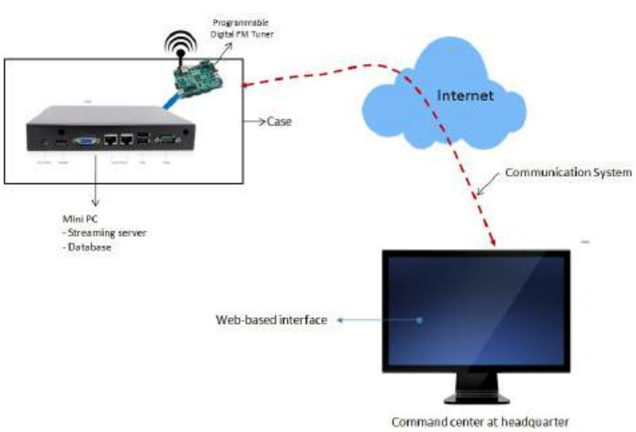

Figure 2 The system in more detail

The FM receiver built is designed to be compact but powerful enough to perform all related processes. A mini PC will be used as the heart of the FM receiver system.

The access from distance is made available by using http protocol via Internet connection between the FM receiver and the PC at headquarter. The Internet connection made available via Ethernet port at the mini PC. WiFi interface can also be provided. Considering the fact that there is possibility no local area network available, the mini PC will also be equipped with GSM modem by providing a GSM dongle (connected through USB port). Some, at least one, cellular operators offers 3G connection in almost all of Indonesia, so the connection is available for distance control via Internet.

The connection via cellular network is not reliable as fixed networks. Anticipating failed big file size download, we provide chunked recorded audio. Chunked recorded audio will be smaller in size so downloading them via cellular network will have lower probability to fail.

The size of chunked recorded audio files is adjusted by varying codec rate. Since we are only concern on the broadcasting content, not the quality, we can adjust the rate to the lowest available so that we can have the smallest file size but the conversation from the disk jockey (anchor) is still intelligible. The small file size also makes the successful download probability higher.

There is another thing we have to consider when we use the cellular network for Internet connection. The cellular network provider usually applied a private IP address for each Internet connection point. This will be problem for our server because it means that our server will be behind a NAT server. It can't be accessed directly from the Internet. To overcome this, there are some solutions available to be used.

To manage the stored audio files, we need to design a database, providing commissioner with the capability to select and download the required files.

To provide streaming service, we build a streaming server (based on open sources products available) in the mini PC. An http-based protocol is used to transfer the information between client at the headquarter and the mini PC.

A web interface will be provided also to enable commissioners performing some task such as tuning the FM receiver frequency, and streaming or downloading the audio.

For the FM receiver itself, we use a programmable FM receiver chip. An Arduino is used to program and control the chip. GPIO interface of Arduino, can be used to control several programmable FM receiver chip. So if we want to monitor more than one FM broadcasting station, we can do it by attaching more than one programmable FM receiver chip.

IV. Preliminary Results

In this section we present the preliminary results of our system implementation.

We have implemented the FM receiver using a programmable digital FM tuner from Sparkfun. It controlled by an Arduino. We managed to program and control the receiver locally. It means that we can control and manage the receiver via direct connection from a PC. We can change the tuning frequency to certain station available in our area. We have also implemented the audio recording function.

Figure 3 below shows the FM receiver which still implemented in a breadboard. You can see the Arduino to control the programmable digital FM tuner. Connection to the PC is using a USB cable connected to the USP port in the PC.

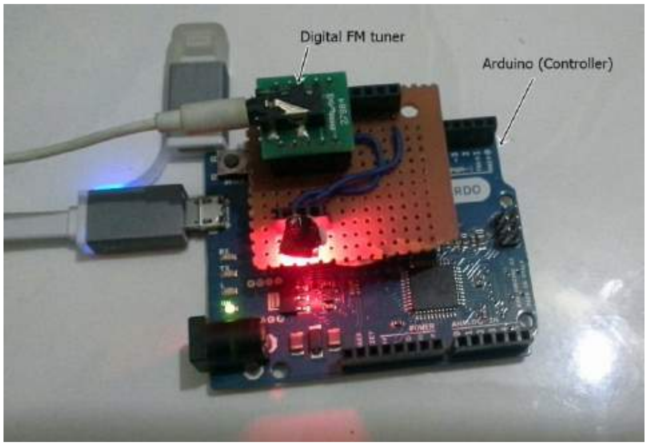

Figure 3 The FM receiver implementation

V. CONCLUSSION AND FUTURE WORKS

The design of an FM broadcasting remote monitoring system was presented in this paper. The system features are the followings:

- It can be managed via Internet
- It can be tuned remotely (via Internet) to any FM broadcasting frequency
- It can store the audio of monitored broadcasting station content to local storage
- The audio rate can be varied to suite storage capacity
- Stored audio can be downloaded and be played back remotely via Internet-streaming
- It provide live streaming

In the initial phase we have managed to implement locally controlled FM receiver and record the audio from the broadcasting station.

In the future we will implement a complete system which can be managed via Internet. The design of the case will also perform to make the FM receiver compact and easy to install.

REFERENCES

[1]http://www.audemat.com/radio-products-11-16-3.html
[2]http://www.devabroadcast.com/db4004
[3]http://www.davicom.com/client_file/upload/document/Davicom%20Low%20Power %20FM%20Site%20Monitoring%20Kit.pdf