# Experiment Evaluation and Analysis of Delay Handover in VoIP on Campus Network

Rendy Munadi, Asep Mulyana, Iman Hedi Santoso
School of Electric Engineering
Telkom University, Tel-U
Bandung, Indonesia
rendymunadi@telkomuniversity.ac.id, asepmulyana@telkomuniversity.ac.id
imanhedis@telkomuniversity.ac.id

***Abstract* - The majority of VoC user will be smart phone users. This means that VoWLAN will be dominant. Most of researchers agree that the general problems in VoWLAN are handover delay and VoIP QoS. So this study focused on two issues, handover delay and VoWLAN QoS. The experiment was initiated by determining the coverage area of two adjacent APs and their overlap area. Then we arranged a VoWLAN model by using Asterisk as VoC server, two APs, and two clients that consisted of a smart phone as a mobile user, and a laptop as a soft phone. Handover was done by making a phone call while moving (walking) through the overlap area to achieve handover process. All of processes were recorded using wireshark that produced data that will be analyzed. There were two handover scenarios that were investigated, i.e intra-subnet handover and inter-subnet handover. The result was handover delay of 118 msec (for intra-subnet), and 33 sec (for inter-subnet). While for the QoS parameter we obtained : 20.20 msec delay in average, 0.41 msec for mean jitter, 0.00% of packet loss, and 52 Kbps for throughput. From our experiment, we can conclude : the effect of handover delay for intra-subnet scenario, does not cause disconnection of the call session, it only suspends voice streaming during 118 msec (with zero packet loss). As handover process has been completed, call session can be continued. The different thing in inter-subnet handover scenario is that handover process causes disconnection of call session with delay 33 sec.**

***Keywords-handover delay; VoWLAN; VoIP; QoS; VoIP disconnection***

## I. INTRODUCTION

VoIP on Campus (VoC) is one of the basic needs of the academic society as a cheap means of communication to support the daily activities especially the academic processes. Wide spread use of smart phones, supported by availability of WLAN infrastructure with large amount of Access Points have spread across the campus area, then the smart phone users will become dominant in the VoC users. This means that VoWLAN is worth investigated. As stated in the abstract, this study will be focused on two cases, especially handover delay and VoWLAN QoS. This become interesting for investigation, because initially, 802.11 was intended for non real-time services like email, FTP, etc. That is why IEEE does not need to specify the mechanism of handover (layer 2). This situation brings the vendor (manufacturer of AP) to specify the handover mechanism independently. As a result, after the real-time services such as voice migrated into internet world, there is a problem. Another problem in VoWLAN is QoS due to complexity of radio environments such as multi-channel systems which operate in overlap mode, interference problem occurs occasionally. In addition, another problems such as obstacle, reflection, etc are the potential problems which are not easy to be cleaned.

### A. Power Link Budget

Power Link Budget in a radio communication system is the way to allocate the level of power along the radio path from the transmitter to receiver to ensure that the power level received by the receiver is in the range as needed. It is illustrated in Figure below [1].

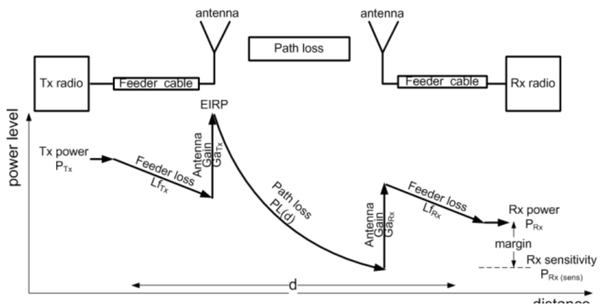


Figure 1. Power link budget in radio sistem

Power Link Budget can be expressed as follows:

$$P_{Rx}=P_{Tx}-Lf_{Tx}+Ga_{Tx}-PL(d)+Ga_{Rx}-FM \qquad (1)$$

where :

$P_{Rx}$ : Received power level (dBm)
$P_{Tx}$ : Transmitted power level (dBm)
$Lf_{Tx}$ : Feeder loss or cable loss (dB)
$Ga_{Tx}$ : Antenna gain of transmitting side (dBi)
*PL(d)* : Path loss (dB) for distance d (m)
$Ga_{Rx}$ : Antenna gain of receiving side (dBi)
*FM* : Fading Margin (margin to anticipate various loss due to various obstruction, reflection etc during movement)

<u>Note</u> : if PRx is replaced by PRxsens (sensitivity of the receiver), then PL(d) becomes the Maximum Allowable Path Loss (MAPL), and *d* is become the maximum value of distance or radius of coverage.

The main purpose of power link budget is to determine the MAPL based on transmitter and receiver specification that includes fading margin. After determining MAPL, the next step is to predict the coverage area (maximum distance between AP and STA) by using the propagation models.

### *B. Propagation Model [2]*

Propagation model is a mathematical model (equation) to predict along the path of communication. It is used to ensure, that the power level received by the receiver meets the requirement. The propagation model that will be discussed in this study is the model that fits the standard 802.11b/g. Moreover, the environment of radio propagation is limited for indoor situation which is line of sight. Such model presented as follows :

$$PL(d) = PL_{FS}(d_0) + 10n\log( \qquad (2)$$

$$PL_{FS}(d_0) = 20\log\left(\frac{4\pi}{\;}\right. \qquad (3)$$

where,

*PL(d)* : path loss at a distance *d* (dB).
$PL_{FS}(d_o)$: free space path loss at a close reference distance $d_0$ (dB)
$d_o$ : reference known distance (m)
*n* : path loss exponent typically 2<n<6

λ : electromagnetic wave length (m), λ= where : *c* is the speed of electromagnetic wave (m/sec) that approximate the speed of light , and *f* is the frequency used in communication, in Hz)

Let, $d_0$ = 1m, f= 2.4 GHz, c =3x$10^8$ ,

then,

$$\lambda = \frac{3(10)^8}{2.4(10)^9} = \frac{1}{8}\ m$$

By using equation (2), we get:

$$PL_{FS}(1m) = 20\log(32) = 40dB$$

Then equation (2) will be :

$$PL(d) = 40 + 10n\log \qquad (4)$$

### *C. Handover Process and Delay (Latency)*

Handover is the mechanism that allows users to move within a network or from a network to another one without losing connectivity [3,4]. When an STA moves away from current attached AP (Old AP), the STA will detect a decrease in RSSI received from Old AP. When RSSI reaches predefined threshold (Δ RSSI), then the AP will initiate handoff process as shown in Figure 2.

Handover process consists of three phases :

- *Scanning*
- *Authentication*
- *Re-association*

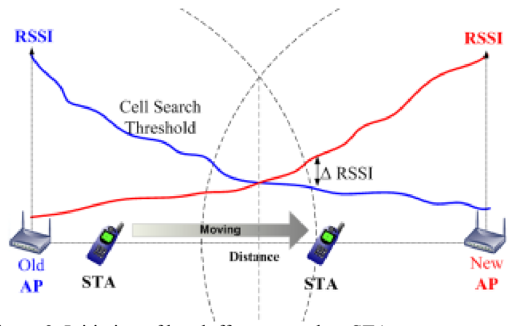


Figure 2. Initiation of handoff process when STA move away from current attached AP(old AP) [5]

The 802.11 standard does not define the handover process [6] but it does describe the basic processes as the re-association. Therefore the search method is vendor dependent. But in general there are two types of scanning modes : passive scanning, and active scanning. In passive scanning, an STA searches AP(s) that will be selected by listening to Beacon Frames that are regularly (every 100 ms) broadcasted by AP(s). In active scanning, an STA searches for AP(s) that will be selected by sending Probe Request frames and waiting for incoming Probe Response-s from AP(s). Figure 3 is a scanning mode followed by most of vendors.

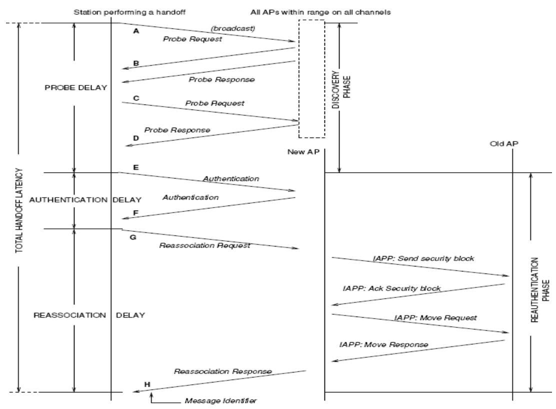


Figure 3. The IEEE 802.11 handoff procedure (followed by most cards) [7]

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Once the wireless device has determined which AP has the best respond to its selection criteria, it will go through the authentication process, which is the exchange of information between the AP and the STA, where each side proves the knowledge of a shared secret. When the STA is authenticated, it will start the re-association process, which is the exchange of information about the STAs and AP capabilities. Only after the re-association process is completed, an STA is capable to continue transmitting and receiving data frames.

*D. QoS Metrics for Voice Over IP [8]*

In VoIP services, a summary of the key QoS requirements and recommendations for VoIP are [9]:

- One-way delay (mouth-to-ear) should be no more than 150 ms (see Table I)
- One-way jitter should be targeted under 30 ms;
- Packet loss should be no more than 1%;
- Throughput should be guaranteed at 21–320 kbps per call (see Table II)

TABLE I. LIMITS FOR ONE-WAY TRANSMISSION TIME (ITU-T REC. G.114)

| One-way transmission time | Effect of Delay on Human Voice Perception |
|---|---|
| 0 to 150 ms | Acceptable for most users |
| 150 to 400 ms | Acceptable, but had impact |
| 400 ms and above | Unacceptable |

TABLE II. CODEC REQUIREMENT [5]

| Coding | Algorithm | Band-width (Kbps) |
|---|---|---|
| G.711 | PCM | 64 |
| G.723 | ACELP | 5.6 |
| G.723.1 | ACELP | 6.4 |
| G.726 | ADPCM | 32 |
| G.728 | LD-CELP | 16 |
| G.729 | CS-ACELP | 8 |

## II. DETERMINING COVERAGE AREA OF AP

Prior to investigating the handover process, the first step is to determine the coverage of two APs and their overlap region in order that they are able to do handover process. To achieve this situation, we use two approaches, i.e : prediction (calculation) approach (based on Power Link Budget and Propagation Model) to get maximum distance or radius of coverage, and the second approach is by performing measurement of signal strength (RSSI) at several point of location that has the same signal strength as the minimum receive power level or sensitivity of the receiver ($P_{Rx\ sensitive}$) = –74 dBm.

*A. Prediction Approach*

Based on technical specification of AP (Ubiquity-Unifi UAP) and STA (802.11a/b/g Radio Card)

- AP: Transmit power ($P_{Tx}$): 20 dBm, Feeder loss ($Lf_{Tx}$) 0 dB (assumption), Antenna gain ($Ga_{Tx}$) : 0 dBi (assumption)
- STA: Receive power sensitivity ($P_{Rxsens}$): -74 dBm [10], Feeder loss ($Lf_{Rx}$) 0 dB (assumption), Antenna gain ($Ga_{Tx}$) : 0 dBi (assumption)
- Fade Margin (FM) : 30 dB [1]

Based on equation (1), if $PR_x$ is replaced by $PR_{xsens}$, then PL(d) becomes MAPL, and *d* is maximum value or radius of coverage

$$PR_{xsens}=P_{Tx}-Lf_{Tx}+Ga_{Tx}-MAPL+Ga_{Rx}-FM$$

$$-74=20-0-0-MAPL+0-30$$

$$MAPL=84\ dB$$

Based on equation (4), if *d* is maximum distance (or radius of coverage), then *PL(d)* is MAPL. Let n=3 [2]

$$MAPL = 40 + 10(3)log(d_{max})$$

$$84 = 40 + 30log(d_{max})$$

$$d_{max} = log^{-1}\left(\frac{44}{30}\right)$$

$$d_{max} = 29.3\ m$$

*B. Measurement Approach*

In this approach, the measurement of signal strength (RSSI) is done by using a smart phone in which a WiFI analyzer application is installed. Each point of position that has the same signal strength (i.e equal to $PR_{x\ sensitivity}$ = −74 dBm) around the AP is marked as a point of border of coverage line. It is done repeatedly as much as needed to get illustration of the coverage line (see Figure 4).

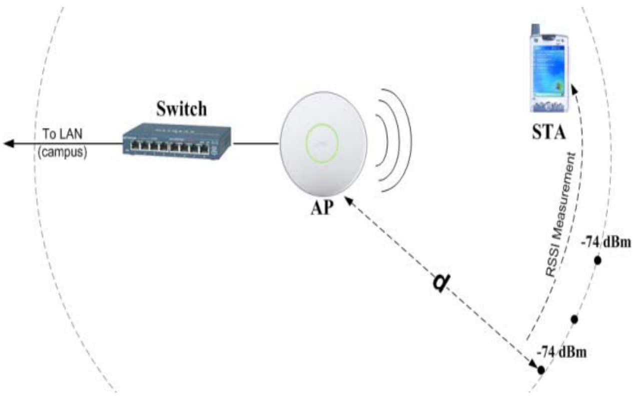


Figure 4. Measurement of $P_{Rx\ sensitive}$ to get coverage area of AP

The final result of measurement is the average of distance d which is 26,6 meter (note: compared to the distance based on prediction/calculation which is 29.3 meter). The same manner is performed for AP2 to achieve its coverage border. The position of AP2 is placed to achieve overlap area by 5 meter. The final result of arrangement of the coverage is shown in Figure 4.

## III. Experimental Set-Up

Basically, there are two configurations of test bed of handover delay that will be examined. They are : intra-subnet handover, and inter-subnet handover. Intra-subnet handover delay should be shorter than inter-subnet handover delay, due to only layer 2 that is involved. While in inter-subnet process, there are two layers that are involved, i.e layer 2 and layer 3.

### *A. Intra-Subnet Handover Configuration*

Intra-subnet handover configuration is shown in Figure 5:

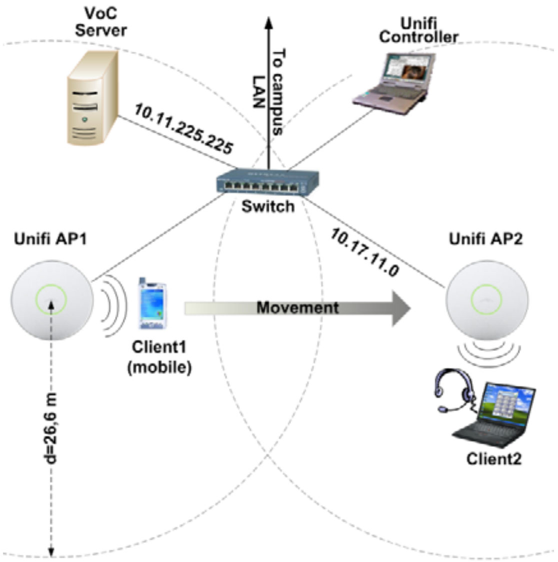


Figure 5. Intra-subnet handover configuration

UniFi Controller software is released by Ubiquity Unifi. Using this software the user can control all Unifi AP devices connected in a network, such as restart, set the strength of the signal, set the SSID and password, as well as simulated position via a web GUI interface as shown in Figure 6. Both AP are set with the alias name "switching_1" and "switching_2".The distance between AP-1 and AP-2 can be set based on the low and high position, which is between 40m to 120m. Each AP has already had an IP address and each client can set its presence in one of the AP.

### *B. Inter-Subnet Handover Configuration*

To investigate inter-subnet handover, we need a router to split the existing configuration into two different subnet as shown in Figure 7.

In the implementation of data collection handoff, the network uses a different network address, for example Ethernet-4 and Ethernet-5, in Figure 8, it can be seen in the address list that is generated. Handover process that is done by sending a ping from the client 1 via a router with a different subnet has worked out well. Changes that occur are the address of the subnet mask and default gateway as well as the emergence of an IPv4 address that previously did not exist in the process of intra-subnet handover.


Figure 6. GUI *web interface* of Unifi Controller

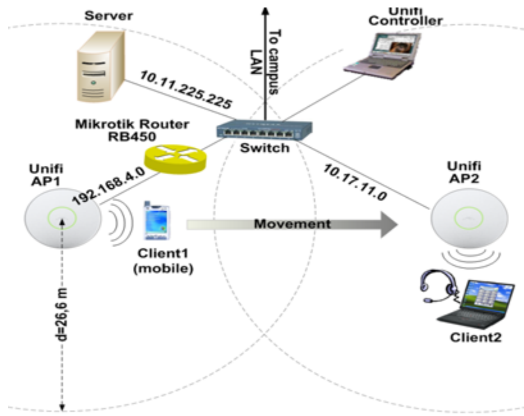


Figure 7. Inter-subnet handover configuration.

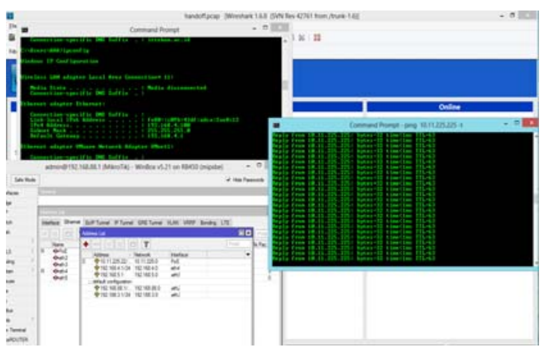

Figure 8. Capture result of handover inter-subnet

## IV. Experiment Results and Evaluation

The experiment was executed for delay handover with intra-subnet and inter-subnet. The other results are traffic display and performance table which are the measurement results done by wireshark on a smart phone.

### A. *Phase and Delay Handoff*

The packets were captured using wireshark during handover phase where the voice call was temporarily disconnected as users (smart phone and soft phone) did not hear anything for G.711 PCMU. The wireshark capture showed a 0% packet loss for this codec.

Before handover, the packets are send between 10.11.225.225 (source) and 10.17.11.32(destination). During handover process 10.11.225.225 keeps sending packets to 10.17.11.32 but does not receive any replay back as shown in Figure 9. In this, while ARP request is also broadcasted for the new IP address obtained 10.17.11.31. During this duration, user does not hear anything as packets are lost due to temporary disconnection of call [11].

When handover process is complete, 10.17.11.31 sends packets to 10.11.225.225 and than 10.11.225.225 replies back to 10.17.11.32 whose packets are routed to the new address as shown in figure above. Handover phase occurs at no.4643 with handoff delay 118.933 msec.

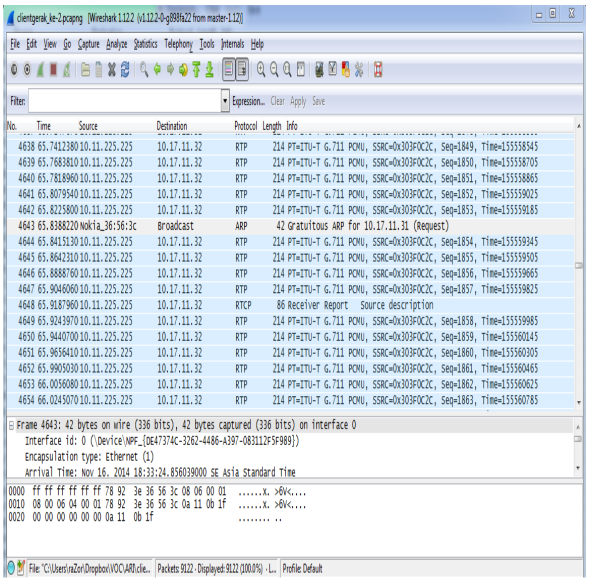


Figure 9. Screenshot of Wireshark during intra-subnet handover phase.

But for inter-subnet handover will be different, so more complicated and take longer time, depending on how tightly two subnet systems are coupled [11]. Before handover, the packets are send between 10.17.11.29 (source) and 10.11.225.225 (destination). During handover process 10.17.11.29 keeps sending packets to 10.11.225.225 but does not receive any replay back as shown in Figure 10. In this, while ARP, SSDP, DHCPv6 request is also broadcasted for the new IP address obtained 10.17.11.1?. During this duration, user does not hear anything as packets are lost due to temporary disconnection of call.

When handover process is complete 10.17.11.29 sends packets to 10.11.225.225 and than 10.11.225.225 replies back to 10.17.11.29. Handover phase occurs at no.12226-12261 with handoff delay 33.562348 second.

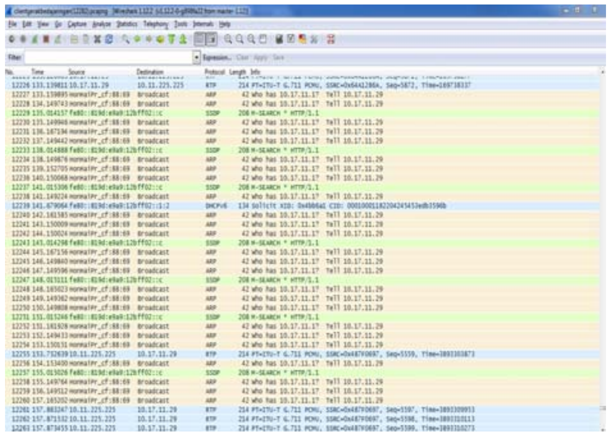

Figure 10.Screenshot of Wireshark during inter-subnet handover phase

### B. *Traffic Display and Performance Table*

The display traffic is an output of observation that is done by wireshark to show average packet per second in a system design as test handoff document. Average packet rate is 30.132 packets/sec and average bit rate is 0.052 Mbit/sec. This value in line with VoIP packet size by Codec G.711 with max.bit rate is 64 Kbps. Other parameters can be showed at Figure 11. From this evaluation, it can be concluded that the display traffic will depend on start monitor at the first packet and the last packet.

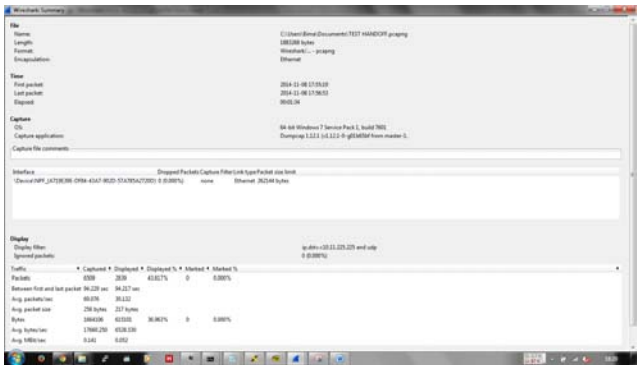

Figure 11. Traffic Display

Another observation that is done by using wireshark is a table of performance resulting from the stream analysis from a source port to the destination port, in Figure 12, it is conducted by 10.11.225.6 port 50 416 to 10.11.225.225 port 17338.The results that can be displayed is a table which gives the value of delay, jitter and bandwidth magnitude of

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each packet being monitored. Thus from the number of packets to be monitored, it will be obtained maximum delay, maximum and minimum jitter, so it can concluded that RTP stream analysis is needed to see the performance of a system that is built.

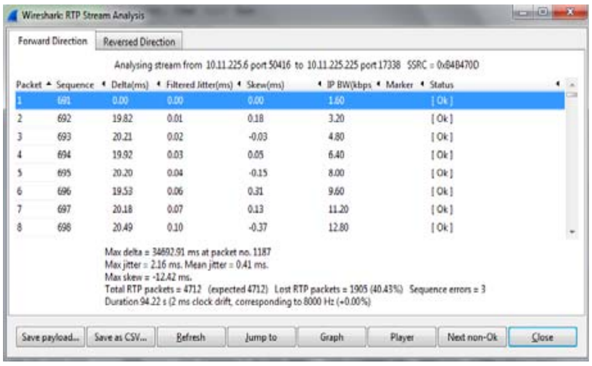


Figure 12. RTP Streaming analysis

## V. Conclusion and Future Work

This research is focused to evaluate handoff delay in a VoIP network on Campus from intra-subnet handoff and inter-subnet handoff. The results of some experiments and tests that have been carried out exhibit that the handoff intra-subnet is very fast is around 120 msec. The difference results is very noticeable occurs in case the inter-subnet handoff delay is around 33 seconds. This delay causes disconnection of the call. This is the highlight challenge for the future research : How to reduce the inter-subnet handover delay to avoid disconnection of a call.

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## Biographies

**Rendy Munadi** received his Doctor in Telecommunication Engineering from Indonesia University. He is senior lecturer of Telkom University Bandung-INDONESIA and he is presently as Head of the Expertise in Networks and Multimedia. He has served on the program committee of several conference and as reviewer of papers. He is current research in the area of Next Generation Network and New Generation Network, Wireless Network, Wireless Sensor Network, IMS, IP/MPLS Network, Routing Management and protocols/interface Next Generation Network.

**Asep Mulyana** received his Master Degree in Computer Engineering from Bandung Institute of Technology. He has experienced in the field on switching system (exchange), experienced as trainer on various technology of switching system, signalling system and telecommunication networks in the Training Centre PT. TELKOM Bandung. Currently he is a lecturer on Switching Technique, Telecommunication Networks, Traffic Engineering, Access Networks, Signalling System and Next Generation Networks (NGN). At this moment he is interest and concern to research on NGN and IMS.

**Iman Hedi Santoso** received his Master Degree in Telecommunication Engineering from the Bandung Institute of Technology in 2007. He has been involved with joint projects between Telkom University and Telecommunication Ministry of Indonesia, related to major issue in Indonesian telecommunication and maritime, those project are: Survey Analysis the Performance of Services in the Indonesian Mobile Operator, Analysis of the Impact of new generation network application in Indonesia, Maritime Call Sign Regulation in Indonesia, Operational Standard in Maritime Telecommunication in Indonesia, and Digital Maritime. He also participated in researches funding by Directorate General of Higher Education of Indonesia. His research and teaching interests include traffic engineering, switching technique, information and telecommunication network, computer network, and wireless network.