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THE ANALYSIS OF SIGNALLING PROCESS OF THE SERVICES IN INTEGRATED IMS

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ABSTRACT

This paper presented the analysis of communication process and determine the performance parameters of integrated IMS, i.e., jitter, max delta, and delay. An IMS core network testbed based on OpenIMSCore is developed, in which the clients have access through wired LAN and WiFi access points. The characteristics and performance of both access methods are compared and studied. The OpenIMSCore served clients well and produce communication processes in accordance with the SIP standards of RFC 3261. Based on the results of comparison testing using wired LAN and WiFi access points, it can be concluded that the QoS through wired LAN access fulfilled the QoS requirements and recommendation of standards set by the ITU-T, whereas the QoS through WiFi access does not meet the ITU-T standards.

Keywords: internet protocol multimedia subsystem (IMS), internet protocol (IP), session initiation protocol (SIP), OpenIMSCore, IMS core network, quality of service (QoS), jitter, max delta, delay.

INTRODUCTION

IP Multimedia Subsystem (IMS), which is defined first by the 3rd Generation Partnership Project (3GPP) Release 5 [ETSI-TS23.228_2002], and then updated by the 3GPP, 3GPP2 and ETSI TISPAN, is focused to provide a new network architecture enabling the convergene of fixed and mobile networks into single IP-based network infrastructure. The convergence also includes all applications and services (voice, data, video) within the networks [ETSI-TS23.228_2007].

In the development of mobile multimedia communication, IMS is a solution since providing various multimedia services such Push to Talk Over Celullar (POC), real time video sharing, video call, voice messaging, video conference, video on demand, IPTV, etc. Moreover, with the capability and functionality as a soft-switch IMS converges various platforms of telecommunication technology into single network which become the characteristic of Next Generation Network (NGN).

Several extensive works have been done in the area of IMS, particularly the use of IMS as the platform for femtocell network in the purpose of extending networks scalability and capacity of LTE-based mobile network [Ulvan_2009][Ulvan_2010]. The delay performance of IMS signaling in the UDP-based transport protocol has also been studied in [Pranoto_2013]. The development of a testbed of IMS system to provide a testing environment for the next generation wireless network (NGWN) has been conducted in [Ulvan_2013]. The concept of testbed is adopted into our work, therefore some performance tests of IMS can be conducted in this work.

The Session Initiation Protocol (SIP) is used as the signalling protocol in IMS network. The Call Session Control Function (CSCF) acts as the main SIP signalling server in the IMS architecture. The SIP has several functionalities i.e., registration, session establishment, and user control, including establishing, modifying and terminating service sessions from one or more users. In the mean time, the Wireless Fidelity (WiFi) access point, based on the standard IEEE802.11, is the most common wireless network used by users to get the connection into the internet, including accessing available network applications and services. WiFi access point is ussually used in the residentials, offices, and others indoor environment. In this paper, we propose the integration of WiFi access and |IMS core network, in order to provide user services in both end-to-end services and from the application server (AS).

This work investigates and analyse the communication process between the user equipment (UE) and the AS. The integrated test-bed of WiFi, local area network (LAN) and IMS networks is developed. The communication process to the AS through WiFi access is compared with the one through LAN. The performance of access networks in both WIFI and LAN, and the characteristic of the OpenIMSCore based IMS core network is measured and analysed in term of signalling delay and jitter.

This article consists of the concept, architeture, signalling and functionality of IMS, which is followed by the analysis of communication process in the form of SIP signalling flows. The description of the proposed integrated test-bed is also presented, as well as the measurement results and discussion. The work is finally concluded in last part of the paper.

IP MULTIMEDIA SUBSYSTEM (IMS)

IMS is a framework of internet protocol-based multimedia communication technology used for wired and wireless systems. It provides an extensible and scalable realtime networking with interactive multimedia applications and services. IMS is designed to support the high mobility users, which has not been limited by coverage area or particular domain. The principle of IMS network is to use the session to handle any requested service from each user.

IMS is able to improve the ability of packet switched mode (PS) of mobile networks to support IP-based services and applications via protocols Session Initiation Protocol (SIP). Effectively, IMS provides an unifying architecture that supports large coverage of IP-based services over packet networks and circuit switched (CS), utilizing the access technology differences of fixed and wireless networks.

In addition, IMS is designed to provide a number of key capabilities required to enable new IP services over mobile networks. The new field of IP services should consider the multimedia complexity, network restrictions, mobility settings and controlling capability for the emergence of many applications. IMS uses SIP protocol for multimedia session negotiation and management sessions. The availability of routing, network location, and addressing functions are included in IMS core network. Unlike the CS and PS domains, IMS domain allows any type of media session to be made. In addition, it also allows service providers to perform a combination services from CS and PS domains in the same session. This capability opens up a number of new and innovative services for user-to-user and multi-user services such as enhanced voice, video telephony, chat, push-to-talk and multimedia conferences, all of this is based on the concept of a multimedia session [Poikselka_2006] .

Architecture of IMS

IMS architecture is divided into 3 layers, the Application Server Layer (providing end user logic), Session Control Layer (comprising CSCF that control everything, from registration sessions to data communication), Transport and Endpoint Layer (to initiate and terminate SIP signalling).

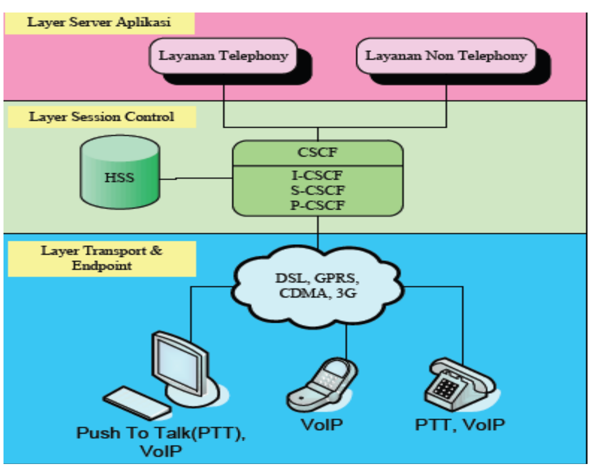

Figure-1. The architecture of IMS [ETSI-TS23.228_2007].

IMS is capable of tackling softswitch inefficiency by generating multiple services in a single session. The central role in this regard is the SIP protocol with 3 different servers: Serving-CSCF, Proxy-CSCF, and Interrogating-CSCF. Each CSCF have different tasks. In general, all types of CSCF have a role during session registration and session establishment. Figure-1 shows the architecture of the IMS [ETSI-TS23.228_2007].

IMS Signalling

SIP is used as signalling protocol in IMS environments. SIP protocol is defined in RFC 3261 [Rosenberg_2002] in [Ulvan, M_2010] which has function of registration, establishing and managing a session, and arranging the participants, including creating, modifying, and terminating sessions with one or more participants. SIP signaling is the main method used for user registration and session control in the IMS architecture. CSCF is the core signaling servers in the IMS architecture which work both as a SIP registrar and stateful SIP proxy server. Figure-2 describesthe signaling procedures in the IMS core.

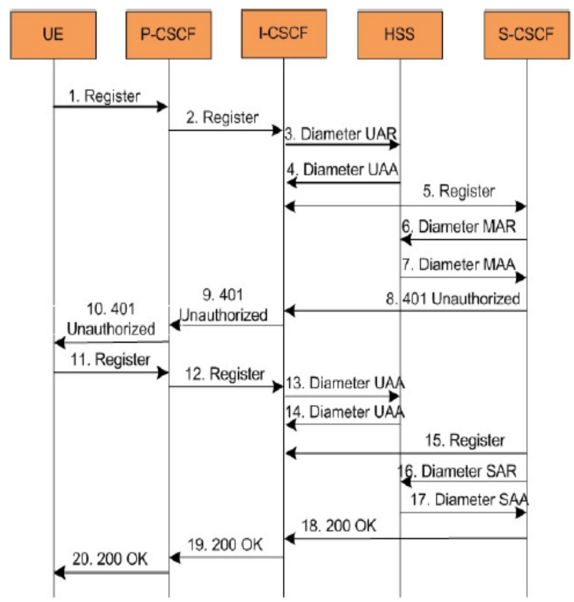

Figure-2. Signalling flow of registration messages [Ulvan, M._2010].

The signalling procedure begins with a SIP REGISTER of the user asks to be sent to the P-CSCF. Due to the limited bandwidth of the air interface, the message is compressed before it is sent by the user and decompressed at the P-CSCF. If several S-CSCFs are available in the users' home network, an I- CSCF needs to be deployed to select the S-CSCF serving the user session. In this case, the P-CSCF completing the addressing of user's home I-CSCF using the home domain and redirects REGISTER to I-CSCF [Ulvan_2009].

Upon sending User Authorization Request (UAR) to the Home Subscriber Server (HSS), which returns the address of S-CSCF, I-CSCF selects one of the

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S-CSCF and forwards the REGISTER message. S-CSCF sends a Multimedia Message Authentication Request (MAR) to HSS to download for user authentication. S-CSCF also stores a Uniform Resource Indicator (URI) in the HSS. The HSS then responds with Multimedia Message Authentication Answer (MAA).

S-CSCF forms a respond of SIP 401Unauthorised forming a response to a question that should be answered by the UE. Once the question is answered, then the authentication succeeded. S-CSCF sends a Server Assignment Request (SAR) to notify the HSS that the user has registered and HSS can download the user's profile. HSS responds with answers Server Assignment Answer (SAA). Finally, the S-CSCF sends 200 OK message to inform the EU registration process is successful.

Session initiation protocol

Session Initation Protocol (SIP) is an application-layer control protocol established, modify, and terminate multimedia sessions such as Internet phone calls (VoIP).

In general, SIP network consists of two basic components, namely SIP user agent and SIP network server. SIP user agent is a component of the client who initiated and answered calls. SIP architecture consists of several functional elements, such as: the User Agent, User Agent Server, Back to Back User Agent (B2BUA), Proxy Server, Registrar, Redirect Server [Poikselka_2006].

INTEGRATED IMS TESTBED AND MEASUREMENT SCENARIO

The testbed composes of three main systems i.e., the IMS core network based on OpenIMSCore system developed by FOKUS, the local area network system based on IEEE802.3 fast Ethernet LAN, and the IEEE802.11 wireless local area network system (WiFi). The testbed, in fact, is a modification of the integrated femtocell-IMS system developed previously by [Ulvan. A_2013]. The femtocell environment and devices has been replaced by WiFi system and devices. The topology of the testbed is shown in Figur-3.

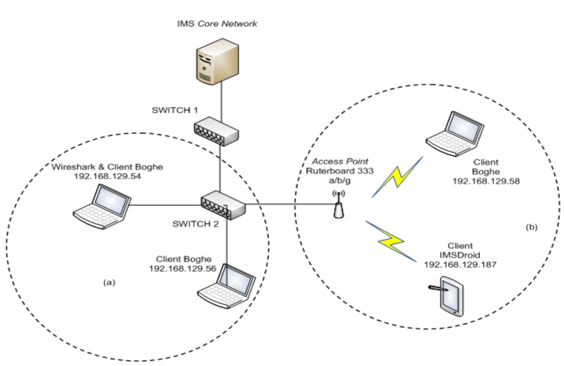

Figure-3. Testbed topology, (a) service access via LAN, (b) service via access point

IMS Core Network server and the WiFi access point is integrated into LAN through a network switch. LAN client (client 1) is connected to IMS directly from the switch, while the client 2 is connected to the IMS through WiFi access point. Client 1 used Boghe application as IMS'client installed in a personal computer. In addition to client 2, a smartphone with IMSdroid client application is used to get connection to IMS core network.

In this research, the Home Subscriber Server (HSS) is based on the FOKUS HSS (FHoSS) and the application server (AS) were installed in the same server engine as the OpenIMSCore system application. It is due to the small number of client took part in the research scenario, therefore, preparing the HSS and AS in the separate servers will waste the equipment resource.

Once the overall testbed is built, then the communication procedure and transactions are carried out. It is folllowing by the measurements on the communication process between UE and AS, and the measure of QoS of AS services such as voice call, video call, and instant messaging, in both via LAN cable and WiFi access point. Measurements were performed using Wireshark software, so that data packets generated during ongoing services can be recorded and analyzed.

RESULT AND DISCUSSION

Analysis of Signalling process

In this paper, the analysis of communication process between the IMS core network and its client were conducted in three different scenarios i.e., registration of client into IMS core network, voice/video call setup, and instant messaging.

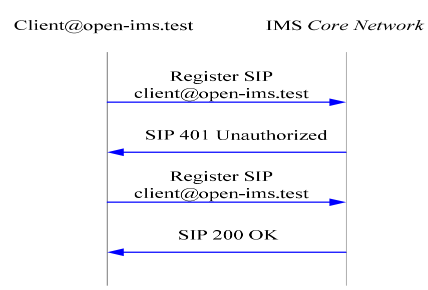

Figure-4. Signalling flow in registration process

Signalling flow of registration process, as shown in Figure-4, presents the generated communication session between the IMS's client and the IMS Core Network. The first message is "SIP REGISTER" which is sent by the client to the IMS Core Network. The IMS Core Network provides response SIP 401 Unauthorized, indicating that IMS Core Network requires the authentication from the client whether the client has been registered or not. As a response, the client sent the "SIP REGISTER" message back to the IMS Core Network, nd it informs that the client has been registered as a client of IMS Core Network. Then IMS Core Network provides response "SIP 200 OK" indicates that the client has

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successfully connected to the IMS Core Network and ready to use the available services.

Based on this analysis, it is shown that the nature of communication process between the client and IMS Core Network is request-and-respond mechanism, or also known as two ways handshaking.

Voice call and video call sessions

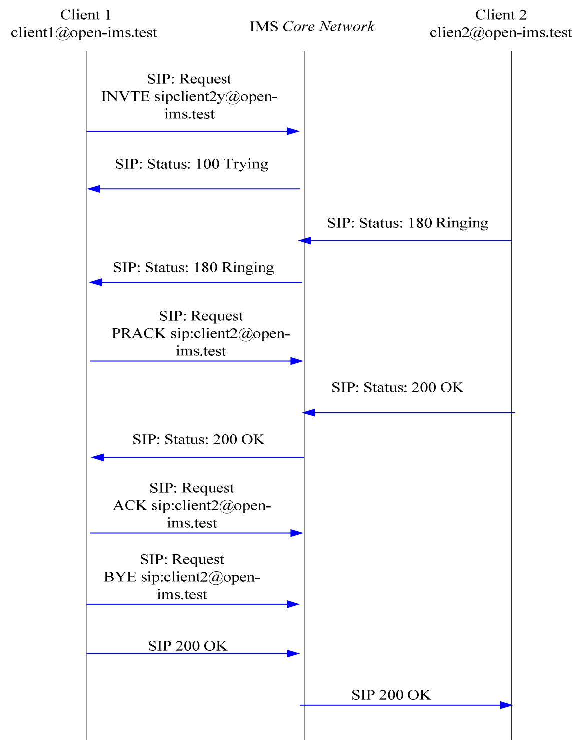

Figure-5. Signalling flow for voice and video call sessions.

Figure-5 shows the signalling process when two IMS'clients performed the voice call or video call sessions. The analysis were conducted from the beginning of the session call until the termination of call. The first line of signaling message code above indicates that the softphone client 1 wants to make a call to the counterpart client (client 2) by sending "SIP Request: INVITE" message. The IMS Core Network receives a SIP from the SIP softphone client 1 and provide a response to client 1 by sending the "SIP 100 TRYING" message. "SIP 100 TRYING" indicates that the "INVITE" has been received and IMS Core Network will work to continue "INVITE" to client 2. Once client 2 receive "INVITE" message, the IMS Core notifies client 2 that is a call from client 1. Upon receiving "SIP Request: INVITE", client 2 will provide the status of "SIP 180 RINGING" as a response which is sent to the IMS Core Network, the IMS Core Network then forwarding "SIP 180 RINGING" response to the client 2. SIP 180 RINGING response obtained by client 1 indicated the audio ringback tone on the softphone client 1. In this scenario, it is assumed that client 2 decided to accept the call from client 1. Upon receiving the ring back tone, the client 1 sent the "Request PRACK" to client 2. Client 2 provides the status of "SIP 200 OK" as response to the IMS Core Network, which is then forwarded the client 1. Response "200 OK" indicates that the call from the client 1 to client 2 has been received. Both clients that changing their data based on the agreed media protocol.

In case client 1 decided to terminate the call, client 1 will provide "BYE" message to the IMS Core Network, followed by "200 OK" message. The IMS Core Network forward both messages to the client 2, then the call session between the client 1 and client 2 is end.

Instant messaging sessions

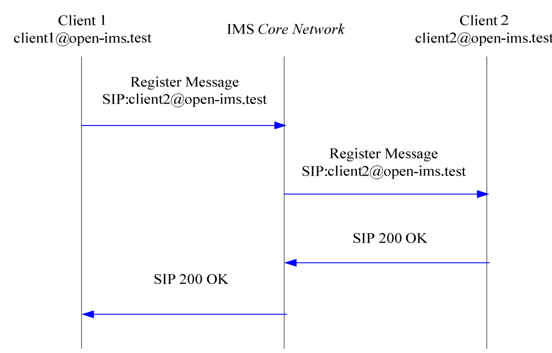

Figure-6. Signalling flow of messaging session

The figure of messaging sessions flow can be seen in Figure-6, in which all elements i.e., client 1, IMS Core Network, and client 2, get involved during the session. The first signal is "SIP REGISTER" message generated by client 1 as the party who sent the messaging content. This message is send the IMS Core Network, which is then forwarded to client 2. As the respond, the client 2 sents "SIP: 200 OK" message, indicates that the client 2 has received the message from the client 1.

Measured the QoS

Signalling delay in instant messaging

Figure-7 shows the duration of the packet stream generated by the SIP signaling on the instant messaging registration session by using a LAN cable and WiFi access point.

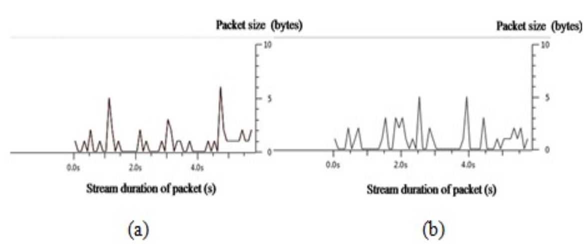

Figure-7. Signalling delay for instant messaging (a) LAN (b) WiFi access point.

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In the signalling session of SIP registration when using a LAN cable, the request for registration is started in particular time at 3.590 second with package size of 1 (835 bytes), the respond status of "401 Unauthorized" is generated in the second of 3.611 with package size 971 bytes. The re-registration request by client 1 is sending back to IMS Core Network at the second of 3,623, in which the generated packet size is 1119 bytes. The registration process was accomplished by receiving "200 OK" status at 6.349 with packet size of 467 bytes. The length of signaling time used from sending request register signalling until the user receives a message 200 OK is 2.759 seconds.

Comparatively, the signaling session for SIP registration by using WiFi access point started at some particular time of 3.073 second in which the package size is 836 bytes. The status of "401 Unauthorized" has been generated at the time of 3,109 second with package size of 972 byte. As usual, the system demands for sending the re-registration back at 3.112 second with generated packet size of 1120 bytes. The registration was accomplished when the "200 OK" is received by client 1 at 7.566 second.

In the case of instant messaging, the length of time for setup the service by WiFi connection is approximately 4.493 seconds, which is longer than wire connection by 2.759 seconds.

Voice call

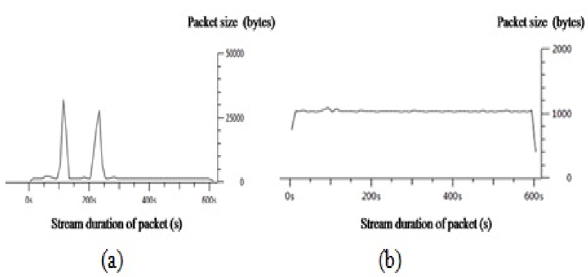

Figure-8. Signalling delay for voice call
(a) LAN (b) WIFI access point.

Figure 8 shows the duration of the voice calls packet stream generated by the SIP signalling at the Application Server. The the simulation tests conducted in two network scenarios, i.e., LAN cable and WiFi access point, with the calling time of about 10 minutes.

Based on the signaling procedure, the client 1 initated the call by sending the "Request INVITE" message to the called party (client 2) through the Mobile Core Network. After exchanging several signaling message the "200 OK" status confirmed that session establishment was conducted successfully. Next phase is transform the information (voice) package by using RTP protocol. Once the RTP protocol stopped, it means the conversation has ended, signalling SIP resumed with resultant demand BYE. The length of time used from client 1 to the counterpart for "INVITE request" up to 200 OK is 0.070 seconds. In addition the length of time used from BYE request up to 200 OK is 0.019.

In addition to WiFi access point connection, the similar procedure was occurred. The length of time used from client 1 to the called party form the "INVITE request" up to "200 OK" is 0.105 seconds. Moreover, the length of time used by "BYE request up to 200 OK is 0,018 seconds.

Video call

The video call services in IMS conceives the similar procedure of SIP signalling as voice call services. Obviously, the video call streaming has larger packet size compare to voice call. However, with the similar scenario, as the simulation tests taken for 10 minutes, it shows the similar characteristic of delay in both LAN and WiFi accesses, compare to voice call service, as shown in Figure-9.

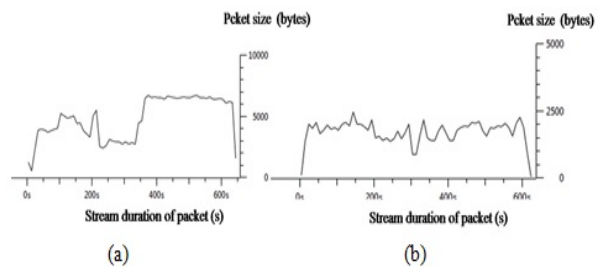

Figure-9. Signalling delay for video call service
(a) LAN (b) WiFi access point.

When the video call services were delivered in LAN cable, it just took 2.737 seconds to established the session, meanwhile it required 25.698 seconds to start exchanging the information through RTP protocol. The overall length of time to start the termination of service by sending "BYE" request until the party receive "200 OK" is 0.032 second.

On the other hand for WiFi access, the video call service session establishment took 8.917 seconds. However, it just required 12.781 seconds to start the RTP protocol. As we can see from the Figure-9, the nature of WiFi protocols are to reduce (or compress) the generated packed size, particularly for a real time packet streaming, therefore the overall size of packet streaming through WiFi are less than those in LAN cable. In a short time video call streaming, it might not influence the service quality, however in a large size packet, the service performance of LAN is far better than WiFi.

Jitter and max delta

The other performance parameters we concerned in this work is the jitter and the max delta. Both parameters were simulated and measured in the application server where the service are provided for the system. The jitter and max delta were measured in voice call and video call scenarios in both LAN and WiFi

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accesses. Figure-10 and Figure-11 show the measurement results of jitter and max delta in each scenarios and access medium.

As can be seen in Figure-10, the jitters and max deltas generated during a one minute voice call service. LAN cable produced 13.44 ms of maximum jitter, 43.32 ms of max delta, and 12.39 ms of average jitter. Meanwhile for WiFi access, the result for maximum jitter is 26.73 ms, while the max delta and the average jitter are 177.47 ms and 22.85 ms consecutively.

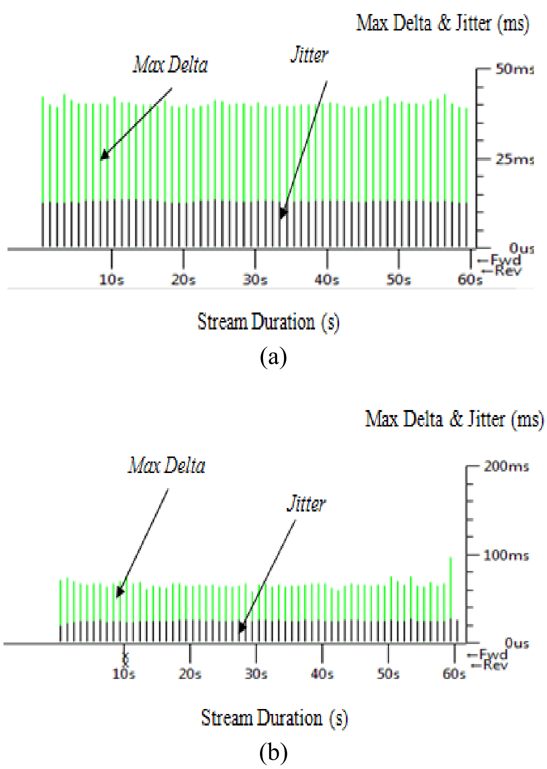

Figure-10. Jitter of voice call service at application server (a) LAN cable, (b) WiFi access point.

In case of video call, beside the access medium, two video codecs i.e., H.264 and G.729 are also considered on the test to analyse the characteristic of their performance in the system. Figure-11 shows the performance jitter and max delta generated in the application server for video call services. The test scenario were made for 60 seconds.

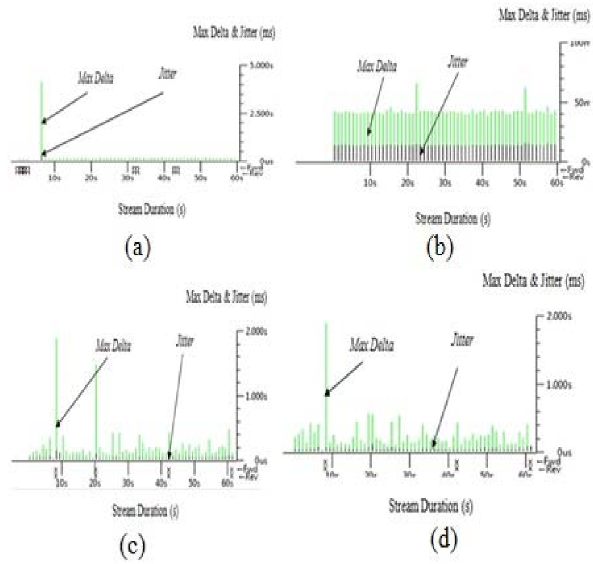

Figure-11. Jitter of video call at application server (a) LAN codec H264; (b) LAN codec G.729; (c) WiFi access point codec H264; (d) WiFi access point codec G.729.

Based on the test results, the maximum jitter for H.264 is 258.90 ms while the G.729 is just 15.24 ms. In addition, the performance of max delta in H.264 codec is 4098.10 ms, compare to the codec of G. 729 is 65.56 ms. Moreover, the average jitter of the system is 9.75 ms for H264 and the G.729 is 12.89 ms.

In case of WiFi access point, the maximum jitter generated in the application server for H.264 codec is 149.19 ms, while it is 110.45 ms for G.729. The max delta are 1876.01 ms 1884.07 ms for H.264 and G.729 respectively. Meanwhile, the average jitter is 31.53 ms and 25.57 ms for H.264 and G.729 respectively.

When compared jitter and max delta of voice and viseo calls in the application server it was confirmed that the WiFi access point generated the jitter and maxdelta greater than a LAN cable. It is due to the effect of outside interences of WiFi access point. In addition to generated jitter and max delta, the video codec of H.264 is larger compared G.729 voice codec. It is due to packet size sent by video codec is still larger than the biggest voice packet by voice codec. From Figure-11, it can also be seen that there are several points that have large max deltas compared to the others, this could be due to big packets passing the point. Packet size, obviously, affects the value of max delta as well by causing the long reception process.

CONCLUSIONS

In this work, the performance of signalling process of some services i.e., instant messaging, voice call, and video call of have been analysed in the integrated IMS Core Network using OpenIMSCore. Based on the obtained tests, the processes occurred from client terminals to the application server shown in Figure-4, Figure-5 and Figure-

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6 complied with the standard RFC 3261. It means, the developed IMS testbed is working properly as in standard. The performances of IMS have been measured concerning two parameters i.e., the jitter and the latency (max delta) of streamed packets, through two different access media i.e., LAN cable and WiFi access point. Based on the testing results for all scenarios, the LAN cable has better performance than WiFi access point.

Some improvement might be conducted to obtained a precise testing result. A dedicated application server separated from the IMS core network will provide a dedicated processing environment out of the high SIP traffics to and from the core network. In addition to the test bed, in fact it was working properly, but the real traffics environment need to be increased. The used of traffic generator application might be the solution.

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