The implementation of voice over internet protocol on private automatic branch exchange using analog telephone adapter

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Abstract. Voice over internet protocol is a technology that has offer to the telephone by Internet Protocol and has developed to be VoIP IP PBX which can be connected between VoIP and Private Automatic Branch eXchange, the use of technology VoIP IP PBX would be being easy more and emphasizing of the cost to the increasing extension telephone analog which is on the private automatic branch exchange without added Central Private Automatic Branch exchange. To implemented that idea, so we make server VoIP by using elastix as IP PBX server which is installed to the PC, and then using analog telephone adapter as signal conversion analog to the digital to be connected to the private automatic branch exchange. Measuring is done by calling between client as much as 12 times in 5 minutes. And the value of quality of service Jitter is 2,004 ms, Delay is 91,562 ms, and Packet Loss is 3%, so it can be categorized that quality of internet telephone is well done, it is based on ITU-T G.114.

1. Introduction

Several researches on VOIP (Voice Over Internet Protocol), among others is about and Integration of VoIP Technology Infrastructure and Smartphone (Android) and PABX on IPB Environment. It’s about how to make android application to connect VoIP network with PSTN by using VoIP gateway. And another research in the Universitas Jenderal Soedirman, the title is Feasibility Study Network at the Universitas Jenderal Soedirman for VoIP Implementation, it’s about how to replace PABX and PSTN telephone systems in each unit of UNSOED which has 140 phone numbers (extension and PSTN) [1].

In Politeknik Negeri Bandung has done a research about Design and Implementation of Internet Telephone on Telecommunication Laboratory Network of Bandung State Polytechnic. It’s about VoIP server by using asterisk then connected VoIP to PSTN using analogue telephone adapter as analogue and digital conversion tool.

Research position of Implementation of Voice over Internet Protocol on Private Automatic Branch eXchange using Analog Telephone Adapter, it uses elastix as a server and telephone analog as a tool to connect between digital signal which is on Voice over Internet Protocol system with private automatic branch exchange which has analog signals.

Developing of system telecommunication Internet Protocol (IP) by internet which is applied to the Local Area Network (LAN) which has one server Voice over Internet Protocol (VoIP) and two client, and then connected to the server Private Auto Branch eXchange (PABX) that has two extension
which has need Analogue Telephone Adapter (ATA) as a bridge of connection between signal analog and signal digital.

2. Literature review

VoIP (Voice over Internet Protocol) is a phone service that has internet base, so there is no pay. We can compare, economically, we do not need to build a new infrastructure of voice communication and application of wide data, that has smaller, than traditional phone [2][3]. As definition VoIP is a technology that can send a data that has a voice, video and data that has real time form by using internet protocol. So, this technology is how to change analog signal that has voice form to be format digital and translated to the IP packet and then transmitted to the internet [4].

PABX or Private Automatic Branch eXchange, appears to a central mini phone which is set in the office, school or buildings that has minimum traffic. PABX on an instance will be connected to another PABX or central Telecom by using incoming or outgoing traffic phone, each telephone instrument that has connected to the PABX has unique number extension that has been given by PABX. Each extension number could be called or call to the telephone by using operator, manually or automatically [5].

Analogue Telephone Adapter (ATA) or usually called by gateway or Internet Telephone Gateway (ITG) is a tool to connect between internet and analogue telephone to network VoIP [1]. A tool of ATA minimally has Foreign eXchange Subscriber (FXS) is interface (ports or plugs) to face with analogue telephone area [3].

3. Result

3.1. Topology of network VoIP and PABX

On the figure above, is a plan of system topology which is built. In this topology, the topology has three main components of the system [6]. there is a communication between client which is the local network for example number of phone 8003 calls phone number 7002, client of 8003 inserts number 7002, and then give information invite to the server elastix to the wireless to know a number of phone is busy or idle, after that server elastix continues signal to the Analogue Telephone Adapter (ATA), ATA which has function as converter digital sends to the analogue and then sends signal analogue by CO line which is in the server of PABX, CO line PABX will continue signal to the no of phone 7002, so there will be have a ringing to the phone no 7002. When section of communication is continuing, sound of sender will be receiver accepted by microphone, it will change sound of wave to be signal, and then it will continue by a tool, these signals will be continuing by wireless to the server, from the signal server to the receiver. After that those signals will be changed to be sound of wave by using speaker.
3.2. Examination call of VoIP and PABX

Figure 2. Call Process of client VoIP.

On the figure 2 is call process between client zoiper and dial number 8001 calls client x-lite with number dial 8002. (a) dialing process by client zoiper, (b) incoming call process of client zoiper, and (c) it is symbol of incoming call which is happened in the client x-lite when incoming call. On waiting this process client x-lite has signal incoming call from client zoiper which has been waiting, is it rejected or received. Choose telephone logos to receive of communication.

Figure 3. Client VoIP on the phone.

On the figure 3, it is about client VoIP on the phone is continuing. If communication process is going to be run well (client x-lite answered phone), so information which is on the softphone client zoiper status, (a) and client x-lite (b) changed to be same with the third figure above, information “on the phone” shows that client zoiper and client x-lite are doing communication.

Figure 4. Process of call VoIP to PABX.

On the figure 4, it is about process of call between client x-lite which has on digital signal to the client by dial member 7001 which is on analogue signal. Client x-lite presses 7001 then dialing
processed such as shown in figure (a) to the telephone, the phone ringing shown in figure (b) and figure (c) shown “call on CO 1”, it shows that there is an incoming call from port CO line 1. On waiting process, client PABX 7001 has signal “call on CO 1” that has been waiting for call is received or rejected. So, hold the microphone to get communication.

![Image](image.png)

**Figure 5.** Client VoIP on the phone.

On the figure 5, it is about communication process which is continuing between client x-lite and client private automatic branch exchange by number dial 7001. Information showed as figure (a) “on the phone” on x-lite, (b) show “CO 1 0:02:03” on telephone mentioned above is between client x-lite and client 7001 are continuing communication section.

On the figure 2, 3, 4 and 5 are tested of calling which is done between client as much as 12 times of call by duration 5 minutes and get the result value QoS which is get on table 1,2, and 3

**Table 1.** Measuring of Jitter.

| No | Client                | Jitter (ms) |
|----|-----------------------|-------------|
| 1  | Android - Telephone 2 | 2,40521     |
| 2  | Android - Telephone 1 | 2,023977    |
| 3  | Android - Laptop     | 2,343651    |
| 4  | Laptop -Telephone 2  | 0,681298    |
| 5  | Laptop - Telephone 1 | 0,65123     |
| 6  | Laptop - Android     | 0,595492    |
| 7  | Telephone 1 - Telephone 2 | 0,604842 |
| 8  | Telephone 1 - Laptop | 0,349307    |
| 9  | Telephone 1 - Android | 8,160777   |
| 10 | Telephone 2 - Laptop | 2,021751    |
| 11 | Telephone 2 - Telephone 1 | 3,389913 |
| 12 | Telephone 2 - android | 0,828146   |

| Average | 2,0046328 |

Jitter is a variation of delay between packet which happens on network IP, the bigger of jitter value is influenced by variation of traffic load and the bigger of accumulation between packet which is on network IP. On table 1 above can be known from all of total packet shipping on VoIP communication experiment, after 12-time experiment there was jitter with has average 2,004 ms, the highest value of jitter is 8,16 ms on the ninth experiment. According to ITU-T G.114 standardization, VoIP telephone quality is in the best quality because has 2,004 ms on the jitter average.
Table 2. Delay measurement.

| No | Client                  | Packet Captured | Respond (s) | Delay (s) | Delay (ms) |
|----|-------------------------|-----------------|-------------|-----------|------------|
| 1  | Android – Telephone 2   | 10073           | 764,18      | 0,075     | 75,86      |
| 2  | Android - Telephone 1   | 13174           | 1511,81     | 0,114     | 114,75     |
| 3  | Android – Laptop        | 75016           | 2010,37     | 0,026     | 26,79      |
| 4  | Laptop – Telephone 2    | 157510          | 3034,98     | 0,019     | 19,26      |
| 5  | Laptop - Telephone 1    | 203380          | 3581,51     | 0,017     | 17,60      |
| 6  | Laptop - Android        | 116707          | 2607,59     | 0,022     | 22,34      |
| 7  | Telephone 1 – Telephone 2| 235140        | 4156,05     | 0,019     | 19,65      |
| 8  | Telephone 1 – Laptop    | 273615          | 5447,12     | 0,019     | 19,90      |
| 9  | Telephone 1 – Android   | 259718          | 4708,32     | 0,018     | 18,12      |
| 10 | Telephone 2 – Laptop    | 778             | 560,25      | 0,720     | 720,11     |
| 11 | Telephone 2 – Telephone 1| 279093      | 4156,05     | 0,022     | 22,34      |
| 12 | Telephone 2 – android   | 299976          | 6646,71     | 0,022     | 22,15      |

According to the table 2, delay average is 91,562 ms, the impact of this delay is the sound which is received is less more 0,091 s. ITU-T standardization said that this quality of VoIP is still the best, because has delay value by average 91,562 ms, and ITU-T G.114 has been recommended that delay value is less 150 ms.

Table 3. Packet loss measurement.

| No  | Client                  | Packet Loss |
|-----|-------------------------|-------------|
| 1   | Android – Telephone 2   | 0%          |
| 2   | Android - Telephone 1   | 0%          |
| 3   | Android – Laptop        | 1%          |
| 4   | Laptop – Telephone 2    | 0%          |
| 5   | Laptop - Telephone 1    | 0%          |
| 6   | Laptop - Android        | 0%          |
| 7   | Telephone 1 – Telephone 2| 10%         |
| 8   | Telephone 1 – Laptop    | 10%         |
| 9   | Telephone 1 – Android   | 20%         |
| 10  | Telephone 2 – Laptop    | 0%          |
| 11  | Telephone 2 – Telephone 1| 0%          |
| 12  | Telephone 2 – android   | 0%          |

Table 3 is packet loss measurement by average is 3%. So, quality of internet telephone is the best quality, because TIPHON has been categorized it. Table 5,8 and 5,9 is standardization of packet loss ITU-T G.114.

3.3. Quality of service analysis
Ability of a network to serve more better to the specify data traffic by using QoS is not gotten by infrastructure, but it gets implementation to the network itself.
On the figure 6, it’s about measuring graphic of jitter with the same scenario. It showed that jitter of communication from the first telephone to the smartphone android, has long jitter, that is 8,16 ms. It’s because when it got examination, it had been done 20-25 m, so signal from access point router of android has only one and do not stabil. According to standardization of ITU-T G.114, quality of jitter for this VoIP is still in the best categorized, because has 2,004 ms average of jitter.

Figure 7 is a graphics by measuring delay. Process of measuring delay to VoIP communication and to the private automatic branch exchange uses analog telephone adapter that has delay average 91,562 ms. It shows that the highest delay is on the second telephone – laptop, it’s caused, when the second telephone is tested which has analog signal that must be converted to the digital signal which is on a laptop has error, so loading of laptop to receive of calling. Error on laptop, is caused of many of application on laptop. Quality of voice measuring on VoIP serve which is set on PABX is still on well done, it’s based on ITU-T G.114, that delay value is less by 150 ms.
Figure 8 is graphics of measuring packet loss. On the graphics above, the biggest of packet loss is on trying between the first telephone and android, that is 20%. It’s caused, when testing is done by spacing less more 20 – 25 m, so the signal from access point router on android is only one and unstable. Based on measuring, the average of packet loss is 3%. So, it can be categorized that quality of internet telephone is well done, it’s based on ITU-T G.114 that quality of internet by using packet loss 5% is still accepted.

4. Conclusion
On this research, it can be realized that VoIP of Private Automatic Branch eXchange (PABX) by using Analogue Telephone Adapter (ATA) as a tool to convert signal analog to the digital or digital signal to the analogue. And it has been done measuring of QoS to the communication session by using SIP (Session Initiation Protocol) for signaling, protocol RTP (Real Time Protocol) for transport and Codec audio G.711 so it can be concluded, that implementation of this experiment:

- VoIP by using server elastix can be implementated. So client zoiper on android and client x-lite on laptop can be run well of communication.
- It has been formed interconnection between Voice over Internet Protocol (VoIP) and Private Automatic Branch eXchange (PABX) so, between client VoIP and extension from PABX, is well communication.
- Experiment of VoIP is taken 12 times calling between client by communication session is less more 5 minutes, and duration of calling for quality of service is jitter 2,004 ms, delay 91,562 ms, and packet loss 3%.

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