VOICE CODEC QUALITY COMPARISON AND INTERCONNECTION TESTING BETWEEN ASTERISK SERVER AND PSTN CONNECTION

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Abstract

Nowadays information technology, especially the Internet developed very rapidly, which is actually a Internet computers connected to each other. Telephony technology is also developed very fast and there is some alternative to use VoIP beside analog telephone because the cost is cheaper. VoIP also use codec that can compress voice data but the quality is still good. This research design an open source system of Asterisk server because company need of VoIP that can support traditional analog telephony system. Beside design an open source system, some codec technology is also tested, which are G.711 as commonly codec and also G.729 and G.723.1 as propiteary codecs, offering less bandwidth and more clearly sound than G.711. G.729 and G.723.1 is limited for one user only so it can be tested only for one user. After codec testing is arranged then an interconnection system of PSTN or analog telephony system is also tested. Using Linksys SPA-3102 interconnection to analog telephony is also tested and worked for one client.

Keyword: VoIP, codecs, PSTN, asterisk, open source, interconnection

Abstrak

Saat ini teknologi informasi, terutama Internet berkembang sangat pesat, sehingga ada teknologi jaringan internet yang saling menghubungkan komputer tersebut. Teknologi telephony juga berkembang sangat cepat dan ada beberapa alternatif untuk menggunakan VoIP disamping telepon analog karena biayanya lebih murah. VoIP menggunakan codec yang bisa mengkompresi data suara namun kualitasnya tetap bagus. Penelitian ini merancang sistem open source server Asterisk karena perusahaan membutuhkan VoIP yang dapat mendukung sistem telepon analog. Selain merancang sistem open source, beberapa teknologi codec juga diuji, yaitu G.711 sebagai codec yang berlaku umum dan juga G.729 dan G.723.1 sebagai codec propiteary, yang menawarkan bandwidth lebih sedikit dengan suara yang lebih jelas daripada G.711. G.729 dan G.723.1 terbatas hanya untuk satu pengguna sehingga hanya bisa diuji untuk satu pengguna saja. Setelah pengujian codec dilakukan maka sistem interkoneksi PSTN atau sistem telepon analog juga diuji. Interkoneksi dilakukan dengan voice gateway Linksys SPA-3102 dihubungkan ke telepon analog juga diuji dan dilakukan untuk satu klien.

Kata kunci: VoIP, codecs, PSTN, asterisk, open source, interkoneksi

1. Introduction

Nowaday the information is transmitted through a computer network is not only the shape of the data but can also be in the form of voice and video. Voice Over Internet Protocol (VoIP) can be used as variation with existing equipment and also existing technology such as PSTN system as old analog telephony system.

Telephone devices connected to the IP network are commonly referred as IP telephony. VoIP technology someday should be able to replace the analog telephone network Public Switching Telephone Network (PSTN) because of flexibility and also have a lower price.

However, because the PSTN is still widely used in some company, so this research implement VoIP system that have interconnection with the existing PSTN. Voice gateway device and the IP PBX server can provide an alternative solution to interconnection problem. PSTN device can be call from VoIP client and vice versa, so that two-way communication can be occured.

VoIP's cost will be cheaper for long-distance, because only use the intranet bandwidth, if it used locally and also lower bandwith when used externally. VoIP use less bandwidth than a regular phone, because the development of voice codecs technology. Bandwidth for voice data can be reduced to 8 kbps only, using G729 or G723.1 codecs, when standard codecs must use overall 64 kbps bandwidth.

VoIP server or IP PBX server that where used in this reseach is Asterisk server. Asterisk server is easy to modified, so the development of codec testing and also interconnection testing can be flexible. Asterisk is also play a role as IP PBX server and can connected to Asterisk PSTN network using internal company network.

2. Methodology

2.1. VoIP

Voice over Internet Protocol (VoIP) is a technology that capable to passing voice, video, and data in the form of packets over an IP network. The interesting thing about VoIP is that, the use of VoIP does not require additional infrastructure, so that could be saving expense from developing new infrastructure.

VoIP transmits voice over Internet infrastructure standards, using the IP protocol. That is why it can be done at communication without having to pay more, in addition to the price of the Internet connection.

2.2. Advantage and disadvantage using VoIP

The advantage of using VoIP are:

1. Utilizing data network infrastructure that already exists for the sound.
2. Use less bandwidth than a regular phone. Data compression technique enables voice only need about 8 kbps of bandwidth to be able to at least hear the speech.

   The disadvantage of using VoIP are:

1. The sound quality is not as clear as PSTN. This is the effect of sound compression using a small bandwidth.
2. There is a delay in communication caused by process of changing data into sound that need some time and there is also a network processing delay.
3. If using the Internet and computers behind the NAT (Network Address Translation), then a special configuration is required to make the VoIP running well.
4. There is some effort to make adjustment to new telephony numbering system because there is an existing numbering system.

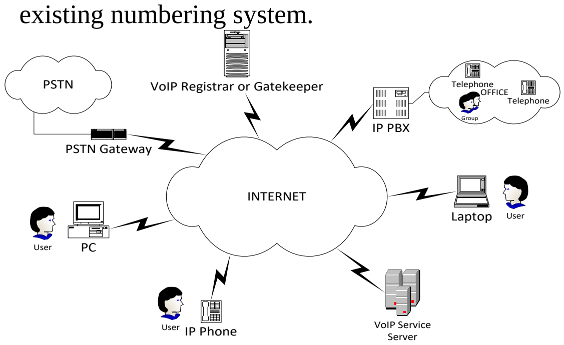

Figure 1. VoIP Network [5]

VoIP is generally divided into 2 parts, the control signaling and voice data:

1. VoIP control traffic of voice data. VoIP also keep the entire operation of the network (router to router communications), which also known as Packet Signalling [5].
2. Voice data is the form of user traffic information delivered end-to-end, also known as Voice Packet.

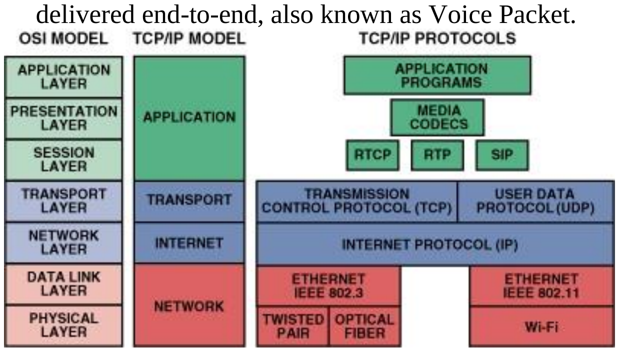

Figure 2. ***Open Systems Interconnection*** **and TCP/IP model of VoIP**

2.3. Session Initiation Protocol (SIP)

Session Initiation Protocol (SIP) is a standard protocol multimedia which producted by Internet Engineering Task Force (IETF) and has been used as a standard VoIP usage. SIP is a protocol that exist in application layer. SIP used to define the beginning of the process, change, and termination of a multimedia communication session. Request and response to the request is called a SIP transaction. In VoIP system, there are five components in the SIP system [5]:

1. User Agent

   User Agent is the system used to berkomuikasi, in which the user agent has two parts, namely:

   a. User Agent Client (UAC)

      UAC is designed application on the client to initiate SIP requests.

   b. User Agent Server (UAS)

      UAS is a server application that notifies the user when receiving the request and also provide a response to the request. The response can be either accept or reject the request that was sent.

2. Network Server

   User joined on the network can initiate a SIP call and also can be called after register known location firstly. Registrar server never forwards the request. Registration can be done by sending a REGISTER message to the SIP server. In the network there are two types of SIP network servers:

   a. Proxy Server

      Requests can be served alone or delivered (forward) to proxy server. Proxy server has the task to translating and or to rewrite a request message, before passing the user agent or proxy. Proxy servers store the entire state communication session between the UAC and UAS.

b. Redirect Server
   Redirect server is a component that receives a request message from the user agent, and generate 3xx responses to requests it receives, directing the client to contact an altenate set of URIs. Redirect server does not store the session state of communication between the UAC and the UAS [11].

3. Registrar Server
   Component that receives a request message REGISTER. Registrar can add a user authentication function for validation. Registrar store the user database for authentication and actual location (such as IP and port) [11].

2.4. List of SIP request method

- REGISTER: used by a UA to indicate its IP address and the URL to which you want to accept the call.
- INVITE: ysed to establish a media session between agent users.
- ACK: confirm message reliably.
- CANCEL: terminate the pending request.
- BYE: stop the session between two users in a conference.
- OPTIONS: requests for information about the ability of the caller, without setting call [6].

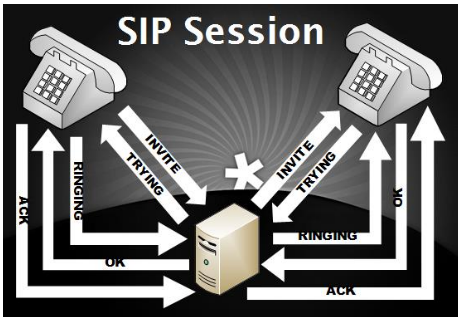

Figure 3. SIP Session

2.5. VoIP Data Protocol
2.5.1. RTP

RTP was developed by a group of audio and video transport from the IETF. RTP is used to connect with other protocols such as H.323 and RTSP. Standard RTP protocol uses a pair of RTP and RTCP. RTP is used to transmit multimedia data and RTCP is used periodically to transmit control information and QoS parameters [7].

RTP is a protocol designed to reserve part of the bandwidth available for UDP traffic. RTP tolerate jitter and desequencing that occur on IP networks. RTP was not developed only for voice data traffic, but also be used for video data traffic because it is maintaining or supporting the bandwidth that will be used by the UDP traffic. RTP header frame contains information to identify and manage each individual call from endpoint to endpoint. This information will be specified into two sub-protocols, namely RTP and RTCP.

- Data transfer protocol, RTP, which is associated with the transfer of data in real-time. The sequence numbers are used for sorting the data packets and detect any lost packets, and payload format that indicates the format of the data that has been encoded.
- RTCP protocol is used to determine the feedback of quality of service (QoS) and synchronization between multimedia flows. Bandwidth of RTCP compared to RTP is about 5% smaller.
- Signalling protocols available is H.323, MGCP, and SIP
- Optional Protocol for exchanging media such as the Session Description Protocol (SDP).

RTP sessions constructed for each multimedia stream. Session consists of an IP address with a pair of RTP and RTCP port. For example, the audio and video streams have separate RTP sessions, the receiver is enabled to cancel a particular streams [7].

Ports that used in session initiation is negotiated using other protocols such as RTSP (using SDP in its setup method) and SIP. According to the specification, RTP port will be even and the RTCP port is an odd number higher. RTP and RTCP using unprivileged UDP ports between 1024-65535, but also can use other ports, depending design of the protocols [11].

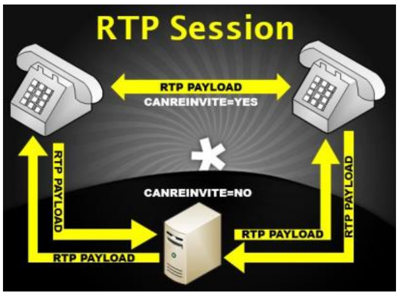

Figure 4. RTP Session

2.6. Architecture System of VoIP

Asterisk VoIP communication using VoIP gateways and also the architecture can be divided into four layers:

- Infrastructure layer, is a layer which carries data between all network devices and applications and consists of a voice gateway that serves as a swith and routers.
- Call-processing layer, is a physical layer of the infrastructure that provides call control. Provided by Asterisk PBX itself as an IP PBX that provides diversion and call processing.

- Application layer is a logical layer that is freely provide the call processing functions and voice processing. At this layer calls inialization occured and also the session during the call.
- Client layer is a layer that serve voice applications for the user. The devices in this layer is a device [10].

2.6.1. VoIP Gateway

VoIP gateway used to process the voice coming from the analog telephony, then converted into a digital signal. This digital signal is processed by digital devices such as IP phones. IP phones can do the dialing and also look for an existing extension number [1].
VoIP gateway will be the first device that dial PSTN from the outside. PSTN device has its own dial number and will be recognized by the VoIP Gateway. VoIP gateway then give a second dial tone in order to access an existing extension number to IP PBX server. IP PBX then will give an access to a number of existing VoIP extension so the state become in-dialing [2].
VoIP gateway also can forward original phone number from the PSTN Gateway into a digital number of the VoIP gateway. VoIP gateway phone number can be dialed from PSTN phone with extension number [9].
VoIP gateway connect two different telephony system, which support connection from FXO or FXS analog phone and digital phone which connected to the internal network or Internet network.
VoIP gateway have an advantage to support routing network instantly without having to connect to an external router to be accessed from outside. There is also a web-based GUI interface that can be accessed from a web browser using the IP.

2.6.2. Asterisk

Asterisk is a hybrid PBX software which support TDM and packet-voice that has an IVR and ACD platform with open source code. Asterisk use GPL and non-GPL license and written by C [3].
Asterisk support popular codecs, such as ADPCM, G.711 (A-law, μ-law), G.722, G.723.1, G.726, GSM, iLBC, and also LPC-10. Asterisk is able to serve users who communicate with different codecs because have availability to many codecs [3][4].

2.6.3. Codec

ITU-T have a standard for voice codec (coder / decoder) which are available for VoIP implementations. Voice codec commonly known are: G.711, G.723, G.726, G.728, and G.729. Here's a brief overview of each type of codecs above:

a. G.711 is an International standard for audio compression technique using Pulse Code Modulation (PCM) in the delivery of voice. This codec used in traditional TDM voice T1. [8]
b. G.723.1 is voice coding which recommended for multimedia terminals with low bit rate.
c. G.726 is an ADPCM voice coding techniques with multiple bit rates vary, which is 40 kbps, 32 kbps, 24 kbps, and 16 kbps. This codec is suitable for interconnection to the PBX with a bit rate of 32 kbps.
d. G.728 is codec with good sound quality and specifically designed for low-latency applications. This codec compress the voice into a 16 kbps stream.
e. G.729 specifically designed for low-latency applications. This codec compress the sampling signal 16 bits at 8 kHz via 10 ms frame, become a standard bit rate of 8 kbps.
f. G.729a have an algorithm that is simpler and requires less processing power than G.729.

MOS (Mean Opinion Score) is an assessment of the quality of the voice call value between 1 (very poor) to 5 (excellent). Good MOS value for VoIP is between 3.5 - 4.2. The total bandwidth used by each codec is as follows.

Table 1 Comparisson of Codec [3]

Compresion Method		Bit Rate(kbps)	Sample Size(ms)	MOS Score
G.711	PCM	64	0.125	4.1
G.726	ADPCM	32	0.125	3.85
G.728	LD-CELP	15	0.625	3.61
G.729	CS-ACELP	8	10	3.92
G.729a	CS-ACELP	8	10	3.7
G.723.1	MP-MLQ	6.3	30	3.9
G.723.1	ACELP	5.3	30	3.65

a. G.711, G.723, G.729

Coding algorithm		Bandwidth	Sample	IP bandwidth
G.711	PCM	64kbps	0.125ms	80kbps
G.723.1	ACELP	5.6kbps	30ms	16.27kbps
	MP-MLQ	6.4kbps		17.07kbps
G.726	ADPCM	32kbps	0.125ms	48kbps
G.728	LD-CELP	16kbps	0.625ms	32kbps
G.729(A)	CS-ACELP	8kbps	10ms	24kbps

Figure 5 Total Bandwith G.711, G.723, G.729

b. GSM

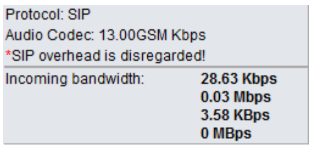

Figure 6 Total GSM bandwidth

3. Experiment and Analysis

3.1. Network Topology

VoIP gateway have interconnection with the PSTN line connected to the PBX company network. There is a hub that connected to Internet network. There are two firewall for internal security and also firewall connected with outside network directly. There is NAT process to access Internet from internal network of the company because there are two layer firewall.

3.2. Testing Quality of VoIP
3.2.1. Testing of packet data flow

Packet data flow is obtained by doing sniffing at two side. One side is from the user who requests the call and one side is from user who receive the call. Packet data flow that has been sniffed can detail the registration process and also the discussion session. Packet sniffer tools like wireshark can sniff data information of each session, formed in accordance with the protocol functionality.

This is the result of sniffing performed at the user with IP address 10.10.57.2 call the user with IP address 10.10.57.3. This session resulting in status that occurs when requesting a conversation and also sequence of handshaking process when call is formed. Codec have been set to disallow all and allow µlaw. Sniffing tools also provide a feature to see the flow of the data flow as in a Figure 9.

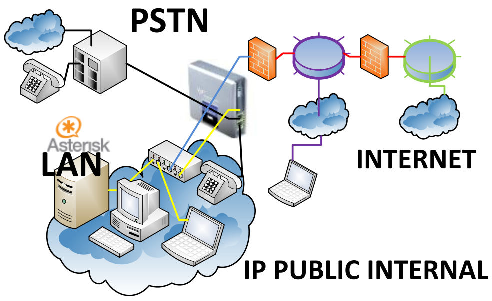

Figure 7. Network Topology

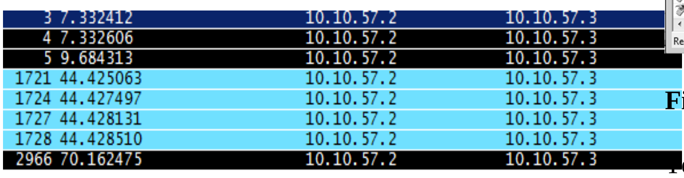

Figure 8. Sniffing request process

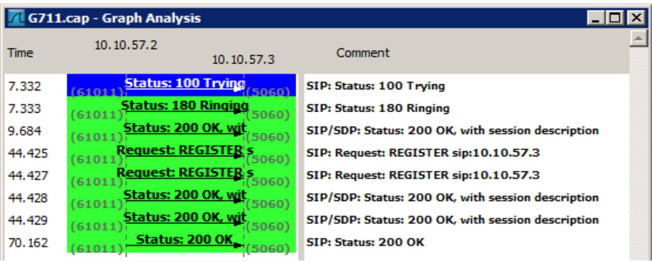

Figure 9. Flow data analysis from user with IP 10.10.57.2 to the server IP 10.10.57.3

Based on the status of the request obtained by the flow of a call, status begins with 100 which is SIP trying, after that followed by status 180 which is SIP ringing and if the destination number is available there would be a status 200, which means the request is received. When the communication session is formed there will be status Registered.

RTP packet stream could recognize the types of payload if the capture process happened before dialing calls. If the call session has been established, the conversation at the moment only captured UDP packets, which are raw data, so could not be analyzed what type of package that arrive and sended.

3.2.2. Testing and analysis of the bandwidth requirements in wide variety of codecs

The test has been done to gained real bandwidth respectively from each codec is done in a span of one minute. Monitoring tools that used are NB bandwidth monitor that can monitor the application level. VoIP uses UDP connection to send the RTP packets. Calculation only considered bit rate of packet's use from each codec that works on the RTP protocol.

Codec will change the sound into a code that sent in packets. Each packet will be shipped with different packetization interval of each codec. The number of packets that can be sent will be affected by transfer rate which calculated based on the number of bytes per second. Based on testing performed by the codec G.711 codec, it obtained the bandwidth used for one minute as in figure 10. Bandwidth used in the form of UDP packets.

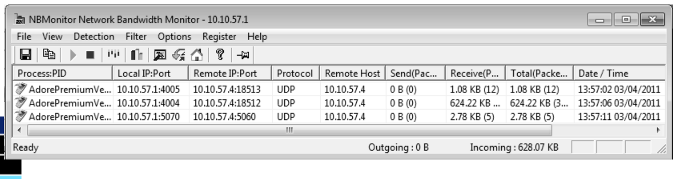

Figure 10 Flow packet data using codec g711

To obtain the bit rate of bandwidth usage, calculation is done by adding up all UDP packets and then divided the result by 60 seconds to obtain the bandwidth per second, then multiplied by 8 bits because the units used in the capturing process is Byte. For comparison 1 byte is equal to 8 bits. The number of UDP packets that obtained were 624.22 MB and after divided by 60 seconds and multiplied by 8 bits, the overall bandwidth used is 83.2 kbit / s.

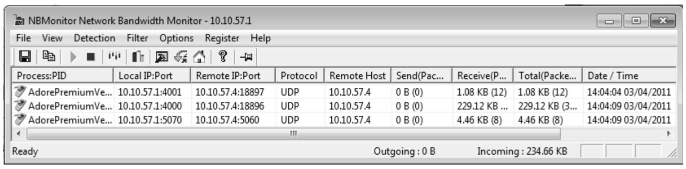

Figure 11. Flow packet data using codec gsm

Based on testing performed by GSM codec, bandwidth capturing is obtained as in figure 11. Entire bandwidth is also used the form of UDP packets. UDP packets obtained is 229.12 KB and after divided by 60 seconds and multiplied by 8 bits, the overall bandwidth used is 30.55 kbit / s.

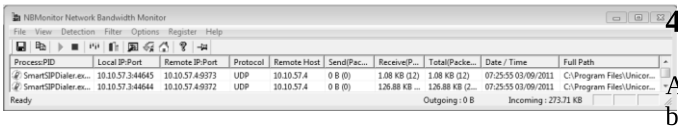

Figure 12. Flow packet data using codec g723

Based on testing performed by G.723 codec, bandwidth capturing is obtained as in figure 12. Entire bandwidth is also used the form of UDP packets. UDP packets obtained is 126.88 KB and after divided by 60 seconds and multiplied by 8 bits, the overall bandwidth used is 16.92 kbit / s.

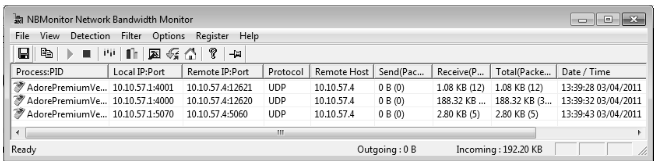

Figure 13. Flow packet data using codec g729

Based on testing performed by G.729 codec, bandwidth capturing is obtained as in figure 13. Entire bandwidth is also used the form of UDP packets. UDP packets obtained is 188.32 KB and after divided by 60 seconds and multiplied by 8 bits, the overall bandwidth used is 25.11 kbit / s.

3.2.3. Testing using Internet interconnection

Testing using Internet interconnection is done to review the sound quality when asterisk server is behind a NAT topology. In this experiment the codec used is also limited to a few codecs that g711, gsm, g729, and g723.

Testing have been done to obtain an indication of the optimal bandwidth that can be used for connections with low bandwidth traffic such as GPRS on the mobile phone. Interconnection to the Internet increase the flexibility for clients who are outside company network.

In the experimental results using GPRS, sound quality is still good enough even though there is some lagging because buffer of voice data. Calls from PSTN or intranet is better than GPRS calls. This is because the use of GPRS is strongly influenced by the signal. When GPRS have a good signal, the capacity to make calls then can being met.

When compared with the GSM, g729 codec quality is still better because there is no noise appeared. The using of g729 codec is dependent on the availability at each softphone. It requires more computational work to perform translational codec g729 codec so device must adjusting their capabilities.

4. Conclusion

Asterisk telephony system could be an alternative that is based on Open Source, because it was developed and can be freely configured by the GUI. G.729 codec is the best codec for sound quality because there was no noise. G.729 in another side using more bandwidth if compared with G.723. Extensions available on Asterisk can interconnect with existing PSTN numbers via voice gateway with Linksys brand such as SPA-3102.

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