Analysis and Design Quality of Service In the Voice Over Internet Protocol network Based on Session Initiation Protocol

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ABSTRACT

VoIP technology is beneficial because it uses IP-based network that already has a strong network in the world so that the cost to make calls much more efficient than using an analog phone. But VoIP has the disadvantage that the bandwidth required for transmission use RTP packets. This is a problem because the price of bandwidth in our country is still expensive, while the traffic, other traffic should also be sent by either the network connection is available. From this problem came an idea about how to manage the existing network connections and improve performance of VoIP networks themselves. One of the solutions is using QoS (Quality of Service). QoS itself is well known as one of a reliable method in dealing with network management issues, particularly for the delivery of important data. To implement these ideas, VoIP system on LAN was build which applying QoS Priority Queuing techniques. Then, the performances of VoIP systems before and after using QoS are analyzed. Is voice generated by the VoIP system meet the ITU-T standards based on delay, jitter and packet loss. From the testing using G711 codec, we found that the network performance after priority queuing is increasing sharply compared with before using priority queing. Parameters necessary to maintain the quality of VoIP networks can be maintained toward standardization value issued by the ITU-T which includes delay, jitter, packet loss and MOS (Mean Opinion Score).

Keywords:VoIP, QoS, delay, jitter, dan Priority queing.

1. PRELIMINARY

Voice over Internet Protocol (VoIP) is a technology that can pass the voice traffic, video and data packet form over an IP network. IP network itself is a data communications network based on packet-switching, so the phone using an IP network or the Internet. By using a VoIP phone, a lot can be taken advantage of them is certainly cost much cheaper than traditional phone rates, because the global nature of IP networks. So for international relations could be reduced up to 70%. In addition, maintenance costs can be in the press for voice and data networks separate, so the IP Phone can be added, removed and changed.

Problems at the end of this task is how to design an IP-based telecommunications system via the internet which is applied to the LAN consisting of 3 computers where one as a VoIP server, and two computers as a terminal / client,. the system will be streamed UDP and TCP traffic from one client to know and influence the traffic performance of VoIP systems include LAN network bandwidth, delay, MOS, packet loss. As well as we can determine the best QoS mechanisms

In planning to implement VoIP on a Local Area Network (LAN), to calculate the bandwidth requirements, because the estimation of voice transmission bandwidth allocation becomes very important because it will take most of the existing communications bandwidth. Thus techniques to compress the data to be very strategic to enable communication bandwidth savings.

VoIP can communicate with other systems that operate on packet-switch network. To be able to communicate needs a communication system standards compatible with each other. There are two communication standards used in VoIP are:

1.1 H.323

H.323 standard consists of the components, protocols, and procedures that provide multimedia communications over packet-based networks. Form of packet-based networks that can be taken include the Internet network, the Internet Packet Exchange (IPX)-based, Local Area Network (LAN) and Wide Area Network (WAN). H.323 can be used for services - multimedia services such as voice communications (IP telephony), video with voice communications (video telephony), and the combination of voice, video and data.

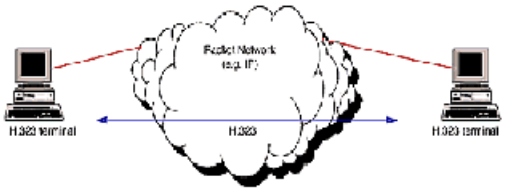


Figure I.1 Terminals on the packet network

Destination design and development of H.323 is to allow interoperability with other types of multimedia terminals. Standard H.323 terminal can communicate with H.320 terminals on N-ISDN, ATM terminals in H.321, and H.324 terminals on the Public Switched Telephone Network (PSTN). H.323 terminals enables real-time communication in the form of two-way voice, video and data.

1.2 Session Initiation protocol (SIP)

SIP is a signaling protocol that aims to control the initiation, modification, and terminating multimedia sessions, including sessions on audio or video communications. SIP is a text based protocol similar to HTTP and Simple Mail Transfer Protocol (SMTP). SIP is a protocol peer-to-peer means that the functions of call routing and session management distributed to all nodes (including endpoint and server) in the SIP network. This is different from the conventional telephone system in which telephone terminals is dependent on a centralized switching devices. SIP has the functions that are defined as follows:

a. User location
SIP provides the ability to find the location of end users who intend to establish a session or send a request.

b. User capabilities
SIP allows determination of the media capabilities of the devices involved in the session.

c. User availability
SIP enables the user desires determination to communicate.

d. Session setup
SIP allows modification, transfer, and termination of an active session.

2. MODEL, ANALYSIS, DESIGN AND IMPLEMENTATION

Network configuration used for testing are as follows:

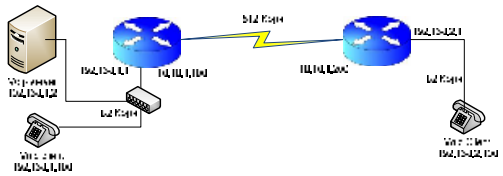


Figure 2.1 Network Configuration

Limited testing conducted as follows:

a. G.711 as the codec used.
b. It is assumed perfect channel conditions, ie there is no transmission errors and link Adaptations
c. Parameters used to observe the quality of services include bandwidth, jitter, MOS and packet loss
d. IP addresses using IP version 4
e. Win32 asterisk is used as a VoIP server.

Below is the configuration of components that are on the network configuration are made:

a. IP Phone
IP Phone is a commonly used hardware for VoIP communication. In this section occurred processing a digital signal to analog and then made a packetization process IP packets.

b. Switch
This device serves as a connecting all the devices in a LAN network.

c. Router
This device serves to perform perutingan ip address from two different networks.

Router used here is a version of RouterOS software. In mikrotik we will do limitation backbone connections for 512 kbps, 256 kbps and 128 kbps.

d. *Server*
Server is an application service provider in a network. in this case is the VoIP server. VoIP server used is an asterisk versions of Windows

2.1 Testing Scenario

Testing Scenario testing to be conducted are as follows:

a. *Testing the system with Internet bandwidth 512 Kbps*
The system will be tested at 512 kbps backbone connection, and will compare how the network performance before and after the applicable priority queuing.

b. *Testing the system with Internet bandwidth 256 Kbps*
The system will be tested at 256 kbps backbone connection, and will compare how the network performance before and after the applicable priority queuing.

c. *Testing the system with Internet bandwidth 128 Kbps.*
The system will be tested at 256 kbps backbone connection, and will compare how the network performance before and after the applicable priority queuing.

3. DATA ANALYSIS RESULTS

Based on the test results conducted using several test scenarios. We can analyze the data generated by the program yagn these commview. To facilitate the reader, the author of the study presented in graphical form as follows:

3.1 Analysis parameter packet loss

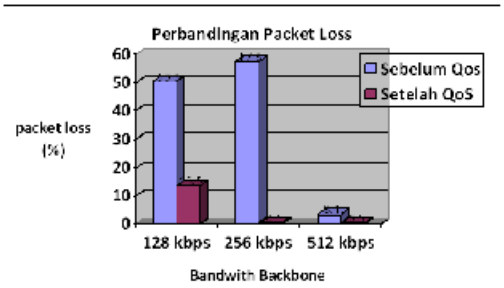


Figure III.1 Graph packet loss ratio

From the graph above it can be seen, the number of packets lost during the trial before the applicable priority queuing at backbone connection 128 kbps and 256 kbps is very much at all, but after priority queuing implemented the missing package is very much reduced. Based on the graph above the highest reduction occurred during testing at 256 kbps backbone connections where there are QoS applied before the missing packet of 50.3%, but when applied to a decline in the number of QoS lost packets to be at 0.5%. The impact of the decline of this package was lost, because the sound became better and can be heard clearly than ever before.

3.2 Analysis parameters jitter

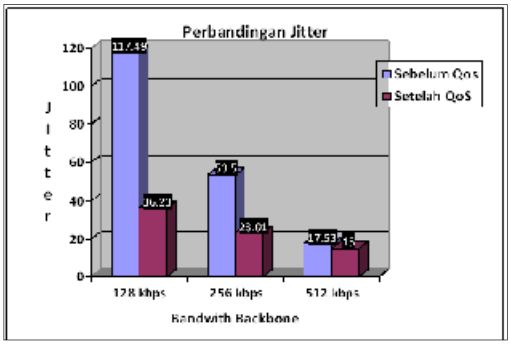


Figure III.2 jitter comparison graph

From the graph above can be known jitter parameters decreased significantly. At the time of the trial before priority queuing applies very much so, but once implemented priority queuing jitter is very much reduced. Can be seen in the graph above at the time of the communication process conducted at 128 kbps connection there backbone jitter of 117.49 but after having implemented priority queuing for a reduction to only 36.23. The

highest reduction occurred during testing at 128 kbps backbone connections where a decline in almost 300% more.

3.3 Analysis parameter MOS

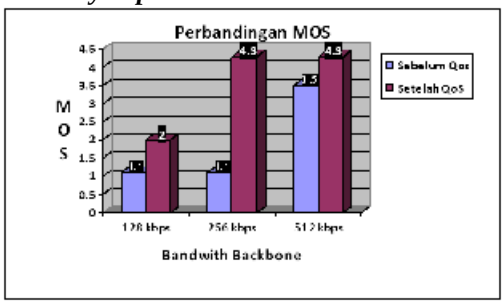


Figure III.3 MOS comparison graph

From the graph above it can be seen the results of MOS grades improved a lot after priority queuing is implemented. The increase is highest at the time of trial in 256 kbps backbone connections, where there was an increase the value of the original MOS 1.1 to 4.3 in value is very good if we implement VoIP using G.711 codec. From the above test results can be concluded that the implementation of priority queuing in a VoIP network system is very effective to help improve the quality of the communication connection in a VoIP system. With this method applied to the resulting impact is very big, where as shown in the graph above there is packet loss reduction is quite significant when compared with the prior applied this method, the result is packet loss reduction also affects the reduction of delay / jitter so as the resulting sound quality, including the good category.

4. CONCLUSION

From the analysis has been done on the planning and implementation of quality of service in voice over Internet protocol, by doing some testing stages, can be deduced as follows:

a. RouterOS can be used to improve performance LAN networks with priority queuing methods.
b. Bandwidth that is used in VoIP communications Average - average of 83 kbps using G.711 codec for each one channel. So the internet connection rate is already low enough at this time we can hold voip service to connect offices located in various regions.
c. Quality of service on a LAN network is a must, because application development is increasingly rapid and bandwidth consumption that is used is also higher.

SUGGESTIONS

a. Voip application is appropriate for institutions or companies that already have internet network between its offices
b. Voip application for more than 3 channels should use the connection to a dedicated Internet connection to prevent any loss of quality when the ISP solid.
c. For further development, it is necessary for the development of interconnection mechanisms with the VPN network or PSTN.

5. REFERENCES

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