Performance Analysis of VoIP Quality of Service in IPv4 and IPv6 environment

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Abstract
This paper focuses on the study of different Internet protocols used in computer networks in order to measure the Quality of Service (QoS) between Internet protocol version 4 and Internet protocol version 6. Routing protocols are the main factors contributing to speed up data transfers within the network. The performance of the routing protocols can be tested by their convergence time, link throughput and application layer service performance such as video streaming and voice conferencing. In routing protocols, convergence time is an important aspect in indicating routing protocol performance. A simulation model was used to compare the performance, based on several parameters that determine the Quality of Service of these internet protocols on Voice over IP (VoIP).

Keywords: Quality of Service (QoS), Voice over IP (VoIP), Routing protocol.

1. Introduction
In recent years, there is a growing trend in real-time voice communication using Internet protocol (IP). VoIP provides a solution that merges both data and voice, which gains benefits, including cost savings, high quality and value added services. Today, VoIP is becoming one of the most widely used technologies. There are various VoIP communication software products that are already available on the Internet: Skype, Google Talk, and Windows live messenger. All of them can provide good quality, cheap and even free phone calls [Skype official], [Windows Live Messenger], [Google Talk]. VoIP is now the most popular telecommunication technology with an estimated user increase in the past three years, due to implementation of broadband internet access, from about 480,000 users in 2006 to almost 70.6 million in 2011 [1]. As a revolutionary technology, VoIP has the potential to revamp completely the world’s phone system and continue to touch every aspect of our lives ranging from the way we interact, how we interact, where we interact from and how effective is our interaction. Figure 1 depicts successful deployment of VoIP platform in Telekom Malaysia(TM). In order to ensure these QoS, it is important that VoIP system developers consider how the voice has been transported to the receiver, and the packets delay encounters during the process of transmission over the network.

![Figure 1. iTalk Buddy Network Diagram for TM](image-url)
2. Related work

2.1. IPv4-IPv6 Translator for VoIP and Video Conferencing

Our contributions include: (a) Translation based on a 4-tuple socket identifier corresponding to a connection. It overcomes some of the limitations of the earlier NAT-PT [3] approach; in particular, the limitation of "one server per service". (b) Support for SIP, SAP and v4-v6 multicast address translation. NAT-PT fail to translate protocols like SIP, which embed IP address and port information in the application payload. (c) Unlike the earlier approaches for SIP translation [1], it is not required to modify IPv4 or IPv6 end hosts. SBIIT is a completely transparent solution. (d) Design and implementation of the complete enhanced solution in hardware, based on the Intel network processor IXP 435 [4] [5].

2.2. A Comparison of VoIP Performance on IPv6 and IPv4 Networks

As the growing popularity of VoIP will make it a significant component of traffic in the future Internet, it is of interest to compare VoIP performance over IPv6 and IPv4. The results would help to determine if there are any differences in VoIP performance over IPv6 compared to IPv4 due to overhead resulting from the larger IPv6 header (and packet size). We focus on comparing VoIP performance with IPv6 and IPv4 during the exchange of voice data only. Tests are conducted on a LAN in the presence of competing UDP traffic using a softphone running on the Linux and Windows operating systems. A softphone on a bare (operating system-less) PC is used as a control to determine the extent to which operating system, conventional protocol stack, and application overhead impacts VoIP performance over IPv6 compared to IPv4. Performance is measured using maximum and mean values of delta (the time between voice packets), maximum and mean jitter (delay variation), packet loss, MOS (Mean Opinion Score), and throughput. We also determine the relative frequency distribution of delta values and compare the IPv4/IPv6 throughput ratio with the theoretical (ideal) value[7,8].

3. Background

3.1. Video Streaming Over IP and Application Layer

Real-time transport of live video is the predominant part of real-time multimedia. In the streaming mode, the video content need not be downloaded in full, but is being played out while parts of the content are being received and decoded. Due to its real-time nature, video streaming typically has bandwidth, delay and loss requirements. Figure 2 shows the architecture for video streaming [10].

![Video Streaming Architecture](image)

Figure 2. Video Streaming Architecture
Raw video and audio data are recompressed by video compression and audio compression algorithms and then saved in storage devices. Upon the client's request, a streaming server retrieves compressed video/audio data from storage devices, and then the application-layer QoS control module adapts the video/audio bit-streams according to the network status and QoS requirements. After the adaptation, the transport protocols packetize the compressed bit-streams and send the video/audio packets to the Internet. Packets may be dropped or experience excessive delays inside the Internet due to congestion. To improve the quality of video/audio transmission, continuous media distribution services (e.g. caching) are deployed in the Internet. For packets that are successfully delivered to the receiver, they first pass through the transport layers and then are processed by the application layer before being decoded at the video/audio decoder. To achieve synchronization between video and audio presentations, media synchronization mechanisms are required [11]. The protocol stack is as shown in Figure 3.

![Protocol Stacks for Multimedia Streaming](image)

**Figure 3. Protocol Stacks for Multimedia Streaming**

At the data plane, (the sending side) the compressed video/audio data is retrieved and packetized at the RTP (real-time transport protocol) layer. The RTP packetized streams provide timing and synchronization information, as well as sequence numbers [3]. Since RTP can be configured for low latency, it is useful for interactive conversations as well as streaming media. The RTP packetized streams are then passed to the UDP/TCP layer and the IP layer. The resulting IP packets are transported over the network. At the receiver side, the media streams are processed in the reversed manner before their presentations. For the control plane, real-time control protocol (RTCP) packets and real-time streaming protocol (RTSP) packets are multiplexed at the UDP/TCP layer and move to the IP layer for transmission over a network. The RTCP is a companion protocol to RTP for gathering statistics on a media connection and information such as bytes sent, packets sent, lost packets, round trip delay and jitter [12].

### 3.2. VoIP Protocols

There are two standard protocols used in VoIP network: Session Initiation Protocol (SIP) and H.323, (Skype) and some others use proprietary signaling and messaging protocols. H.323 (ITU-R Rec. H.323 1999) is International Telecommunication Union (ITU) standard based on Real-time Protocol (RP) and Real-Time Control Protocol (RTCP); H.323 is a set of protocols for sending voice, video and data over IP network to provide real-time multimedia communications. It is reliable and easy to maintain technology and also the recommendation standard by ITU for multimedia communications over LANs [13, 14]. Figure 4 shows the H.323 architecture.
There are four basic entities in a default H.323 network [14, 16], terminal, gateways (GW), gatekeepers (GK) and multipoint control units (MCU). H.323 terminal, also called H.323 client, is the end-user device. It could be IP telephone or a multimedia PC with another H.323 client that provides real-time two-way media communication. A Gateway (GW) is an optional component that provides inter-network translation between terminals whereas a Gatekeeper (GK) is an optional component provides address translations and access control services. A Multipoint Control Unit (MCU) function as a bridge or switch that enables three or more terminals and gateways in a multipoint conference.

3.3. QoS of Multimedia over IP (MoIP)

Quality of Service (QoS) generally describes the assurance of sufficiently low delay and packet loss for certain types of applications or traffic. Guarantees of QoS are important if the network capacity is limited, especially for real-time streaming multimedia services, since these often required fixed bit rates and are delay sensitive. QoS is affected by various factors, which include bandwidth, drop rate, delay, jitter and etc. QoS is fundamental to the operation of MoIP. ITU/ETSI and IETF approaches for general QoS model consist of [15].

- **Intrinsic QoS** – pertains to service features stemming from technical aspects.
- **Perceived QoS** – reflects the customer’s experience of using a particular service.
- **Accessed QoS** – starts to be seen when the customer decides whether to continue using the service or not.

QoS parameters in MoIP focus on intrinsic QoS in packet networks and expressed by at least the following set of parameters that are meaningful for most IP based services [3]:

- **Latency (called delay)** – the time it takes for a multimedia transmission to go from its source to its destination. ITU-T recommendation upper bound is 150 ms for one-way traffic.
- **Jitter (called delay-variation)** – non uniform delays, requires buffering at the endpoints and application level reordering. Increased jitter makes it harder to tell when a packet is missing or just late.
- **Bandwidth and Effective Bandwidth** – bandwidth congestion can cause packet loss and a host of other QoS problems. Proper bandwidth reservation and allocation are essential to MoIP quality. Not only is the available bandwidth of the system affected by the introduction of security mechanisms, but in addition, the effective bandwidth of the MoIP system is significantly depreciated. Where \( N_d \) = number of packet received at destination, \( P_s \) = packet size in unit of byte, and \( t \) = total time of simulation in unit of second. The unit of bandwidth is Mbps.

\[
\text{Bandwidth} = \frac{N_d \times P_s \times 8}{t \times 1 \times 10^6}
\]
• **Packet Loss (called dropped packet)** – MoIP is highly sensitive to packet loss. Loss rates as low as 1% can garble communications. Latency and Jitter can contribute to “virtual packet loss” as packets arriving after their deadlines are as good as “lost”.

### 4. System Design

The design process is divided into two parts; the first is designing the network. The performance for each scenario is analyzed and the second shows performance for all scenarios and comparison between them. The design is for using VoIP as a sample design for all technologies. Consider two types of traffic sources that are video streaming and VoIP. The test is divided into four separate sessions. The first session assesses the performance of video streaming with setting up 10 bi-directional pairs of MPEG-4 encoded video (UDP) stream over IPv6 wired network comparison with IPv4 network. The second session analyzes the performance of video streaming with setting up 10 bi-directional pairs of MPEG-4 encoded video stream over IPv6 wireless network comparison with IPv4 network. The third session assesses the performance of VoIP with setting up 10 pairs of G.711U-based voice over IP calls (RTP) at 64 Kbps per stream over IPv6 wired network comparison with IPv4 network. Begin this test with a fixed duration of five minutes. The final session analyzes the performance of VoIP with setting up 10 pairs of G.711U-based voice over IP calls at 64 Kbps per stream over IPv6 wireless network, which is compared with the IPv4 network. The test begins with a fixed duration of five minutes. Each station was equipped with an Intel Pentium IV 2.60 GHz processor, 1 gigabytes of SDRAM, 60 gigabytes IDE hard drives, 100 Mbps PCI Ethernet network adapters and 802.11g wireless adapters. Operating system client was WindowsXP-SP2, using the Iperf measurement tool for studying the transmission and reception data on a network. Iperf is a simple command line tool to measure network performance, which can measure both TCP and UDP bandwidth performance. It is a tool to measure maximum TCP bandwidth, allowing the tuning of various parameters and UDP characteristics. Iperf reports bandwidth, delay Jitter and packet loss. The video streaming software that used in the experimental is VideoLAN, It is a complete software solution for video streaming and playback, developed by students of the Ecole Centrale Paris and developers from all over the world, under the GNU General Public License (GPL). VideoLAN is designed to stream MPEG videos on high bandwidth networks.

#### 4.1 Performance Metrics

**Throughput** is the rate at which bulk data transfer can be transmitted from one host to another over a sufficiently long time period. According to [17, 18], the application measurement of application throughput, which demonstrates the end-to-end performance can be delivered to the end user over IPv6-IPv4 protocol stacks in the unit of Mbps. The throughput of a connection can be characterized by the following formula. The unit of throughput is Mbps.

\[
\text{Throughput} = \frac{N_d \times P_S \times 8}{(T_{sp} - T_{st}) \times 1 \times 10^6}
\]

Where,

- \(N_d\) = number of packet received at destination,
- \(P_s\) = packet size in unit of byte,
- \(T_{sp}\) = stop time of each traffic flow in unit of second,
- \(T_{st}\) = start time of each traffic flow in unit of second.

**Packet loss** is the percentage of the number of packet that losses in a connection. Normally, when a device/path is overloaded and cannot accept any incoming data at a given time, then packet loss occurs. Heavy packet loss has a great impact, especially on video quality in either continuous or real-time perspective.

\[
\text{Packet Loss} = \frac{N_s - N_d}{N_s} \times 100\%
\]

Where,

- \(N_s\) = number of packet generated at source,
\[ \text{Mean latency} = \frac{\sum_{i=1}^{N} (T_{di} - T_{si})}{Nd} \] (4)

Where
- \( T_{di} \) = time of packet “i” received at destination,
- \( T_{si} \) = time of packet “i” enqueued at source,
- \( Nd \) = number of packet received at destination.

5. Results and Discussion

This section presents and discusses the experimental results obtained from tests of video streaming and VoIP services over IPv4 and IPv6, both wired and wireless networks. All performance evaluation criteria specified previously were calculated and compared to each other.

**Throughput:** The throughput of video streaming service for both IPv4 and IPv6 protocol over wired and wireless networks were measured. The experimental results are illustrated in Figure 5.

![Video Streaming](image1)

**Figure 5.** Throughput of Video Streaming.

Figure 6 shows that the average of throughput is similar for IPv4 and IPv6 with both wired and wireless network. Throughput of IPv6 wired network is the highest. However, the average throughput of video streaming is higher than VoIP service. The experimental results of the throughput of VoIP of IPv4 and IPv6 wired networks are illustrated in Figure 6.

![VoIP](image2)

**Figure 6.** Throughput of VoIP
Figure 7 shows that for the wired network, average throughput with IPv6 is around 19 Mbps higher compared to the IPv4 for VoIP service, which is about 15%. On the wireless network, IPv6 network has throughput higher than IPv4, around 12%.

Mean Latency: The experimental results for latency measurement for both IPv4 and IPv6 protocol over wired and wireless networks are illustrated in Table 2.

<table>
<thead>
<tr>
<th>Case</th>
<th>Video Streaming</th>
<th>VoIP</th>
</tr>
</thead>
<tbody>
<tr>
<td>IPv4 wired network</td>
<td>191</td>
<td>182</td>
</tr>
<tr>
<td>IPv4 wireless network</td>
<td>205</td>
<td>175</td>
</tr>
<tr>
<td>IPv6 wired network</td>
<td>185</td>
<td>181</td>
</tr>
<tr>
<td>IPv6 wireless network</td>
<td>194</td>
<td>185</td>
</tr>
</tbody>
</table>

In general, the latency analysis of both IPv4 and IPv6 networks on wired LAN suggests that the average latency time observe in IPv6 is less than that in IPv4. This latency variation happens because normally, IPv6 modifications resolve the signaling redundancies of IPv4 and thus perform better in terms of latency or delay.

Figure 7 shows the architecture for providing VoIP over IPV6 networks [18]. It provides an operating environment for VoIP that allows calls between IPv4 and IPv6 user agents regardless of the heterogeneity in the network addressing protocols used. The main components of the system are presented in the figure. The SIP server (SER) acts as SIP registrar and proxy for setting up calls. The IPv4-IPv6 gateway acts as the translating bridge between the two networks. The PSTN gateway provides access to the PSTN so that users on normal analogue telephones can interact with users using the VoIP infrastructure. Finally, the MCU acts as a conferencing application for multiparty communication. The architecture also includes various user agents.

The two scenarios for the two networks(IPv4 and IPv6) design have been chosen and implemented for a specific number of workstation and server in one scenarios, then representing multimedia streaming( video and voice) in the two networks( IPv4and IPv6). Two different applications are used that can help us to measure the performance of the internet protocols, video and voice, so that specific parameters are selected and the results of both protocol are compared on one graph based on the selected parameter. The simulated network was tested in some metrics in order to measure the Quality of Service between IP (version 4) and IP (version 6) on Queuing Delay Variation, Throughput (Packet per second and Utilization. The queuing delay variation shows that the delay variation value start from the beginning, exactly at 0000.25 m. However, in IPv4 scenario, the queuing delay variation begins at 0.000025 m. Queuing delay variation for IPv6 is as shown in Figure 8.
Figure 8. IPv4 queuing delay variations

We can say that the throughput of IPV4 scenario in the first time during the simulation is exactly at 0.066 s, is from 0m to 1m. There is a second wave from 1m to 3m during the simulation testing, after that on subnet_HQ the throughput coming at the same time but the second wave is smaller than the wave in subnet branch. After that it is constant and equal to zero, and remains like that until the end of simulation testing as shown in Figure 9.

Figure 9. Throughput Packet per second IP (Version 4)

The utilization values start constant, equal to zero and remain like that until the end of the simulation time. However, in IPV4 scenario, the delay variation is less than the queuing delay and throughput which are shown in Figure 10.

Figure 10. Utilization IP (Version 4)

Figure 11 shows the Queuing delay value is starting from the beginning at 0000.23 s, 0000.25 s, i.e. at the first and second minutes of simulation time and disappears in the next minutes of simulation. It
starts to appear again after six minutes on subnet_HQ and appearing again randomly through the simulation time.

![Figure 11. Queuing delay in IP (Version 6)](image)

Figure 11. Queuing delay in IP (Version 6)

Figure 12 shows the throughput value start at 0.38 packet/second at the first minute of simulation time and decreased gradually at the second minute and going to be equal zero at the third minute and staying constantly and on minutes 5 of the simulation time. It starts to appear again and disappearing, which seems to be unstable during the time of simulation.

![Figure 12. Throughput Packet per second in IP v6](image)

Figure 12. Throughput Packet per second in IP v6

The utilization value starts constant which is equal to zero in IPV6 scenario and remains constantly through all the duration of simulation time. However, in IPV6 scenario, the utilization is less than the other parameters like queuing delay variation and throughput. For the utilization, the result value for IPV4 and IPV6 is the same, as shown in Figure13.
The results of the simulation between IPv4 and IPv6 are compared, based on metrics such as queuing delay, throughput and utilization in order to measure the QoS between internet protocol(version 4) and internet protocol(version 6) on voice over IP (VOIP). For the queuing delay, observation over this result, there are two main connections between IPv4 and IPv6. For the first scenario subnet_HQ_IPv4 to interconnection have one straight connection and after that the result extends to have delay variation at 1\textsuperscript{st} minute. After the connection getting more than one minute, it looks like the entire simulation testing which is 29 minutes is getting stable and consistent and no more intermittent between that duration. However, for the second scenario subnet_HQ_IPv6 to interconnection have also one straight connection and after that the result has a queuing delay variation at the 0m of simulation duration. After the connection getting more than one minute, it looks like getting stable until the eighth minutes start to have a variation. After that it starts getting disappears and appears again, which looks unstable during the entire simulation testing which is 30 minutes, as shown in Figure 14.

For throughput, there are two main connections between IPv4 and IPv6. For the first scenario subnet_HQ_IPv4 to interconnection have one straight connection and after that the result extends to have another wave from 0m to 1m. After the connection getting more than one minute, which looks like the entire simulation testing which is 29 minutes getting stable and consistent and no more intermittent between that durations. However, for the second scenario subnet_HQ_IPv6 to interconnection have also one straight connection and after that the result has one more wave from 0m to 1m. After the connection getting more than one minute, it looks like getting stable, but it has one
more wave at eighth minutes of the simulation time and stays appearing on other times during the entire simulation as shown in Figure 15.

![Figure 15. IPV4/IPV6 Throughput Packet per second](image1.png)

For utilization, there are two connections between IPv4 and IPv6. The utilization value starts with constant equal zero in all scenarios internet protocols, and it stays getting stable and consistent and no intermittent during the entire simulation time which is 30 minutes as shown in Figure 16.

![Figure 16. IPV4/IPV6 Utilization](image2.png)

The Internet protocols like IPV4 and IPV6 are widely being used in computer networking. In this paper, a comparative analysis of selected internet protocols such as IPV4 and IPV6 is presented. The comparative analysis has been done in the same network for same application (video streaming and voice) with proposed protocols for real time applications. Performance has been evaluated the quality of service performance on the based on some parameters aimed to figure out the effects of the internet protocols.
6. Conclusion

To understand and evaluate the performance measurements of protocols, several tests were performed based on different scenarios. After performing a number of scenarios on each application (video and voice) over different internet protocols, a set of required result was obtained which were explained in the result and discussion section. In order to optimize the QoS network performance, one could also implement the designed tool into their running network to evaluate the recoverability performance of different protocols and ultimately chooses the best solution.

7. References


