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Abstract

Carrying voice over Internet Protocol, known as VoIP, is a rapidly growing technology. It is an interesting real-time application which requires constant bit streaming of application data (voice). There are three protocols – TCP, UDP and SCTP – at the transport layer to carry the data between a source node and a destination node in the Internet. TCP is a reliable, byte-oriented protocol with built-in congestion control mechanism. UDP is an unreliable, connectionless protocol without any congestion control mechanism, hence, may cause congestion collapse. SCTP is a reliable message-oriented protocol which has been designed for newer applications, such as voice, which require more sophisticated service than TCP can provide. In this paper, we have made an attempt to evaluate these protocols for voice application under various congestion scenarios. We see that SCTP and UDP compete with each other under the considered quality metrics for voice transmission.

1. Introduction

Voice over Internet Protocol (VoIP) refers to digitization of voice streams and transmitting the digital voice packets over a conventional IP-based network [1]. Carrying voice traffic over Internet Protocol (IP) is an interesting and relatively new real-time application over Internet which requires constant bit streams of application data (voice). The use of VoIP is growing because of low cost of communication compared to Public Switched Telephone Network (PSTN) and increased functionality. However, concerns about the Quality of Service (QoS) of VoIP are inhibiting its proliferation more rapidly. The factors which influence quality of VoIP service include speech codec, packetization, packet loss, delay and delay variation (jitter) [2].

Every VoIP system uses the underlying IP network for the transmission of packetized voice.

The IP traffic is based on best-effort communication, i.e., guarantee for the delivery is not provided. The IP routers transmit the packet on first-come, first-serve basis. These characteristics introduce large delays, large delay variations and packet loss which are the most important concerns of QoS in VoIP. As the guaranteed delivery is provided by the higher layer protocols, transport layer protocols play a major role in deciding the voice quality of a VoIP system. In this paper, we analyze the VoIP traffic and study the performance of transport layer protocols (UDP, TCP and SCTP) under various network scenarios for voice application with reference to three quality metrics: packet loss, delay and delay variation.

Rest of the paper is organized as follows. Section 2 describes the quality of metrics. A brief review of related work is presented in Section 3. We describe our experimental model and results in Section 4. Finally, we conclude in Section 5.

2. Quality Metrics

2.1 Packet loss

VoIP packet loss occurs when a large amount of traffic hits the network and causes it to drop packets. Packet loss is mainly attributed to link failure, high levels of congestion that lead to buffer overflow and occasional misrouted packets. It results in dropped conversations, a delay in receiving the voice communication or extraneous noise on the call. Unfortunately, packet loss occurs frequently in data networks and dropped voice packets are discarded; they are not retransmitted. High packet loss can cause noticeable degradation in the call quality. Packet loss of 1% translates into one voice clip every 3 minutes whereas packet loss of 0.25% translates into one error every 53 minutes [3]. Voice traffic can tolerate less than 3% loss of packets before callers feel perceivable gaps in conversation.

2.2 Delay
Transmission time or delay is the average time it takes for a packet to travel from its source to its destination. Delay causes disruption in the voice quality when voice packets take more time than expected to reach their destination. The International Telecommunications Union (ITU) recommendation G.114 [4] considers network delay for voice applications. This recommendation defines 3 bands of one-way delay with adequately controlled echo as shown in the table 1. A voice call can tolerate maximum latency (delay) of 150 milliseconds, however, 100 milliseconds is preferred.

2.3 Delay Variation

If the delay is small and constant, the speech may be acceptable. However, the delay is variable in packetized networks; this variation in delay is known as jitter. It is effectively a variation of packet delay where delays actually impact quality of the conversation. It can be handled through the de-jitter buffer at the receiving router/gateway. De-jitter buffer imposes a delay on early packets and passes late packets with less delay to compensate the variable delay. Any packet which arrives later than the length of buffer is discarded. The buffer should be able to compensate the maximum delay variation that we expect. It will minimize packet loss too. Now onwards, we will use the term ‘jitter’ for delay variation in the paper.

<table>
<thead>
<tr>
<th>Range in milliseconds</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>0-150</td>
<td>Acceptable for most user applications</td>
</tr>
<tr>
<td>150-400</td>
<td>Acceptable for international connections</td>
</tr>
<tr>
<td>Above 400</td>
<td>Unacceptable for general network planning purposes; however, it is recognised that in some exceptional cases this limit will be exceeded</td>
</tr>
</tbody>
</table>

3. Related Work

Chang and Su [5] compare the performance of TCP and UDP for P2P streaming media with reference to average values of missing index and average download rate. They conclude that both UDP and TCP provide reasonable service quality but TCP is more stable and reliable. It is partly the reason that most P2P streaming systems use TCP as its transport protocol. Kiesel and Scharf [6] study the performance of SCTP, TCP and UDP over a firewall signaling protocol - simple middlebox configuration protocol (SIMCO). They observe that SCTP significantly outperform TCP because of its ability to transmit over multiple streams; SCTP reduces the response time of SIMCO. TCP and UDP both suffer from significantly larger delays.

Ha et al. [7] study the throughput performance of SCTP and TCP over the Linux platform under different scenarios as variable size of user’s input data and fairness under competition of traffic. They observe that SCTP outperform TCP in throughput and fairly competes with it. They also observe that multi-homing SCTP provides better performance than the single-homing case. Camarillo et al. [8] implements SIP over SCTP, UDP and TCP protocols under different network conditions and observe that UDP is good only for the light traffic. Under heavy traffic load, TCP and SCTP are better than UDP. However, SCTP has some advantage over TCP owing to its features as multi-streaming and multi-homing. In general, SCTP performance increases with worsening of network conditions.

Wang et al. [9] study the performance of PR-SCTP (an unreliable extension of SCTP), TCP, and UDP for MPEG-4 video traffic over mobile networks. They observe that PR_SCTP is more suitable for real-time traffic as its selective retransmission can improve the quality of images. Moreover, it can provide unified congestion control to both reliable and unreliable traffic which is not possible by any other protocol.

Kumazoe et al. [10] study the throughput characteristic of high speed transport protocols: HighSpeed TCP (HSTCP), Scalable TCP and Simple Available Bandwidth Utilization Library (SABUL) experimenting on the Japan Gigabit Network (JGN). They observe that in TCP-based protocols the simple and robust strategy of ACK packets for rate and error control works well with variety of network circumstances but it may not be suitable for long distance networks because performance depends on both the distance and receiver-side TCP implementation. Rate control and error control mechanisms should be separated for better performance.

Rajamani et al. [11] study the performance of SCTP and TCP over HTTP protocol and conclude that SCTP can help reduce the latency and the improved throughput. Additionally, other features of SCTP like multihoming and better protection against DoS attacks makes it more suitable for future web applications. Dantas and Jardini [12] compare and observe that Xpress Transport Protocol (XTP) is superior to TCP for reliable multicast communications. Heidemann et al. [13] develop an analytical model and use it to compare the relative performance of TCP, Transaction TCP and request-response protocol based on UDP over HTTP for various network characteristics and workloads.

Hence, the literature suggests that performance study of transport layer protocols is crucial for any IP based network application. VoIP is becoming increasingly popular and has the potential to replace
the existing telephony system. A VoIP application has two phases: signaling and voice transmission. In [8], the authors have evaluated the performance of transport layer protocols for SIP which is a VoIP signaling protocol. It motivated us to evaluate the performance of various transport layer protocols in different network conditions for streaming media applications, e.g., VoIP.

4. Experimental Model and Results

4.1 Experimental model

We consider the simulation model as suggested by Camarillo et al. [8] and use the network simulator (ns 2.31) for the simulation. Figure 1 shows the typical network topology. Nodes 2 and 3 are buffer-limited tail drop routers which treat each packet identically and drop the newly arrived packets if queue is full to its maximum capacity until the queue has enough room to accept incoming traffic. Other nodes are the end nodes. Nodes 1 and 5 are the VoIP source and sink respectively. Nodes 0 and 4 are used for simulating FTP traffic while the nodes 6 and 7 are used to simulate the pack mime HTTP traffic. A single PackMime-HTTP client node generates HTTP connections coming from a “cloud” of web clients. Likewise, a single PackMime-HTTP server node accepts and serves HTTP connections destined for a “cloud” of web servers. There are many client applications assigned to a single client node and many server applications assigned to a single server node [15]. All the simulated traffic shares the common link 2-3. Thus, this link acts as a bottleneck for the entire traffic. We can easily study the network behavior in different networking scenarios by changing the parameters for this bottleneck link.

Figure 1. Simulated Network Topology

In this monogram, we study the transport layer protocols with the three performance metrics: (i) packet loss, (ii) delay, and (iii) jitter for the VoIP traffic between node 1 and node 5 by changing the parameters of the bottleneck link 2-3 for creating different scenarios. We study three different scenarios: (i) hLhB - a high latency, high bandwidth scenario with a delay of 140ms and bandwidth of 2Mbps, (ii) hLlB - a high latency, low bandwidth scenario with a delay of 140ms and bandwidth of 0.6 Mbps, and (iii) lLhB – a low latency, high bandwidth scenario with a delay of 70ms and bandwidth of 2 Mbps. The parameters between other links (except link 2-3) are as shown in Figure 1 and were not changed during the experiments. We simulate each scenario for 300 seconds and compute mean value of the delay and delay jitter for every consecutive time interval of 5 seconds.

4.2. Results

High Latency, High Bandwidth (hLhB) Scenario: Figures 2 and 3 represent the end to end delay and jitter respectively. Figure 2 shows the expected trends of the delay. The initial peaks in the delay of SCTP and TCP are due to the initial handshake at the time of connection establishment. As SCTP has a four way handshake [18], its initial peak is the highest. The intermediate peaks in TCP and SCTP are due to the high congestion and a possible packet loss, which increases the delay. The UDP shows a constant delay as, being a connectionless protocol, it maintains the data rate. We observe, initial low value of delay in UDP (refer, Figure 2) because the competitive traffic of TCP and SCTP are still in connection establishment stage then, hence giving UDP a free link to communicate. Later, when the TCP, SCTP, FTP and HTTP traffic establishes completely, the delay increases because of the high traffic.

Figure 2. Delay in hLhB scenario

Figure 3 shows the jitter. As UDP does not have a congestion control and packet re-transmission mechanisms [16], it continuously sends the packets at a constant rate in to the network. Thus, we do not observe much difference in the delay. As a result the jitter, which is the variation of packet delays, is very low. In case of TCP and SCTP, we observe peaks of jitter because of the congestion control and packet re-transmission mechanisms in these protocols.
Though UDP performs better in delay and jitter, we observe comparatively more packet loss (refer, Table 2). However, it is well within the limit to cause any noticeable degradation in the call quality.

<table>
<thead>
<tr>
<th>Protocol</th>
<th>Packets Sent</th>
<th>No. of packets lost</th>
<th>% loss</th>
</tr>
</thead>
<tbody>
<tr>
<td>SCTP</td>
<td>4716</td>
<td>54</td>
<td>1.145</td>
</tr>
<tr>
<td>TCP</td>
<td>4978</td>
<td>49</td>
<td>0.984</td>
</tr>
<tr>
<td>UDP</td>
<td>4990</td>
<td>62</td>
<td>1.242</td>
</tr>
</tbody>
</table>

### High latency, Low bandwidth (hLhB) scenario:
It is a typical existing scenario in the Internet. Figure 4 shows the average delay. The average delay of TCP traffic is observed to be in the range of 260-280 ms whereas the delay of UDP and SCTP is observed to be varying between 220-240 ms. The initial peaks in TCP and SCTP, and the initial low delay in UDP may be explained as in the previous (hLhB) scenario. The intermediate high peaks in TCP and SCTP may be attributed to the congestion; due to congestion in the bottleneck link multiple packets get delayed resulting in a momentary increase (a peak) in the average delay. It triggers congestion control mechanism, which reduces throughput in these protocols.

<table>
<thead>
<tr>
<th>Protocol</th>
<th>Packets Sent</th>
<th>No. of packets lost</th>
<th>% loss</th>
</tr>
</thead>
<tbody>
<tr>
<td>SCTP</td>
<td>4721</td>
<td>52</td>
<td>1.101</td>
</tr>
<tr>
<td>TCP</td>
<td>5009</td>
<td>55</td>
<td>1.098</td>
</tr>
<tr>
<td>UDP</td>
<td>4990</td>
<td>50</td>
<td>1.002</td>
</tr>
</tbody>
</table>

Figure 5 shows the jitter. The jitter may be explained as in the previous (hLhB) scenario. However, jitter in TCP is comparatively higher. In this case, packet loss in the UDP is least (refer, Table 3).

### Low latency, High bandwidth (lLhB) scenario:
It is a typical scenario which exists in the backbone of Internet. Hence, it is the most desirable scenario for any study of the data transmission on a network. The delay shows an expected behavior by having very low end to end delay compared to other scenarios. The average delay of VoIP over TCP is observed around 180-190 ms while the average delay of VoIP over SCTP and UDP both is observed around 140-160 ms (refer, Figure 6). The SCTP and UDP are comparable to each other. The initial high delay in SCTP is due to the initial 4-way handshake connection establishment. A low delay in SCTP and UDP favors them for efficient packet transmission.
Table 4. Packet loss in lLhB scenario

<table>
<thead>
<tr>
<th>Protocol</th>
<th>Packets Sent</th>
<th>No. of packets lost</th>
<th>% loss</th>
</tr>
</thead>
<tbody>
<tr>
<td>SCTP</td>
<td>5041</td>
<td>53</td>
<td>1.051</td>
</tr>
<tr>
<td>TCP</td>
<td>5025</td>
<td>53</td>
<td>1.055</td>
</tr>
<tr>
<td>UDP</td>
<td>4990</td>
<td>57</td>
<td>1.142</td>
</tr>
</tbody>
</table>

Figure 7 shows the jitter. As expected, the behavior of UDP is better as its delay of all the packets is very similar, hence jitter is negligible. Since TCP is having high variation in delay because of its congestion control mechanisms [17], its jitter is high compared to that of SCTP and UDP. In this scenario, behavior of SCTP is same as that of the UDP. Though, in this case also UDP performs better in delay and jitter, like hLhB, we observe comparatively more packet loss (refer, Table 3). However, it is well within the limit to cause any noticeable degradation in the call quality.

5. Conclusion

Our study supports statistically the superiority of UDP over TCP and SCTP in all the scenarios. The good performance of UDP in VoIP applications makes it a preferred transport layer protocol to carry voice packets from source to destination, e.g., Skype [14] uses UDP. However, it is likely that SCTP may perform better with some modifications/extensions in the protocol as, we observe, its performance is comparable to UDP in most of the cases. We are working towards this goal as such modified SCTP will be more promising because it overcomes the shortcomings of both UDP, e.g., traffic control mechanism, and TCP, e.g., head of line blocking.

References