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The Effect of CJM System on IP Packet Size in the Transmission of Real Time Applications

[1]*Ikram Ud Din,* [2]*Saeed Mahfooz,* [2]*Huma Javed and* [1]*Muhammad Zeeshan*

[1]Department of Information Technology, University of Haripur, Pakistan
[2]Department of Computer Science, University of Peshawar, Pakistan

Abstract: This study examines the effect of jitter and end-to-end delay happen in the transmission of real time applications. The study analyzes the performance during sending and receiving of voice and video packets. The proposed technique is applied on the buffer at the receiver side which divides the incoming packets into same size chunks. The end-to-end delay and jitter is high when the packet size is constant (576 bytes), but the performance is significantly improved if the packet size is reduced from 576 to 256 bytes. Jitter and end-to-end delay are the performance metrics used in this study. The simulation results show that both stable video and voice quality and transmission efficiency is achieved.

Key words: CJM · Jitter · Delay · IP Packet · Buffer

INTRODUCTION

Voice over Internet Protocol (VoIP) is the popular and most widely used service of the Internet. VoIP allows the users to communicate freely at low cost with each other. A variety of network impairments such as packet loss, end-to-end delay and jitter severely degrade network quality. The existing Internet service cannot satisfy the required quality of service (QoS) of promising real time applications. Composition of several building blocks makes VoIP application. Voice or video signals are sampled periodically by encoder at sender side and then they are decoded at the destination end. Prior to play these packets, they are stored in the buffer for a while. In playout buffer, the voice or video packets are enforced to be decoded at the same interval at which they were encoded at the sender side. Jitter and delay are acute and critical issues for VoIP applications. If a packet is not received in its assigned time then it is assumed to be dropped [1].

Related Work: The transport service provided by the IP networks is not reliable and the QoS can never be guaranteed. Due to buffer overflow in routers and/or switches, packets can be lost or discarded. The packets can also be discarded due to tremendous bit errors and failure to the cyclic redundancy check (CRC) at the data link layer. For end-to-end recovery from such packet losses, forward error correction (FEC) has normally been proposed. However, the use of FEC schemes in voice or video transmission by end nodes increases packet loss. This loss rate happens because of the additional loads resulting from transmission of redundant packets [2].

The IP network comprises of switches and routers that forward and route IP packets from source to destination. Within these switches or routers packets can be lost or delayed. The delay and loss of IP packets is a problem that needs to be minimized because it causes the quality loss at the receiver end. In the context of video or voice over IP, the jitter is mentioned, which is the variation of delay among the arrived packets [3-6].

For the design of global scale distributed systems [7], the characteristics of the Internet delay space are important to be understood. Zhang and his co-authors [7] have analyzed the delay spaces among different networks and enumerated the major properties that are important for the design of distributed system. They claim that their derived model preserves the important properties of the Internet delay space that the existing models do not capture, but they do not mention jitter which is the critical issue in both voice and video transmission.

Corresponding Author: Ikram Ud Din, Department of Information Technology, University of Haripur, Pakistan.

MATERIALS AND METHODS

In this work, a network is established between two cities of Pakistan i.e., Lahore and Karachi (as shown in Fig. 1). The network has two clients and two servers i.e., video and VoIP, respectively. The distribution of servers and clients is such that the servers are located at site Lahore (in this case) and clients at the other site (say Karachi).

In this study, the two networks are tested using OPNET Modeler 14.0, which are sending and/or receiving both video and voice packets of different sizes, such as 576 and 256 bytes. The scenario was simulated for 10 minutes and the networks were named as Packet_Size_256_Bytes and Packet_Size_576_Bytes.

End-To-End Delay: End-to-end delay means the total time taken by a packet to reach from source to the final destination [8]. The following formula demonstrates the concept of end-to-end delay:

$$Delay = \sum_{R=1}^{n} (D_t + D_{Enc+Dec} + D_p)$$

Here,

Delay = End-to-end delay
R = Number of routers
D_t = Transmission delay
$D_{enc+Dec}$ = Encoding and/or decoding delay
D_p = Propagation delay.

According to the above formula if there are 10 routers in a network and transmission delay is 50ms, encoding and/or decoding delay is 40ms and propagation delay is 25ms, then the end-to-end delay would be:

$$Delay = \sum_{R=1}^{10} (D_{50} + D_{40} + D_{25})$$

This is equal to 115ms, i.e., 50+40+25=115.

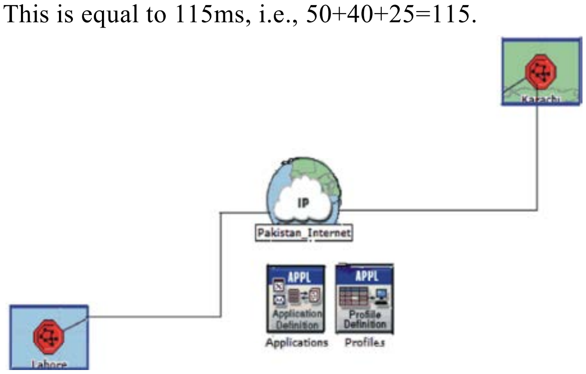

Fig. 1: Structure of the network

In this case we have 10 routers, so if we want to find delay between two routers, then it will become:

$$Delay = \sum_{R=1}^{10} (D_{50} + D_{40} + D_{25})/10 = 11.5$$

In the proposed technique, a buffer is introduced at the receiver end which combines packets in the form of chunks and stores them in the buffer for a short period of time. With this storing of packets in the buffer, the jitter and delay are minimized and better voice quality is achieved as shown in the results' section of this paper.

Jitter: In computer networks, jitter is known as the variations in delay of packets received. In evaluation of network performance, jitter is a fundamental quality of service factor. It is one of the significant issues in packet based network for real time applications [9]. The variation of interpacket delay or jitter is one of the principal factors that disturbs voice quality [10]. Jitter plays a vital role for the measurement of Quality of Service of real time applications [11-14].

QoS requirements for voice are the following:

- Latency is less than or equal to 150 ms
- Jitter is less than or equal to 30 ms
- Loss is less than 1%

QoS requirements for videoconferencing are the following:

- Latency is equal to 150 ms
- Jitter is equal to 30 ms
- Loss is equal to 1% [15].

RESULTS AND DISCUSSION

The main focus of this study is to measure the throughput, delay and jitter in the network with different sizes of IP packets.

As in the earlier publication [16], in this section, the scenario is tested for IP packet sizes of 576 and 256 bytes. Fig. 2 and 3 show the end-to-end delay in video and voice traffic, respectively. Fig. 4 and 5 show the variations in delay (jitter) in video and voice packets, respectively. The X-axis of every diagram shows the simulation time and the Y-axis shows the value of end-to-end delay in Fig. 2 and 3, while it shows the value of jitter in seconds in Fig. 4 and 5.

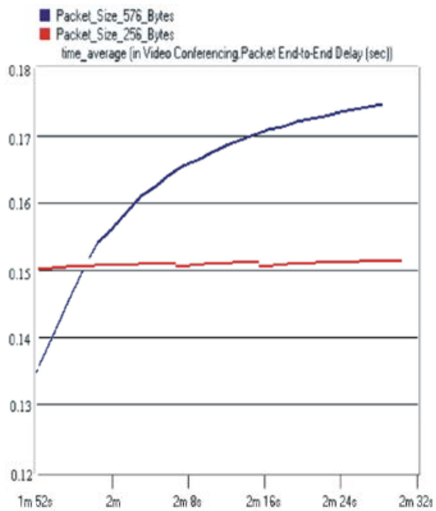

Fig. 2: End-to-end delay in video traffic

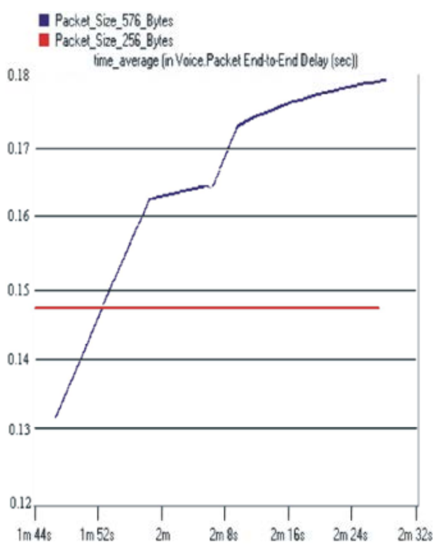

Fig. 3: End-to-end delay in voice traffic

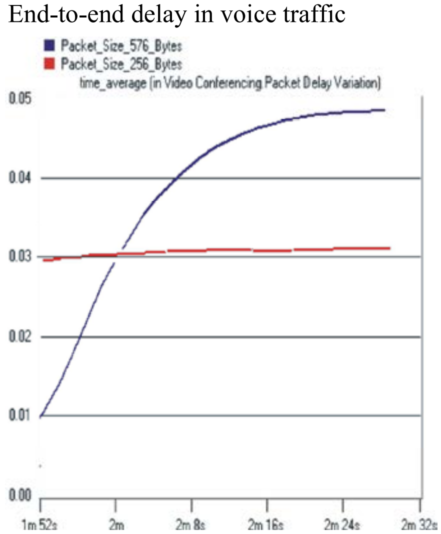

Fig. 4: Jitter in video traffic

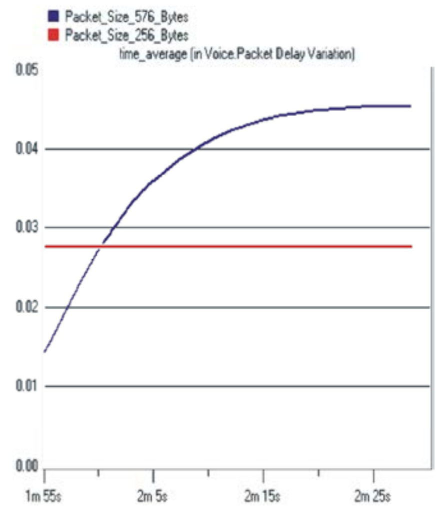

Fig. 5: Jitter in voice traffic

The ITU-T standard for voice jitter and delay is less than or equal to 30ms and 150ms, respectively and for video traffic the maximum delay and jitter is equal to 50ms and 30ms, respectively. Fig. 2 shows maximum delay for video traffic for the network "Packet_Size_576_Bytes" which is almost 170ms and for the network "Packet_Size_256_Bytes", it is equal to 150ms. The jitter shown in Fig. 4 for video traffic is equal to 50ms for the network "Packet_Size_576_Bytes", while it is equal to 30ms for the network "Packet_Size_256_Bytes", which is requirement for the video traffic.

Similarly, for voice traffic, the maximum delay is equal to 180ms in the network "Packet_Size_576_Bytes" and it is less than 150ms in the network "Packet_Size_256_Bytes" as shown in Fig. 3. The jitter shown in Fig. 5 for voice traffic in the network "Packet_Size_576_Bytes" is greater than 40ms, while it is less than 30ms in the network "Packet_Size_256_Bytes" which is requirement for the voice traffic. The obtained results show that the performance of voice and video over IP can be largely improved by applying the Chunk-based Jitter Management (CJM) algorithm on the receiver's buffer [16].

CONCLUSION

In this paper, a network is tested with a scalable and fast technique using OPNET Modeler 14.0. Specifically, a Chunk-based Jitter Management algorithm is proposed to minimize the problem of jitter and end-to-end delay in voice and video transmission. Efficiency of the proposed system is proved through extensive simulations. The system improves the performance of IP networks in the transmission of real time applications. In the future, we will study how to extend the framework for Wireless networks.

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