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Performance Evaluation of SCTP and TCP in Voice Communication

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Abstract: The number of Internet users is increasing quickly with the transmission of voice over IP. Therefore provision of fast transmission without jitter and delay is the fundamental aspect of voice communication. This study evaluates the performance of SCTP and TCP in transmission of voice packets. Generally, when packets are arrived at destination, these are probably affected by jitter. Many researchers have applied different techniques on them in the buffer to minimize jitter. In this study, an algorithm is applied on packets in the buffer for end-to-end delay and jitter management. The paper presents two topologies to analyze the performance of TCP and SCTP in case of delay and jitter. The functionality of these protocols is simulated and tested in NS2. The results show a good increase in throughput and delay and jitter minimization.

Key words: SCTP · TCP · CJM · Jitter · Voice

INTRODUCTION

Primary role of the transport layer is to provide end-to-end delivery of messages between two or more applications running on different machines [1, 2]. For the last three decades, end users and applications use one of the two protocols: User Datagram Protocol (UDP) and Transmission Control Protocol (TCP). The Internet Engineering Task Force (IETF) approved a new protocol in October 2000 as a proposed standard to extend functionality of the transport layer. This protocol is called Stream Control Transmission Protocols (SCTP) [3]. An SCTP association, which is called connection in discussions of TCP, can involve multiple streams (Fig. 1) and employ multiple addresses at each end. When one interface fails then the next interface is functional without interrupting the communication. This property is called Multihoming [4-9] which is a salient feature for voice transmission.

Multistreaming [7] is another feature of SCTP that permits data from the upper layer application to be multiplexed onto one association. If a packet belonging to a particular stream is lost, then from that stream, packets following the lost one will be stored in the receiver's buffer until the lost packet is retransmitted from the source [9].

In this paper, the performance of TCP and SCTP is measured in case of voice transmission. The sender sends voice packets to the destination using both TCP and SCTP. Voice packets are then stored in the buffer for a while; CJM algorithm is applied on these packets and are then played as shown in Fig 2. The result section of this paper shows that when the CJM algorithm is applied on packets in the buffer, jitter and end-to-end delay are significantly reduced.

Related Work: Performance of the Transport Layer Protocols plays a vital role in offering better communication between two end devices. The selection of a transport protocol depends upon the needs of the application. TCP and UDP are the two governing protocols of the transport layer and for peer to peer data communication, TCP is considered as a prime transport protocol. It is being used by popular suite of applications because of its qualities, for example, congestion control, flow control and reliability [10]. To improve its efficiency, new implementations and improvements have been done. Some characteristics of TCP, i.e. Byte-oriented delivery and strict order mechanisms became weaknesses for some tasks, for example, VoIP applications [11, 12].

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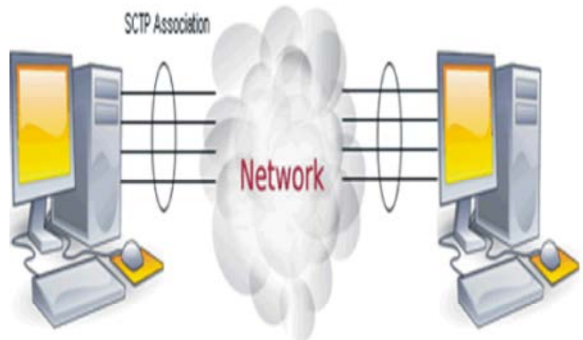

Fig. 1: Transmission using SCTP

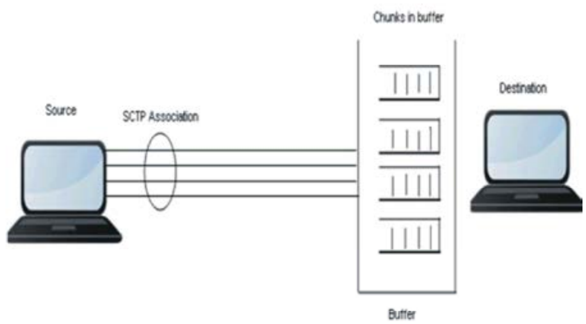

Fig. 2: Architecture of the proposed system

Significant work has been done for the performance improvement of SCTP in different fields, for example, multimedia traffic, satellite communications, broadband networks and web applications etc. in which SCTP offered better results than TCP. Among the current research study, Boussen S, *et al* [13] simulated a WiFi network in NS-2 and showed through their results that the performance of SCTP is far better than UDP and TCP in case of throughput and end-to-end delay.

Tran Cong Hung, *et al* [14] studied the performance of SCTP at transport layer and proved through their results that TCP and UDP can be replaced by SCTP because of its improved performance and reachability especially in case of nasty network conditions.

In various operating systems, the implementation of SCTP is in improving stage and a limited number of scenarios have been practically implemented. This is mainly due to lack of SCTP supporting networking devices.

In this paper, we have also examined the performance of SCTP using NS-2 simulator allied with CJM algorithm, which largely reduces end-to-end delay and improves throughput.

RESULTS AND DISCUSSION

In this section we evaluate the impact of CJM technique for voice transmission through TCP and SCTP. NS-2 is used for testing the performance of TCP and SCTP because it has built-in support for SCTP. The simulated homogeneous network is shown in Fig 1, where all available associations may experience different propagation delays. First of all, we compare the throughput performance of TCP and SCTP over homogeneous network. From the results, we can see that the number of packets received through SCTP is 2400 in Fig 3, while it is 2200 through TCP, which shows a great improvement in packets. In our simulated scenario, the buffer can store 100 chunks and each chunk consists of 50 packets, therefore, the buffer is set to 3MB. The 3 MB buffer is fixed because the maximum size of a packet is 576 bytes and one chunk consists of 50 packets, so 576*50=28800 bytes and the number of chunks that can be stored are 100, therefore, 28800*100=2880000 bytes which are approximately 2.8 MB.

For the measurement of SCTP and TCP's performance, voice packets for 15 minutes are sent from one node to the other. The proposed technique plays chunks with the shortest delay. The maximum delay in TCP (shown in Fig 4) is 180ms, while it is 150ms in case of SCTP, which is the standard ITU-T's QoS (Quality of Service) requirement for voice transmission. Jitter in both TCP and SCTP traffic is shown in Fig 5, in which the standard ITU-T QoS requirement for voice traffic is achieved, which is 30ms. But in case of TCP, the jitter is almost 35ms that can deteriorate voice quality.

Algorithm: The informal description of the proposed chunk-based jitter management algorithm [15, 16] in its pseudocodal form is given below:

- [Reading Packets from network]
- Read packets from network and store in the buffer.
- [Chunking]
- Group the packets into same size chunks.
- [PlayOut Chunks]
- Read chunks until buffer is empty
- While (Buffer is not Empty)
- [Read Chunk]
- Read chunk from the buffer.
- For (PacketCounter=1; PacketCounter <= ChunkSize; PacketCounter++)

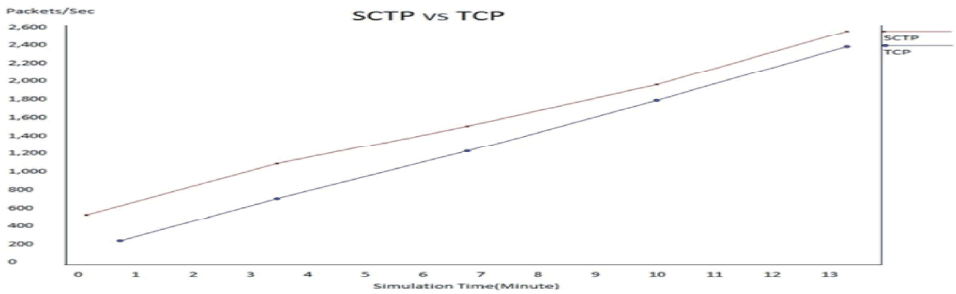

Fig. 3: Number of Packets Received per second

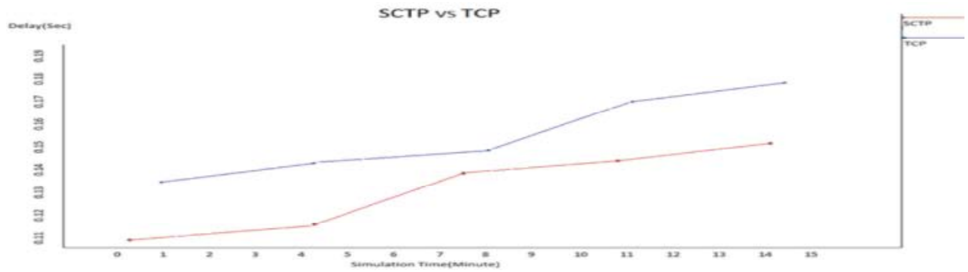

Fig. 4: Voice delay in SCTP Packets and TCP Segments

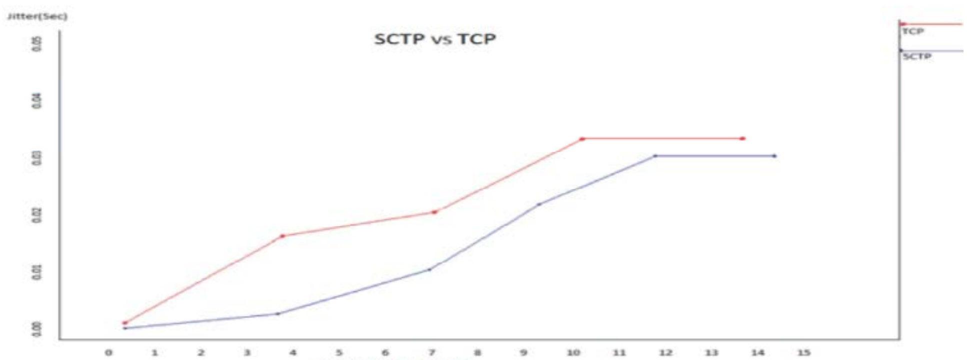

Fig. 5: Jitter in SCTP Packets and TCP Segments

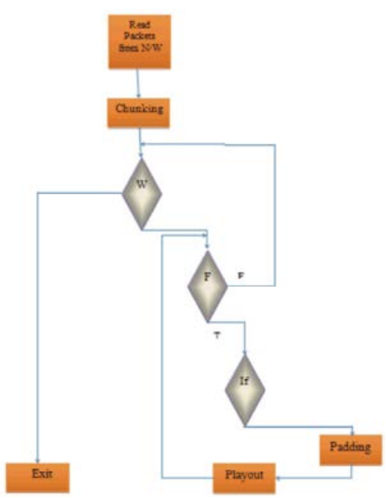

Fig. 6: Flowchart of the System

- [PlayOut Packets]
- StreamOut packets
- [Padding]
- Check chunk size
- If (ChunkSize<MaxValue)
- Append extra bits to make chunk filled
- End of If
- End of For
- End of While
- [Stop]
- Exit

CONCLUSION

In this paper, we compare the performance of TCP and SCTP during voice transmission. At sender side, voice packets are transmitted and then they are stored in the buffer at the receiver side. When packets arrive and are stored in the buffer, then the Chunk-based Jitter Management algorithm is applied on these packets. The obtained results show that the performance of voice traffic can be largely improved by applying the Chunk-based Jitter Management algorithm on the receiver's buffer.

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