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SCTP and FEC based Loss Recovery Technique for VoIP

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Abstract: VoIP (Voice over Internet Protocol) has become one of the most attractive and important applications running and also one of the most emerging technologies in today's world. Different techniques have been proposed to reduce the packet loss and recover the lost audio data packets during the transmission in the network. In VoIP, Stream Control Transmission Protocol (SCTP) is the responsible for the packet transmission and for the retransmission of the lost data packets. But it does not deal with link errors and consumes large buffer size at the receiver endpoint for estimating the packet loss. In this paper, we propose to design a cross-layer architecture for VoIP networks. In this architecture, Forward Error Correction (FEC) technique is applied in SCTP transmission to ensure the reliability of data transmission. While transmitting VoIP packets, FEC technique gets executed at every intermediate node of the network. When the packets reach the receiver endpoint, the packet loss estimator estimates the packet loss and amount of redundancy to be added in FEC technique. By simulation results, we show that the proposed architecture reduces the packet loss and delay and improves the throughput..

Keywords: VOIP, SCTP, FEC, Cross Layer Architecture, Data Transmission

I. INTRODUCTION

1.1 Voice over Internet Protocol (VoIP)

In the last few years, VoIP (Voice over Internet Protocol) has become one of the most attractive and important applications running and also one of the most emerging technologies in today's world. This technology enables voice communication through the Internet. VoIP compress the audio data into data packets which can be sent efficiently over the networks and converted back into the audio data at the receiving end. VoIP sends this audio information in digital form in discrete packets rather than by using the habitual circuit-committed protocols of the Public Switched Telephone Network (PSTN). This technology uses the real-time protocol (RTP) to help ensure that packets get delivered in a timely way.

VoIP is an internet telephony which offers wide range of benefits to talk with each other freely at low rates which allows for the calls, local, long distance, and international over the Internet. VoIP can achieve a greater efficiency since the data packets in the network are directed to their destination by diverse routes, sharing the same facilities most efficiently. VoIP are lower in cost since IP systems will offer a more economical means for providing communication connections which is one of the sources of concern

VoIP networks differ from conventional telephone networks in that voice quality is affected by a variety of network impairments such as delay, packet loss, jitter, echo, network security and throughput. But the major challenge of the VoIP network is maintaining quality i.e. packet loss which is a serious and critical issue for voice over internet protocol applications.[1][2][3]

Issues of VoIP [1][3]

- Delay in packet transmission from sender to receiver
- Packets arriving too late at the receiver side
- Heavy load on the network
- Congestion of routers and gateways.
- The variations in packet inter arrival time create difference between when the packet is expected and when it is actually received is jitter.
- The loss of voice packets from sender to receiver.

1.2 Packet Loss and Recovery Techniques in VoIP [4]

Three types of packet loss can occur in VoIP networks: Random, Burst and real-network loss.

When the packet loss is independent from each other, it is said be a random loss. In other words, it is the loss of packets randomly. In burst loss, multiple consecutive packets will be lost for a fixed period of time. In real-network loss, significant packets will be lost in unidirectional transmission path. (ie) from one edge to another edge.

To recover from the losses, the following recovery techniques are used in VoIP.

Plain Delivery: This technique simply bundles each block of encoded audio data packets into an IP packet and transmits

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it. It does not provide any sender-based effort to improve audio quality when packet loss occurs. The advantage of this technique is it is more common than any other delivery technique in VoIP solutions.

Interleaving: This technique attempts to reduce the degradation of perceptual audio quality by distributing lost data into several small gaps instead of having one large gap of lost data. This technique requires the same bandwidth utilization that plain delivery uses since it does not transmit additional information.

Forward Error Correction: FEC is a sender-based technique for mitigating the undesired effects of packet loss. This works by transmitting redundant packets for error correction. Reed–Solomon encoding scheme and the parity-encoding scheme are the different variants of the FEC technique.

Retransmission: This technique is used only at the explicit request of the recipient, so it requires a round trip time that inherently induces a large end-to-end delay. The receiving endpoint implements a loss-detection algorithm to detect lost packets; if any packets are lost this technique will resends the lost data packets upon request by the recipient.

1.3. Problem Identification and Proposed Solution

In [5], cross layer architecture which consists of different components to reduce the packet loss and recover the lost audio data packets during the transmission in the network.

Here in this architecture, the SCTP (Stream Control Transmission Protocol) is the responsible for the packet transmission and for the retransmission of the lost data packets. When the packets are transmitted, the packets loss occurs due to two reasons one is the packets error due to the bit rate and the other is due to the excessive delay. This packet loss is estimated through the packet loss estimator. These lost packets are retransmitted through the SCTP.

But in this architecture the receiver endpoint develops a buffer to store the transmitted data packets and then estimate the packet loss and request for the retransmission. This can be a drawback for this architecture since it does not deal with the link error or any method to reduce the packet loss estimation or to reduce the buffer size at the receiver endpoint.So to overcome this drawback we propose a forward error correction technique in the VoIP network, where the data packets taken care at every node of the network.

II. PROPOSED SOLUTION

2.1 Overview and System Design

In this paper we have proposed a technique called "SCTP and FEC based Loss Recovery Technique for VoIP". In this approach we implement the FEC technique in order to check whether the data packets are not affected and the transmission link is proper. Then the data packets are transmitted through the SCTP. While the transmitting of the data packets the FEC technique gets executed at every intermediate node of the network to check if any packets are lost. And when the data packets reach the receiver endpoint the packet loss estimator gets executed and request for the retransmission of the lost packets to the sender. Figure 1 presents the architecture diagram of the system design.

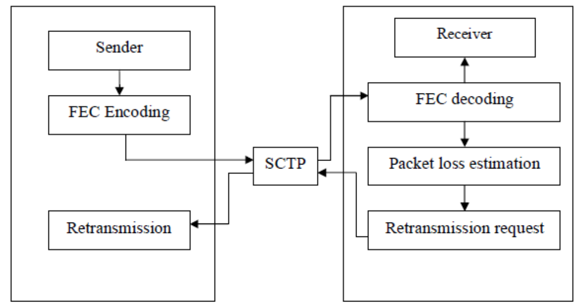

Fig.1. System Design

2.2. Stream Control Transmission Protocol (SCTP)

SCTP is a message based end-to-end connection concerned with transport layer protocol approved by the Internet Engineering Task Force (IETF). SCTP is similar to TCP but has some extra abilities. SCTP has acquired many of the core features of TCP like congestion control and retransmission. Applications can benefit from SCTP features allowing higher performance and reliability than other protocols.

An SCTP packet consists of one or more concatenated building block called as chunks that will be either control or data. In order to provide reliability and handle the congestion, each data chunk in an association is allocated a unique Transmission Sequence Number (TSN), which is similar in function to sequence numbers in TCP. Since SCTP is message-oriented and chunks area unit atomic, TSNs area unit associated solely with chunks of knowledge, as opposition a TCP computer memory unit stream that associates a sequence range with every computer memory unit of knowledge.

SCTP maintains individual parameters for each IP address in an association which can be used to assess the quality and reliability of the network link.

SACK chunks can be bundled with data being transmitted from the receiver to sender. Since full duplex VoIP calls are being used, each endpoint will be transmitting packets at short intervals. As every second packet received must be acknowledged this greatly reduces any potential delay in the SACK chunk.

Partial Reliable SCTP (PR-SCTP) extension [13] allows SCTP to provide UDP like packet delivery at the same time, keeping the TCP friendly congestion control mechanisms of SCTP. It eliminates the problem of head of line blocking caused by the basic SCTP. Thus it is more suitable for the transport of real time applications such as VoIP. PR-SCTP ensures the stability of transport layer for delivering each

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message. It provides varying levels of reliability to upper layer protocols.

It allows the sender to specify a TTL value for each message transmitted. The TTL value indicates the duration for transmitting the message by the sender. The message will be dropped on expiration of this value. As individual packet losses will not affect the overall quality of VoIP, this mechanism is more suitable for VoIP transmissions. However, selecting an optimum TTL value is crucial to the performance of VoIP over SCTP.

The multistreaming feature of SCTP permits freelance streams of knowledge to be transmitted across one association with no reliance on the delivery order of packets in different streams. Multistreaming is used in SCTP to solve the TCP problem of Head of Line (HOL) blocking that arises from TCPs strict byte ordered delivery. Figure 2 shows the SCTP based multistreaming process.

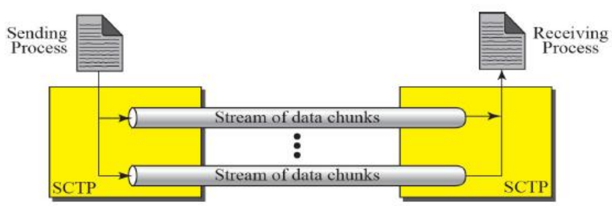

Fig.2. Multistreaming using SCTP

2.3 Forward Error Correction

If delay is a constraint then Forward Error Correction (FEC) is a better approach than Automatic Repeat Request (ARQ). FEC can be either media-specific or media-independent. The error correction code is used by media dependent FEC to get more bits in the data stream that can be used to recover lost packets. Media-independent FEC technique makes use of parity coding which performs an exclusive-OR operation over a block of packets to get an additional payload that can be used in case a single packet is lost within the protected block. Packet attached with both errors and packet loss is known as packet erasure. For every N packets sent the packet-level FEC works by adding another error-recovery packet to it. This FEC packet has the information which can be used to rebuild any single packet within the group of N. during transferring of data if any one N packet is lost then the FEC packet is used on the far end to reconstitute the lost packet. This will avoid the process of resending the lost packet. This will reduce the application response time and improves network efficiency. Encoded Packet P is given by

$$P = G \times M - - - (1)$$

Here M is the information packets and G represents the generation matrix.

The vandermonde matrix G is augmented with the identity matrix (I) and designed matrix (v) forms the r×c generation matrix code which is given by

$$G = \begin{bmatrix} I_{c\times c} \\ \dots \\ V_{r\times c} \end{bmatrix} - - - - (2)$$

FEC is good on high-rate aggregate flow, than on individual flows. And also an ideal FEC will adapt variable traffic changes in the network.

FEC Erasure Recovery and Error Correction Technique

All the coded packets P are received at the receiver and arranged row-wise in a matrix. The received packet R is expressed as,

$$R = E + P \qquad (3)$$

where E is the error packets.

Identify whether the packets are error and find the error locations and their values within the error packets. In case of binary codes, the error values are not needed since by knowing the location one just flips them. Syndrome decoding that depends only on the error packets and the parity check matrix is used.

Erasure recovery and error correction techniques using one code word is as follows:

- First, set the erased positions of the received codeword to zero and normally decode the resulting codeword.
- Measure the Hamming distance between the codeword filed with zeros and the decoded codeword.
- Now, set the erased positions of the received codeword to one and decode the resulting codeword normally.
- Measure the Hamming distance between the codeword filed with ones and the decoded codeword.
- Choose the decoded codeword with the smallest hamming distance.

This technique corrects the single erroneous packet among the group of n received packet.

For erasure recovery technique, assuming no errors, at the receiver side, if there are L lost information packets (L≤ k), then the missing information packets (ML) is expressed as:

$$ML = (G^L)^{-1}.(P)^L \qquad (4)$$

where $(.)^{-1}$ represents the inverse of a matrix,

G^L is an L x L sub matrix of G (where G = n* k) that remains after proper substitution of the received packet

P^L is the received parity packets.

2.4 Packet loss estimation

The packet loss contains two components like packet error due to bit error and the packet loss due to excessive delay. If D_{ete} is a maximum end to end delay and packet loss rate denoted by RTP then E^{RTP} represents the packet loss without error discovery if the D_{ete} closes to the one way delay. If D_{ete} is large enough to allow error recovery through packet retransmission then E^{RTP} denotes the residential packet error when packet retransmission fails. And the VoIP application drops a late arrival packet

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according to the timestamp of the voice packets and computes the late arrival packet loss of the voice session. The packet delay distribution is a statistical representation of network delays observed by the packets in the stream. End to end delay distribution in a wireless network will follows a pareto distribution. Hence the pareto distribution is used in AQP to predict retransmission delay and packet loss rate.

The notations used in this approach are as described below;

D_{ete} is the Maximum end-to-end delay budget

d_i is the Network delay of the i^{th} packet

d_i^n is the network delay

v_i is the delay variance

d_i^b is the introduced bufer delay

F_d is the cumulative function of the pareto distribution of the packet delay.

F'_d is the cumulative function of the pareto distribution of the retransmission packet delay.

ReTx is the retransmission request flag

D_i is the estimated local optimal buffer delay which cannot tolerate retransmission delay

D'_i is the estimated local optimal buffer delay which can tolerate retransmission delay

E^{RTP} is the packet loss rate in RTP.

P_l is the packet loss rate of late arrival

P_e is the packet error rate

P_t is the total packet loss rate of a VoIP session without retransmission estimated algorithm

P'_t is the total packet loss rate of a VoIP session with retransmission estimated algorithm

P_{fr} is the probability of retransmission failure

d_i^c is the mean of packet loss/error detection time.

d_i^n is the mean delay, v_i is the delay variance of packet i, d_i is the network delay of the packet I and α is a smoothing factor. If the value of α is low then the mean delay estimator will react quickly or it will react slowly to delay fluctuation.

If d_i^b is the buffer delay then the playout delay is equal to $d_i^n + d_i^b$. The packet i is considered as lost when $d_i > d_i^n + d_i^b$. If F_d is the cumulative function of the pareto distribution of the packet delay, then the packet loss rate of a session without error recovery P_l, is given by

$$P_l = 1 - F_d\left(d_i^n + d_i^b\right) - - - -(4)$$

Based on P_l, the total packet loss rate when the lost packets are not retransmitted is given by;

$$P_l = 1 - F_d(d_i^n + D_i) + P_e - - - - - -(5)$$

Here P_e is the packet error rate and D_i is the estimated buffer delay when packets in error are not retransmitted.

Retransmission Enabling Mode

The voice quality of VoIP can be improved by extending the buffer delay D_i to a larger value D'_i such that the receiver can tolerate a larger packet delay and the packets in error may be easily recovered by retransmission. The voice quality depends both on the end-to-end delay and the packet loss rate. Hence we have to evaluate that the increasing D_i can effectively increase voice quality before a new buffer delay is used.

Adaptive QoS Playout algorithm (AQP) adjusts the buffering delay for each talk spurt to keep the service quality of VoIP at the best condition. The packet loss estimation modules will first verifies whether a lost packet can be retransmitted one more time else it will just tunes current playback delay. AQP will select the better buffering delay to improve voice quality by evaluating the total packet loss rate P'_T with all the possible strategies. The total packet loss rate P'_T is calculated as given below

$$P'_T = \begin{cases} \left(1 - F_d(d_i^n + D_i^{'})\right) + P_e \times P_{fr}(d_i^n + D^{'}), Retx = disable. \\ 1 - F_d(d_i^n + D_i^{'}), \;\; Retx = enable \end{cases} - - - (6)$$

$$P_{fr} = 1 - F'_d(d_i^n + D_i^{'}), \qquad (7)$$

Here P_{fr} is the probability of failed retransmission and F'_d is the cumulative function of the retransmission packet delay, which also follows a pareto distribution.

The mean and standard deviation of F'_d is assumed as

$$mean = 3 \times d_i^n + d_i^c$$
$$standard\ deviation = 3 \times v_i$$

Here d_i^c is the mean time to detect a packet loss and depends on the error or loss detection strategy adopted in the receiver.

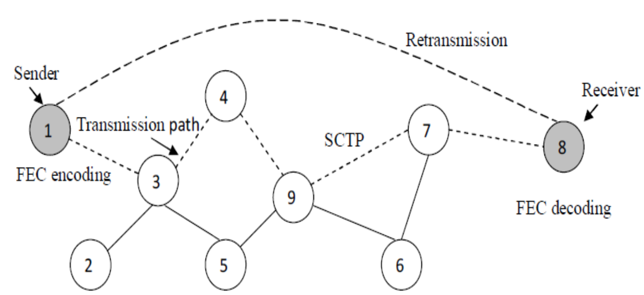

Fig.3. SCTP and FEC based Loss Recovery Technique

Over all Algorithm

1. Sender encodes VoIP data packets using FEC technique.
2. The encoded packets are transmitted through SCTP connection to the destination.
3. At the destination, the packets are decoded.
4. The packet loss estimator estimates the amount of packets lost.

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5. If the packets can be retransmitted within the playback delay time,
5.1 Retransmission request is made.
Else
5.2 AQP adjusts the buffering delay
5.3 The amount of redundancy is determined for FEC based on estimated loss
6. Sender again encodes the packets by adding the estimated amount of redundancy.

III. SIMULATION RESULTS

3.1Simulation Model and Parameters

This section deals with the experimental performance evaluation of our algorithm through simulations. In order to test our technique, NS-2 simulator [15] is used. NS2 is a general-purpose simulation tool that provides discrete event simulation of user defined networks.
We have used the BitTorrent packet-level simulator for P2P networks [14]. A network topology is only used for the packet-level simulator. Based on the assumption that the bottleneck of the network is at the access links of the users and not at the routers, we use a simplified topology in our simulations. We model the network with the help of access and overlay links. Each peer is connected with an asymmetric link to its access router. All access routers are connected directly to each other modeling only an overlay link. This enables us to simulate different upload and download capacities as well as different end-to-end (e2e) delays between different peers.
In the simulation, 11 nodes are used for 30 seconds of simulation time. The simulated traffic is SCTP. The topology is shown in the following figure.

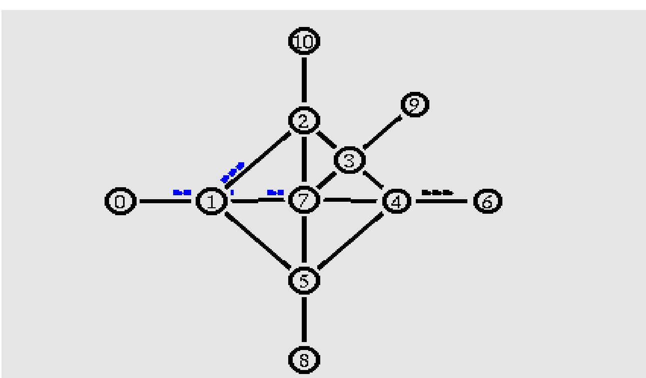

Fig.4. Simulation Topology

The simulation settings and parameters are summarized in table.

No. of Nodes	11
Simulation Time	10,15,20,25 and 30 sec
Traffic Type	SCTP
Packet Size	512
Chunk Size	250,500,750,1000Kb

3.2 Performance Metrics

The proposed SCTP and FEC based Loss Recovery Technique (SCTPFEC) is compared with the standard SCTP technique. The performance is evaluated mainly, according to the following metrics.

- **Packet Delivery Ratio:** It is the ratio between the number of packets received and the number of packets sent.
- **Packet Drop**: It refers the average number of packets dropped during the transmission
- **Throughput**: It is the total number of packets received by the receiver.

A. Based on Chunk Size

In our first experiment we vary the chunk size as 250,500,750 and 1000Kb.

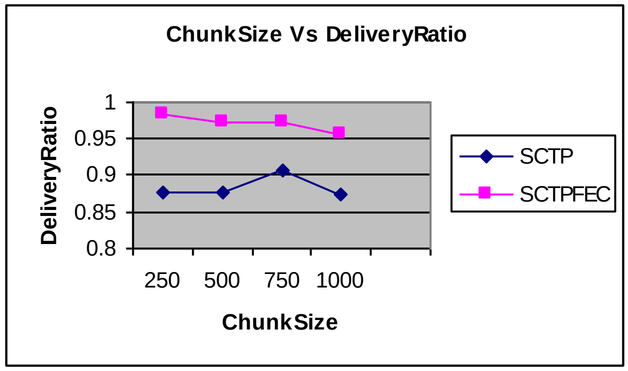

Fig.5. Chunk Size Vs Delivery Ratio

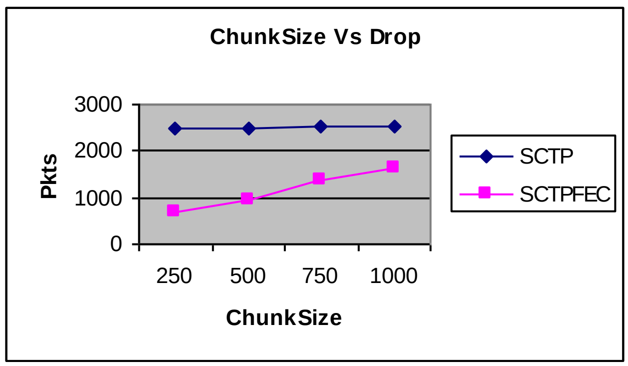

Fig.6. Chunk Size Vs Drop

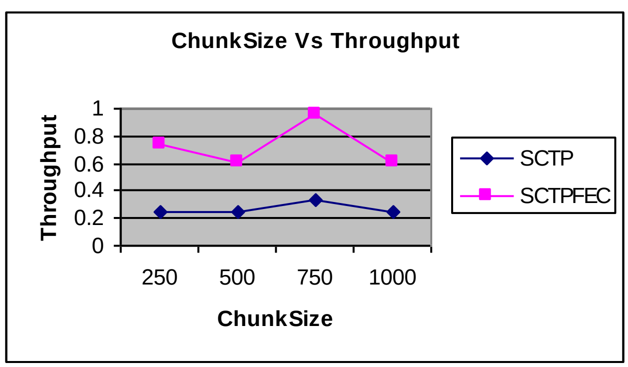

Fig 7: Chunk Size Vs Throughput

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Figure 5 shows the delivery ratio of SCTPFEC and SCTP techniques for different Chunk Size scenario. We can conclude that the delivery ratio of our proposed SCTPFEC approach has 9% of higher than SCTP approach.
Figure 6 shows the drop of SCTPFEC and SCTP techniques for different Chunk Size scenario. We can conclude that the drop of our proposed SCTPFEC approach has 54% of less than SCTP approach.
Figure 7 shows the throughput of SCTPFEC and SCTP techniques for different Chunk Size scenario. We can conclude that the throughput of our proposed SCTPFEC approach has 63% of higher than SCTP approach.

B. Based on Simulation Time

In our second experiment we vary the simulation time as 10,15,20,25 and 30sec.

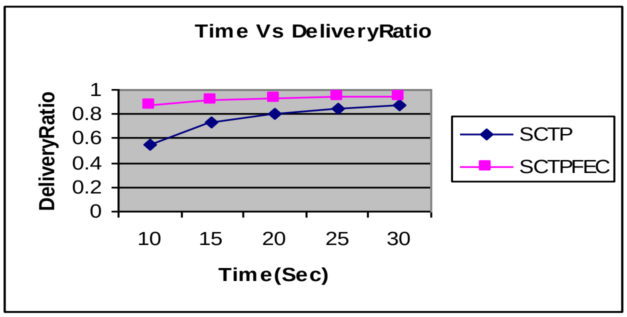

Fig.8. Time Vs Delivery Ratio

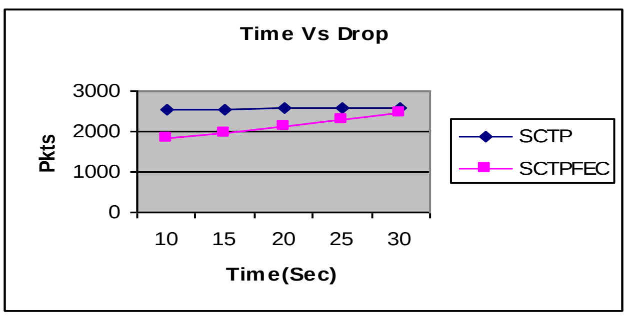

Fig.9. Time Vs Drop

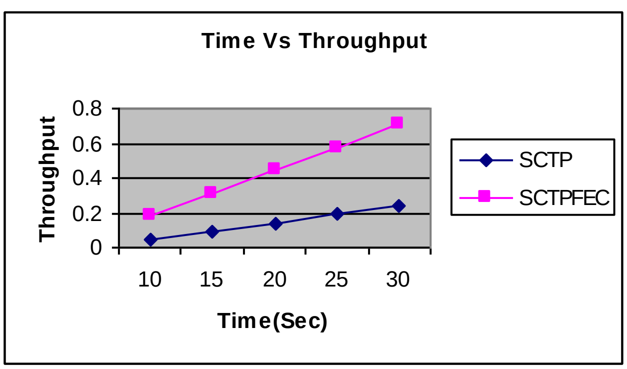

Fig.10. Time Vs Throughput

Figure 8 shows the delivery ratio of SCTPFEC and SCTP techniques for different simulation time scenario. We can conclude that the delivery ratio of our proposed SCTPFEC approach has 18% of higher than SCTP approach.
Figure 9 shows the drop of SCTPFEC and SCTP techniques for different simulation time scenario. We can conclude that the drop of our proposed SCTPFEC approach has 17% of less than SCTP approach.
Figure 10 shows the throughput of SCTPFEC and SCTP techniques for different simulation time scenario. We can conclude that the throughput of our proposed SCTPFEC approach has 70% of higher than SCTP approach.

IV. CONCLUSION

In this paper we have proposed a loss Recovery Technique for VoIP. Here to check that the data packets are not affected and transmission link is proper, FEC technique is applied. Now the data packets are transmitted through the SCTP. During the transmission of the data packets the FEC technique gets executed at every intermediate node of the network to check if any packets are lost. The packet loss estimator gets executed when the data packets reach the receiver endpoint and sends the request to the sender to retransmit the lost packets. By simulation results, we have shown that the proposed architecture reduces the packet loss and delay and improves the throughput.

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BIOGRAPHY

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