

# Efficient Fault Recovery Two Phase Routing for VoIP Applications

J. Faritha Banu and V. Rama Chandran

**Abstract**—The need for real-time multimedia applications like streaming video, VOIP has increased over commercial organizations, reliability and availability has become important issues. Such Networks are prone to failures because of different reasons (e.g. unreliable equipment, software bugs or cable cut). Such faults and failures may affect the operation of LSPs and cause packet loss. VoIP packets are sensitive to packet losses and transmission delay to assure correct communication. This paper proposes Efficient Fault Recovery Two Phase Routing for VoIP Applications and its performance based on forward error correction technique. With this approach packet losses against link failures can be recovered by using Vandermonde Matrix Level FEC mechanism. VoIP packets can be routed in two phases in contrast to the traditional single path routing approach to perform load balancing. Simulation results are reported to show the efficiency of the proposed technique when compared with single path routing for effective VoIP flows.

**Index Terms**—Best effort routing, FEC, load balancing, QoS, VOIP, two phase routing.

## I. INTRODUCTION

VoIP can be obtained on any network that is using IP such as the Internet, intranets and local networks in which digitized voice packets are passed over the IP networks. Existing network infrastructures can be used to carry both data and voice traffic, which is very attractive to new users [1]. The critical problem is performing load balancing and reducing packet losses. Numerous technologies for load balancing in existence have made use of the mechanism of dispersing the traffic over multiple parallel paths and to improve the QoS.

In case of VoIP networks, the preferred QoS and required channel capacity differ depending on used codec or the compression technique [2]. The quality of VoIP applications are based on two main factors: 1. Basic network characteristics such as delay, packet loss rate and jitter. 2. Technologies such as codec type, Packet Loss Concealment (PLC) mechanisms etc., [3]. When designing a network, considerations like application requirements, available budget, quality of service requirements, and downtime ramifications, have to be taken into account. VoIP calls are very receptive to packet losses and needed to be enclosed on transmission delay to guarantee correct communication [4]. VoIP Packet format is given in Fig.1:

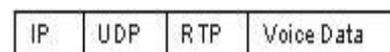


Fig. 1. VoIP packet format.

VoIP packet consists of Internet Protocol (IP) [5] header, User Datagram Protocol (UDP) header, Real Time Transport Protocol (RTP) header and voice data. Internet Packet loss happens due to the fact that the router fails to deliver some packets because its buffer memory is full or bit errors might have occurred during transmission. A router can drop packets or all packets and this depends upon the status of the network, that is whether it is fully congested or not. This packet loss can degrade the performance of voice and video traffic.

Usually Transport Control Protocol (TCP) addresses this kind of problem by sending a retransmission request. However, in real-time applications retransmission is not feasible because UDP is used. This protocol is connectionless and provides best effort service, meaning that packets may be lost, or received in incorrect order due to variable delay. It gives the equal treatment to all traffic types and does not give any guarantee of delivery.

To enhance the QoS and recover from packet losses, Two phase routing [6] combined with forward error correction technique is proposed in this paper. This Two Phase routing scheme guarantees a variety of benefits such as fault tolerance, increased bandwidth, load sharing and QoS. The proposed strategy can recover packet losses and errors against link failures using Vandermonde Matrix Level FEC mechanism. The packet lost due to link failure and overflow of buffers in the intermediate forwarding network nodes are recovered. The rest of this paper is organized as follows. Section II presents the related work. We then point out the proposed system model in Section III. Performance evaluation is reported in Section IV. This paper concludes in Section V.

## II. RELATED WORK

To achieve Fault Recovery and to improve Quality of Service several techniques has been widely used. Simple fault tolerance mechanism sends the packet along all discovered routes. The destination node can successfully receive the packet as long as at least one of the routes does not fail. Obviously, this broadcasting approach is not bandwidth efficient.

Packet Loss Rate often called as Noticeble Loss Rate (NLR) metrics is proposed by IETF, which is the percentage of lost packets with the loss distance smaller then the loss constraint distance. It counts losses of “Close” packets and ignores losses of distant packets. Hanoch Levy et al [7] proposed a model in which packets of a certain session are

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J. Faritha banu is with the Department of Computer Science and Engineering department, Research scholar, Sathyabama University, Chennai, India (e-mail: banujahir@gmail.com).

V. Ramachandran is with now Director of National Institute of Technology, Nagaland,Chumukedima, Dimapur.

dispersed over multiple paths, in contrast to the traditional approach to perform load balancing. They have focused on the network performance by calculating the loss rate and loss burstiness on the applications. They have analyzed NLR metrics for various packet dispersion strategies over memoryless (Bernoulli) loss model or bursty (Gilbert) loss model to minimize the packet loss rate.

There are several receiving end Packet Loss Concealment (PLC) techniques [8] are available as shown in Fig.2. Frame Estimation: where the missing packet is estimated from prior and posterior packets. Frame Substitution: replacing the lost frame by a silent or a default “noisy” frame. Frame stretching: stretches the surrounding packets to cover the missing packet interval. However, if the missing packet contains non-redundant information, like a transient, this will inherently be lost. Frame Delaying (or) Adaptive play out: stretch the packets preceding the missing one to allow more time for the missing packet to arrive. After the arrival of missing packet, succeeding packets must be compressed in order to return to the original play-out rate.

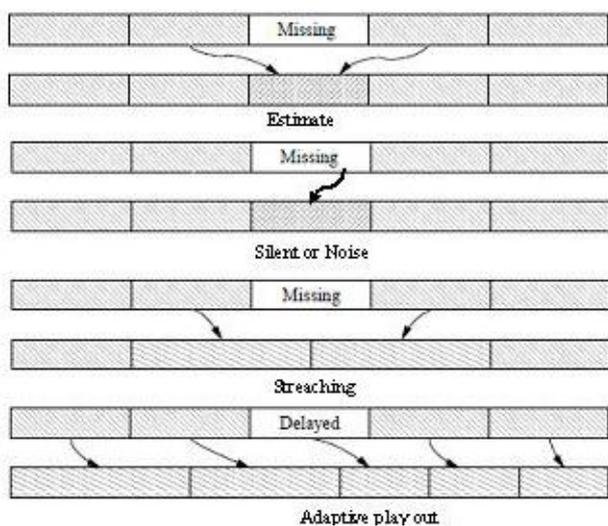


Fig. 2. PLC techniques.

FEC can be either media-specific or media-independent. Media-specific FEC uses error correcting codes in order to produce additional bits in the data stream that can be used to recover lost packets. Media-independent FEC technique uses parity coding that performs an exclusive-OR operation over a block of packets in order to produce an additional payload that can be used in case a single packet is lost within the protected block [9].

Several FEC techniques such as low density parity check (LDPC) and REED Solomon (R-S) codes etc., are used to recover from packet losses [10, 11]. In link recovery, each primary path has a back path or collection of backup path. The backup path protects against all link or node failure along primary path. But failure notification has to propagate from the node that detects failure to the sender of the packets to take path diversity. Fatiha Merazka et al [12] proposed a concealment method based on forward error correction (FEC) to improve speech quality deterioration caused by packet losses for CELP based coders. ITU-T G729 standard speech coder is used to evaluate the speech quality. The perceptual evaluation of speech quality (PESQ) and enhanced modified bark spectral distortion (EMBSD) tests are proposed under

various packet loss conditions to recover the packet losses. This method retrieves only one loss in a group of n packets.

Another Hidden Markov Model (HMM) [13, 14] based adaptive FEC strategy is proposed for interactive streaming real time applications. HMM is used to model and track the evolution of speech signal parameters such as spectral envelope, pitch, voicing, and energy. Within the HMM framework, a Minimum-Mean-Squared Error (MMSE) approach is utilized to provide estimates of the missing speech parameters for both the extrapolation and interpolation procedures of PLC. Discrete time HMM with continuous observations approach is used to model speech evolution where the emission in each state is modeled using a single Gaussian probability density function. Since the PLC algorithm is to work at the receiving end, the HMM should be based on parameters present in conventional speech coders. HMM policy decision is taken to keep the perceived loss rate below some pre determined threshold value. This model recovers more number of loss packets for real time applications. But the HMM states need to be updated for every 5 seconds.

Adaptive load balancing policies use real time system state information based on various metrics like bandwidth etc to take load balancing decisions. To minimize the packet loss and to achieve load balancing Georgios Kambourakis et al [15] has proposed a lightweight SIP load balancing scheme that aims to provide services towards call transmission, link failures, device failures etc. The proposed load balancing scheme are evaluated on real time services on the Web Servers. Balancing the load of SIP transactions can affect the following factors such as availability, redundancy and QoS.

Network traffic is hard to measure in real time. Fauzia Idrees et al [16] proposed an effective VoIP traffic detection mechanism which is based on identifying generic characteristics of VoIP applications. The characteristics include packet arrival time, total number of packets received, average packets / seconds etc. The network traffic traces were captured using the different tools like wireshark, snort and ntop on Skype, MSN, YAHOO and Google Talk. These characteristics are monitored for security purposes to ensure their correct usage. This strategy also helps to evaluate the QoS for subscribers.

Hong Li et al [17] proposed a novel mechanism to select the optimal multipath that provides the best R-factor for VoIP phone calls with adaptive playback scheduling is applied at the receiver. The R-factor is a Score defined in ITU – TE model. It measures the subjective voice quality under many mouth-to-ear impairments. The VoIP performance is evaluated by comparing the R-factor, end-to-end delay of the VoIP calls through the optimal multipath with the direct path (the shortest hop path decided by the underlying network). Adaptive playback scheduling is a technique that is used at the receiver to adjust the playback delay and playback loss for the received voice signal. At each node, active probing UDP packets are sent to all other nodes to measure network delay, loss and jitter characteristics to improve the quality of VoIP applications.

To resolve congestion and packet loss, A QoS based power aware routing protocol (Q-PAR) is proposed. This protocol [18] works in two phases. In the first phase, the bandwidth

and energy constraints are built in into the DSR route discovery mechanism. The second phase deals about link failure. When link failure occurs, a repair mechanism is invoked to search for an energy stable alternate path. However, a priori estimation of the bandwidth and admission control is required to ensure the required QoS.

Another approach to resolve congestion for VoIP application uses Datagram Congestion Control Protocol (DCCP) [19]. It is a new message and connection oriented transport layer protocol. It differs from UDP, in that, it includes congestion control mechanism and it differs from TCP, in that, it does not provide guaranteed reliability. Each DCCP packet carries a sequence number so that losses can be detected and reported. But there is no re-transmission of lost packets. Thus efficient fault recovery routing scheme is necessary to minimize packet losses as proposed in this paper.

### III. PROPOSED METHODOLOGY

The proposed technique is mainly dealt with Forward Error Correction and extends a two phase routing with Encoding and decoding process to recover from packet losses using Vandermonde Matrix packet Level FEC mechanism.

#### A. Forward Error Correction

FEC is more attractive alternative to Automatic Repeat Request (ARQ) when delay is constraint. Packet impaired with both errors and packet loss is called as packet erasure. Vandermonde packet Level FEC can be used to recover from packet erasure. Packet-level FEC works by adding another error-recovery packet for every N packets that are sent.

This FEC packet contains information that can be used to reconstruct any single packet within the group of N. If one of these N packets happens to be lost during transfer, the FEC packet is used on the far end to reconstitute the lost packet. This eliminates the need to retransmit the lost packet, which dramatically reduces application response time and improves network efficiency. Encoded Packet P is given by

$$P = G \times M \tag{1}$$

where M represents the information packets and G represents the generation matrix. The vandermonde matrix G is augmented with the identity matrix (I) and designed matrix (v) (systematic code) forms the  $r \times c$  generation matrix code which is given by

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$$G = \begin{bmatrix} \mathbf{I}_{c \times c} \\ \dots \\ \mathbf{V}_{r \times c} \end{bmatrix} \tag{2}$$

For example, Encoded Packet P can be derived as shown in Fig. 3. A group of 6 VoIP packets (information Packets)

labeled M0 through M5 are used to create 4 FEC packets labeled FECP0 through FECP3. The reduced form of identity matrix is used as the generator matrix. Generator matrix is multiplied with the information matrix to produce 10 encoded packets. The first 6 encoded packets will be identical to M0 through M5, and the last 4 encoded packets will be the FEC packets.

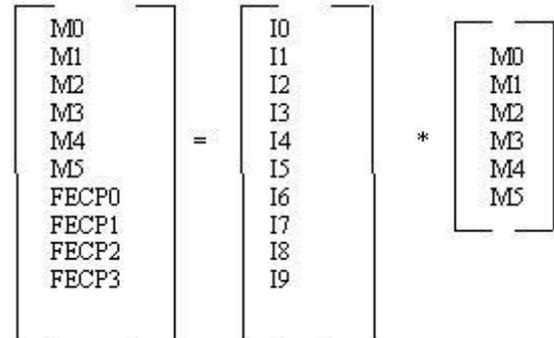


Fig. 3. FEC generation.

FEC works best on a high-rate aggregate flow, rather than on individual flows. As a result, it is best implemented in environments that use tunnels or aggregated flows. In addition an ideal FEC will adapt variable traffic changes in the network.

#### B. FEC Erasure Recovery and Error Correction Technique

At the receiver, all the coded packets P are received. The received packets arranged row-wise in a matrix  $R = E + P$  where E is the error packets. It is to be inferred whether which packet(s) is (are) in error, and then, within these packet(s), where are the error locations and their values. For binary codes, the error values are not required since by knowing their positions one just flips them. Syndrome decoding is used which depends only on the error packets and the parity check matrix. Erasure recovery and error correction techniques using one code word is given as follows.

First the erased positions of the received codeword are set to zero and the resulting codeword is decoded normally. The Hamming distance is measured between the codeword filed with zeros and the decoded codeword. Next the erased positions of the received codeword are set to one and the resulting codeword is decoded normally. The Hamming distance is measured between the codeword filed with ones and the decoded codeword. Then, the decoded codeword with the smallest hamming distance is chosen. This strategy is used to correct single erroneous packet in the group of n received packet.

For erasure recovery technique, at the receiver side, assuming no errors, if there are L lost information packets ( $L \leq K$ ), the missing information packets,  $M^L$  as follows.

$$M^L = (G^L)^{-1} \cdot (P)^L \tag{3}$$

where  $(.)^{-1}$  represents the inverse of a matrix,  $G^L$  is an  $L \times L$  sub matrix of G ( where  $G = n \times k$ ) that remains after proper substitution of the received packet.  $P^L$  is the received parity packets.

### C. FEC Erasure Recovery and Error Correction Technique

The two phase routing scheme is robust to extreme traffic fluctuation. In two phase routing, there is no necessity for the network to detect changes in the traffic distribution or reconfigure the network in response to it. The proposed scheme assumes the limits imposed by the ingress-egress constraints at each node. The packets lost due to overflow of buffers in the intermediate nodes are recovered.

In the direct routing the source node needs to know the destination of a packet for routing it. In two phase routing [20] source routes packets to intermediate nodes independent of their intended destination. The proposed FEC routing scheme operates in two phases as shown in Fig. 4.

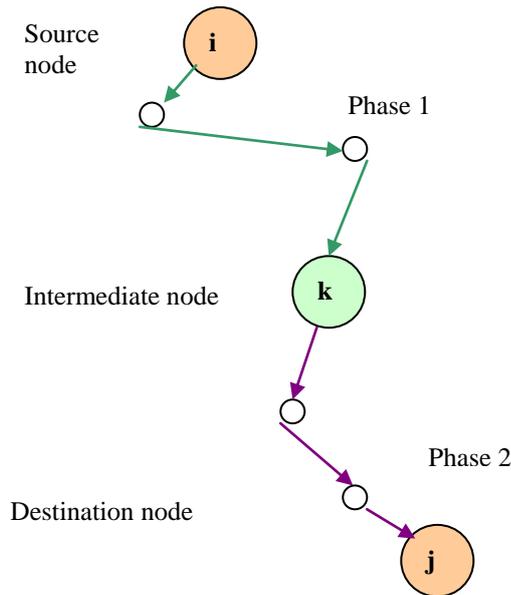


Fig. 4. Two phase routing.

Phase 1: The maximum incoming traffic  $T_i$  is encoded by the Vandermonde encoding technique before it is being forwarded to intermediate node. The encoded VoIP packets are transmitted to the Rendezvous Point (Intermediate node). The decision of RP does not depend on the destination node. RP is selected by evaluating link weight.

Phase 2: As a result of the routing in Phase 1, each RP node receives traffic destined for different destinations. These traffic are routed to their respective destinations in this phase. The destination node applies Vandermonde decoding if packet loss occurs. The two phase routing strategy depends on the ingress-egress capacities  $T_i$ ,  $C_i$  and the traffic split ratios  $f_j$ .

Algorithm: Phase 1

- 1) The maximum incoming traffic  $T_i$  from the source node  $i$  with traffic split ratio  $f_i$  is encoded by using Vandermonde encoding technique before it is forwarded to node  $j$ .
- 2) The split ratio of the traffic  $f_1, f_2, \dots, f_n$  is  $\sum_{i=1}^n f_i = 1$
- 3) Compute the intermediate node  $k$  with the smallest weight  $V(k)$  as well as shortest path  $R_i$  from node  $i$  to node  $k$ .
- 4) Send encoded traffic  $P[T_i]$  to node  $k$  on path  $R_i$  for all  $i$  towards the intermediate node  $k$ .

Phase 2

- 5) All traffic is initially split without regard to the final destination. Therefore find shortest path  $Q_j$  for all  $j$  and disperse the traffic from node  $k$  to  $j$  on path  $Q_j$ .
- 6) The maximum traffic now reaches the destination node. The destination node decodes the maximum traffic using the Vandermonde decoding methodology such as the erasure recovery and error correction technologies. The maximum traffic from node  $i$  to node  $j = f_j T_i + f_i C_j$ .

## IV. EXPERIMENTAL RESULTS

### A. Simulation Setup

The proposed Efficient Fault Recovery Two phase routing is simulated using the network simulator-2 (Ns-2) [21]. This routing scheme can be implemented in the network by forming fixed bandwidth paths between the nodes. The simulation topology Fig.5 consists of 11 nodes. It has one sender (ingress node) and 2 destinations (egress nodes) with 3 junction nodes. The link bandwidth is 6Mb and link delay is set as 10ms.

VoIP traffic for the ingress node with the following specifications is taken for routing. Packet size: 1000kb, Traffic Model: Exponential, VoIP codec: GSM.AMR, Number of VoIP frames per packet: 2, Rate: 5Mb, Encoder and Decoder: VoipEncoder, VoipDecoderOptimal.

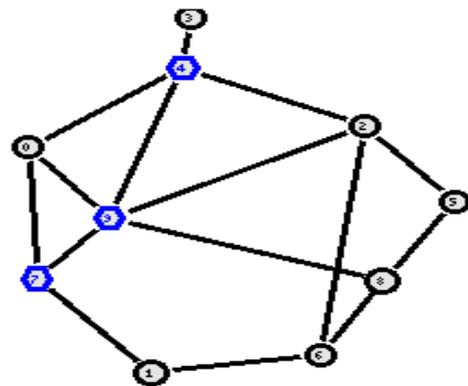


Fig. 5. Simulation topology for two phase FEC routing.

### B. Simulation Results

The proposed strategy is evaluated on voice samples based on the two primary QoS parameters such as received bandwidth and packet loss rate. Loss rate is given by the ratio of average number of packets lost at the receivers to the average number of packets sent. We also focus our experiment on voice samples.

The proposed Two Phase fault recovery Routing is compared with the two-phase routing with no Forward Error Correction technique. Since FEC adds overhead to the original data stream, it also has the potential to increase network load when no correction is needed. Fig. 6 shows that the received bandwidth (no of bytes) is more for the proposed strategy, than the two-phase routing with no FEC technique. In Fig. 7, the packet loss is more in the 2-phase routing, and it is least in proposed routing.

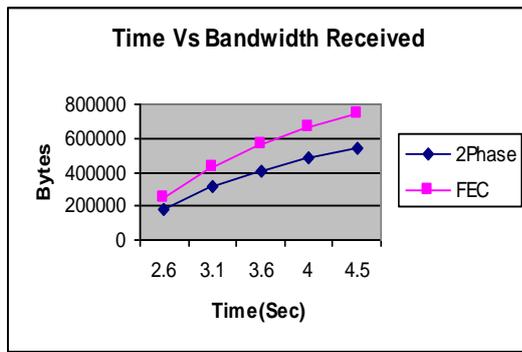


Fig. 6. Time Vs bandwidth received.

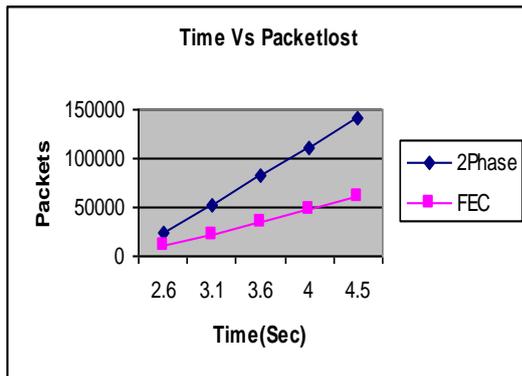


Fig. 7. Time Vs packet lost.

V. CONCLUSION

Efficient Fault Recovery Two phase routing for VoIP applications is presented to recover packets affected by both errors and loss against link failures. This routing scheme is implemented in the network by forming fixed bandwidth paths between the node and the traffic split ratios. The maximum incoming traffic is encoded by the Vandermonde encoding technique. The encoded VoIP packets are transmitted to the Rendezvous Point by splitting the traffic without regard to the final destination. Then each RP node routes the encoded packets to their respective destinations. The destination node decodes the traffic using erasure recovery and error correction techniques to recover from packet losses. By simulation result, it is shown that the proposed scheme improves the received bandwidth and packet loss rate for VoIP flows. As future work, this strategy can be extended into interactive streaming applications for mobile nodes.

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**Ms. J. Faritha Banu** received her Bachelor of Engineering in Computer Science and Engineering from Sathyabama Engineering College, Madras University, Chennai, India in 2001, Master of engineering in Computer Science and engineering from Sathyabama University, Chennai, India in 2005. Currently pursuing her PhD at Sathyabama University, Chennai, India.



**Dr. V. Ramachandran** received his PhD degree in Computer Applications in Power Systems from College of Engineering, Guindy, Anna University, Chennai, India. He is currently working as a Director in National Institute of Technology, Nagaland. He has received A Patent titled "Mechanism and Device for the Automatic Removal of Electronic Equivalent of Junk Mails". He has published more than 100 International and national journals, pioneered several new research directions, made a number of landmark contributions in his field. His areas of interests are Media Streaming, Hybrid Fault Tolerant Systems, Enhanced Network Routing Models, Fuzzy Based Network Security, distributed Models in Power System Analysis, Evolutionary Programming in Power System Applications, Service Oriented Architecture Models in Power System Analysis, e-Governance and Cloud Computing.