

## **Proposal for Robust and Adaptive Forward Error Correction for real-time Audio-Video streaming solutions**

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**ABSTRACT :** *In this paper, we present a robust and adaptive forward error correction technique for high end Audio-Video Streaming solutions. This solution targets all kinds of network loss conditions and can be used in any embedded platforms. This technique uses optimum solution for packet loss recovery that involves switching between adaptive FEC based packet loss recovery and retransmission management. Whenever network loss increases the permissible limit of adaptive FEC loss recovery, then it switches to re-transmission of media. Adaptive FEC based packet loss recovery works on best of adaptive matrix size selection, loss recovery method selection (on-the-fly or store and compute) and FEC codec selection (Parity 1D, 2D, Raptor, Reed Solomon and Hamming etc.) This technique also incorporates support for embedded systems which do not have much computation power. For such embedded systems, the adaptive FEC recovery technique switches to fixed matrix of having store and save approach for single and burst packet loss recovery. This technique provides inter-operability w.r.t FEC codec selection which also takes care of embedded system capability.*

**Keywords** - Adaptive FEC, Adaptive matrix size, FEC coding (Parity 1D, 2D, Raptor, read-soloman, hamming etc.), , FEC SDP negotiation, packet loss calculation techniques

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### **I. INTRODUCTION**

The conversation quality between the sender and the receiver in the Internet telephony system depends on specific impairments such as packet loss, delay, jitter, etc., which related to the characteristic of packet network. Forward error correction can be either media-specific or media-independent. Packet loss causes degradation in the quality of audio transmission applications. Forward Error Correction (FEC) methods which add redundant information to audio packets can be used to minimize the effects of packet loss. While these methods can reduce the effects of packet loss, they increase the amount of bandwidth used by an audio-video stream. Most of audio-video transmission applications use UDP as the transport layer protocol. The Real-time Transport Protocol (RTP) is used to provide additional functionality including sequence numbers and time stamps. The Real Time Control Protocol (RTCP) is used to return receiver statistics (e.g., the number of packets detected as lost) to the sender. RTCP packets are sent every 5 seconds by a receiver to a sender and consume very little bandwidth. IP networks are inherently best effort networks with variable packet delays and loss. Most of FEC technique does not target all network conditions and ignores the embedded system environment. Here in this paper, we have proposed solution for tackling all network conditions and all embedded system environments.

### **II. PROPOSED DESIGN**

Adaptive FEC based packet loss recovery technique provides the optimal solution for varying network conditions and for varying requirement of embedded solutions. Fig. 1 shows the architecture of receiver using adaptive loss recovery technique.

In Fig. 1, audio and video RTP and FEC packets are received by "RTP rcv" module and their corresponding RTCP packets are received by "RTCP rcv" module. Here "Media Loss-sense module" and "Adaptive Loss Recovery" module does adaptive FEC for packet loss recovery. "Retransmission Management" module triggers the retransmission of Audio-Video media. This triggering is done after having feedback from "RTCP loss Calculation" module of receiver.

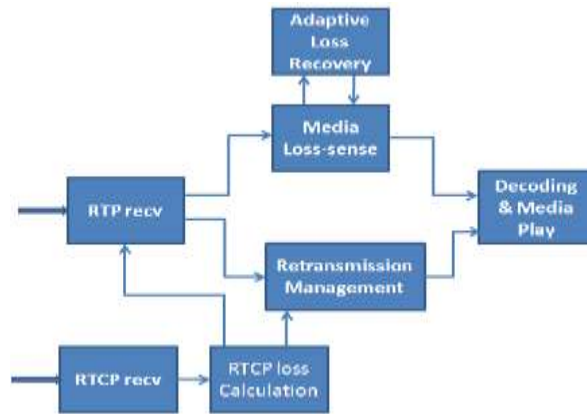


Figure 1. Architecture of receiver using adaptive packet loss recovery.

Packet loss recovery is done based on packet to packet loss calculation and switchover among adaptive FEC techniques is to be done based on packet loss patten found by “media loss-sense” module. Further switching between FEC based packet loss recovery and packet retransmission is to be done by “RTCP loss calculation”. Here our main concern is to make use of adaptive FEC techniques for packet loss recovery.

This adaptive FEC approach targets recovery of single packet loss, varying size burst packet loss and tackles varying embedded platform needs. Adaptability among FEC codecs and between lost packet recovery processing algorithms is done based on embedded solution requirement. Figure 1 shows the architectural diagram of receiver side having adaptive FEC recovery module. In this figure, media packet loss recovery uses adaptive FEC and retransmission technique. Here our main concern is targeting on all kind of network conditions. In real-time RTP streaming solutions, Adaptability of loss recovery algorithm comes either with loss-sense FEC algorithm or from RTCP statistics. Whenever there is heavy loss in the network then solution will switch to retransmission technique based on RTCP statistics, Otherwise it switches to loss-sense FEC module. This loss-sense module checks the type of network loss and switches the matrix size, recovery processing algorithm and FEC codec.

In case of single and burst packet loss, FEC recovery uses adaptive loss recovery module. This adaptive loss recovery module uses optimum solution for recovery of lost packets. This optimal solution works on best deal of matrix size, recovery processing algorithm, and FEC coding and decoding algorithm. While there is single and permissible burst loss, then FEC recovery module keeps fixed size of matrix. In case of more burst loss over the network, then this module increases the row size of matrix and decrease in case of small burst loss. For video streaming, excess burst loss creates glitches over screen and creating very annoying picture.

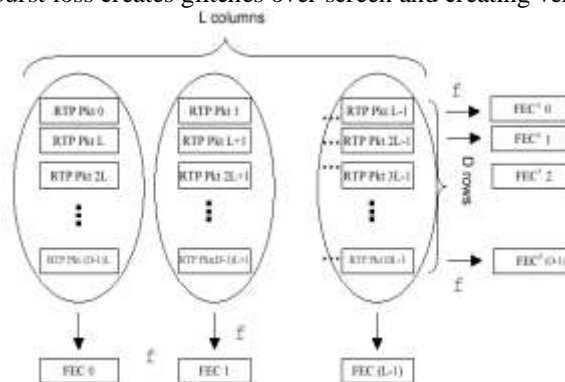


Figure 2. Considered parameters for Optimal FEC recovery .

Fig. 2 shows how to generate FEC packets from RTP packets. This 2D matrix is actually “CoP 3” algorithm depicted in [4]. In our adaptive FEC technique, we are changing the matrix size according to the size of burst loss. Whenever burst loss is more then row size of matrix will increased and will be decreased for less burst loss. So extending the idea of “CoP3” for tackling the varying burst loss network condition, we have proposed this in adaptive FEC recovery module. We have also proposed other ways along with the changing the size of matrix in this research paper. These are detailed in section 2 only.

There are some embedded platforms where there is a need of selecting between processing power and memory requirements. In case of having low speed embedded systems, FEC recovery modules selects store and compute processing algorithm and selects on-the-fly processing approach for high speed embedded solutions. Store and compute algorithm based FEC processing first saves at least  $m*n$  (row size \* column size) number of

media packets locally and then does the processing for recovering lost packet. For Store and Compute algorithm, Processing related to FEC is done for recovery of lost packets only and on the other hand, on-the-fly processing algorithm demands FEC decoding operations for every received media packet. Our FEC recovery module uses on-the-fly processing algorithm for packet loss recovery for those embedded platforms which has memory constraints.

Our FEC recovery module provides easy switching among FEC codecs to target inter-operable conditions. Adaptability among FEC codecs provides control on processing and bandwidth requirement for FEC packets. Fig. 3 shows the parameters considered for optimal FEC recovery. This shows how tuning among matrix size, loss recovery algorithm and FEC codecs helps us for fighting with different network conditions. Matrix size can have fixed row and column size and also can have different row and column size. Loss recovery algorithms involves on-the-fly and store and compute algorithms. FEC codec involves use of Parity-2D, raptor, read soloman, hamming codecs and etc.

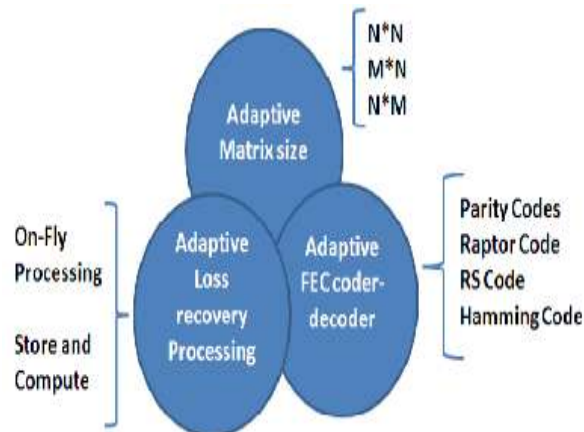


Figure 3. Considered parameters for Optimal FEC recovery .

### 2.1. Proposed Session Description Protocol (SDP) for Adaptive FEC

Adaptive FEC demands special handling for negotiating matrix/2D based FEC.

Below is an SDP required for static 2D based Audio FEC:

```
v=0
o=Adam 289083124 289083124 IN IP4 host.example.com
s=ULP FEC Seminar
t=0 0
c=IN IP4 224.2.17.12/127
a=group: FEC 1 2
  a=group: FEC 1 3
  a=group: FEC 4 5
m=audio 30000 RTP/AVP 0
a=mid:1
m=application 30002 RTP/AVP 100
a=rtpmap:100 ulpfec/8000
  a=mid: 2
m=application 30003 RTP/AVP 100
a=rtpmap:100 ulpfec/8000
a=mid: 3
m=video 30004 RTP/AVP 31
  a=mid:4
m=application 30004 RTP/AVP 101
c=IN IP4 224.2.17.13/127
a=rtpmap:101 ulpfec/8000
a=mid:5
```

Here two media (m) lines are added for row and column FEC packets. Attribute lines carrying group related parameters of FEC describes which media line is associated with these two row and column FEC streams. Whenever FEC codec is to be changed in between a call then SDP parameters are to be re-negotiated by signaling. Even whenever FEC coding technique is required to switch between 1D and 2D then also it is required to renegotiate SDP which will use only one m line for FEC streaming. In case of adaptive matrix size selection, SDP negotiation is the best way to settle with specific row and column size. Although matrix size selection can be intimated to end points with FEC packets also, but it will not provide sufficient time before

hand to change the buffer size required for row and column FEC packets. Overall adaptive FEC demands renegotiation of SDP for FEC codec switchover and 1D to 2D switchover.

M lines for adaptive FEC codec selection:

```
m=audio 30000 RTP/AVP 0
a=mid:1
m=application 30002 RTP/AVP 100
a=rtpmap:102 ulpfec1/8000
a=mid: 2
m=application 30003 RTP/AVP 100
a=rtpmap:102 ulpfec1/8000
a=mid: 3
```

In case of switching from 2D to 1D FEC technique, then SDP renegotiation will be with only one attribute line for grouping row FEC stream with specific media stream. Below is SDP for audio media stream, 1D FEC for audio stream and video media stream.

```
a=group: FEC 1 2
a=group: FEC 3 4
m=audio 30000 RTP/AVP 0
a=mid:1
m=application 30002 RTP/AVP 100
a=rtpmap:100 ulpfec/8000
m=video 30004 RTP/AVP 31
a=mid:4
m=application 30004 RTP/AVP 101
c=IN IP4 224.2.17.13/127
a=rtpmap:101 ulpfec/8000
a=mid:5
```

In the same way, SDP can also be added for FEC video stream. FEC technique can be differently applied for audio and video stream according to network bandwidth availability.

This switching among all techniques is to be done after sensing the network loss and available bandwidth. Adaptiveness among all loss techniques can be extended for forward error correction over selected media. Most of streaming applications includes at least audio and video streaming, so further it can be extended for adaptive FEC for selected media streaming. This adaptive FEC is going to be very useful for 3D audio-video content streaming. As multi-media applications are increasing day by day and will consume too much network bandwidth. For high definition 3D video streaming, adaptive FEC is going to be very useful feature for streaming devices.

### III. CONCLUSION

Matrix size switching targets the varying size of burst loss. Recovery processing technique switching targets the changing requirement of all embedded platforms. Switching among FEC codecs targets the increasing need of extra processing for recovery of lost. Further switching of FEC over selected media will surely reduce network congestion. This proposed solution tackles all varying network conditions and it is suitable for all embedded platform. In today's world when multi-media applications are moving toward 3D and holographic streaming then network is going to be very congested. For such cases, moving toward dynamic FEC solution will compensate for increasing network congestion conditions and will provide better end to end multi-media quality even for low end bandwidth carriers.

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**ABBREVIATIONS AND ACRONYMS**

FEC – Forward error correction

RS - Reed solomon coding

SDP – Session description protocol