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SURVEY ARTICLE

A Survey on FEC Based Packet Loss Recovery Technique in Networks

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Abstract— We survey a number of packet loss recovery techniques for many applications which are operated using IP multicast. We begin the discussion with the loss and delay characteristics of an IP multicast channel, and from this we show the need for the packet loss recovery. And these recovery techniques may be divided into two classes: Sender-based and Receiver -based. We compare and contrast several sender-based recovery schemes: forward error correction (both media-specific as well as media-independent), retransmission and interleaving. We conclude with a series of recommendations for repair schemes to be used based on application requirements and network conditions.

Keywords— interleaving, reliability, retransmission, congestion, interleaving, security, packet loss

I. INTRODUCTION

Generally in Networks packet losses occur due to many reasons like buffer overflow, congestion control and many more. For recovering lost packets in IP networks we have many techniques and they are classified as Sender based and receiver based etc. A multicast channel typically has quite high variation in end-to-end delay and relatively high latency [1]. This delay variation is a reason for concern when developing loss-tolerant real-time applications and so the packets delayed too long will have to be discarded in order to meet the application's time requirement, leads to the appearance of loss. This problem is more acute for interactive applications: if interactivity is unimportant, a large playout delay may be inserted to allow for these delayed packets.

It should be noted that the characteristics of an IP multicast channel are significantly different from those of an asynchronous transfer mode (ATM) or integrated services digital network (ISDN) channel. The techniques discussed here do not necessarily generalize to conferencing applications built on such network technologies. The majority techniques are applicable to unicast IP networks, although the heterogeneity and scaling issues are clearly simpler in this case.

We shall discuss the techniques which require the participation of the sender of an audio stream to achieve recovery from packet loss.

These techniques may be divided into two major classes:

1. Active retransmission
2. Passive channel coding.

It is further possible to subdivide the set of channel coding techniques [2], with traditional forward error correction (FEC) and interleaving-based schemes being used.

In order to simplify we distinguish a *unit* of data from a *packet*. A unit is an interval of audio data, as stored internally in an audio tool. A packet comprises one or more units, encapsulated for transmission over the network.

Approaches available to recover lost or dropped packets in the network are classified into two types :

1. Automatic Repeat request (ARQ)
2. Packet-level Forward Error Correction (FEC).

ARQ is based on retransmitting the lost packets and has been applied to the TCP (Transmission Control Protocol) for best-effort packet delivery. The idea behind FEC-based packet recovery, however, is to introduce controlled redundancy into the original message prior to transmission. Instead of retransmission, this redundancy is exploited to reconstruct any lost packets at the receiver.

Of these two techniques, FEC has been more commonly suggested for real time application, such as applications using the RTP (Real-time Transport Protocol), due to strict delay requirements. In addition, the amount of redundancy is expected to be minimal, which further reduces transmission delays.

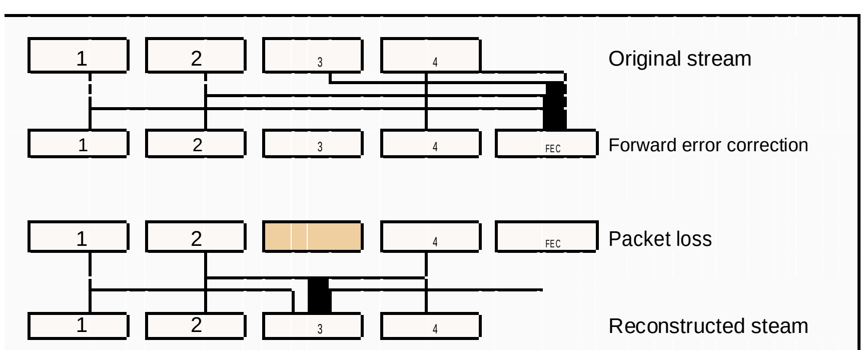

Figure 1: Repair using parity FEC.

II. LITERATURE SURVEY

Forward Error Correction:

It is the process in which a duplicate data (repair data) is to be transmitted along with the original data in order to avoid the process of retransmission this is known as controlled redundancy [3], it will be used to recover the lost data at the receiver. FEC is suggested to be used in the real time applications such as RTP (Real Time Transport Protocol) because of the requirements on the delay time [4]. The redundancy of this data should be maintained minimal to avoid the transmission delay. FEC can be done by adding the repair data streams to the original data which is to be recovered from them if any loss happened during transmission.

FEC Broadly divided into two types

1. Media Independent FEC
2. Media dependent FEC

2.1 Media Independent FEC:

In this additional packets of data will be produced in the form of blocks or algebraic codes to recover the lost data during transmission i.e., to transmit n packets of data over a network it has to generate n-k additional check packets for the code word of k data packets. Out of a large number of block coding techniques we discuss only parity coding and Reed-Solomon coding which are suitable for RTP.

Parity Coding:

In Parity coding parity packets can be generated by conducting an XOR operation across a group of packets. Eg: As implemented by Rosenberg after n-1 data packets transmission on parity packet will be transmitted provided there should be only 1 packet loss in n packets and that can be recovered many parity packets are generated by XOR operation performed on different packet combinations and these are proposed by Budge and summarized by Rosenberge and Schulzrinne[2].

Reed Solomon codes are well known for error correcting properties particularly their resistance against burst losses[5] . Encoding depends on the polynomial properties over number bases. Coefficients of these polynomials are the set of code words taken by the RS encoders. Encoding process is straight forward and Berlekamp-Massey algorithm is used for decoding[6]. When packet losses doesn't exist then the computational cost of decoding is same as that of encoding when losses occur it is costlier/expensive.

Advantages:

- Media Independent
- Operation doesn't depend on content of packets
- Simple computation to derive error correction packets and easy to implement.

Disadvantages:

- Additional delay in transmission
- Difficult decoder Implementation
- Increased Bandwidth.

2.2 Media Specific FEC:

To protect the packet loss during transmission every audio unit will be sent in multiple packets i.e., if one packet is lost then the other packet having the same content will be utilized to recover the lost data. This process was advocated by Hardman and Bolot and is simulated by Podolsky

The first audio data is considered as Primary encoding and the subsequent transmissions are considered as secondary encodings. It is the sender's choice that the secondary encodings is of the same encoding scheme or not usually secondary encoding will be of low bandwidth and low quality as compared to that of the primary encoding and basically depends on bandwidth requirements and computational complexity Erdol proposed short term energy and zero crossing measurements for secondary encodings because at the time of loss receiver interpolates the audio signal about crossing using zero measurements it can be coded compactly and is computationally cheaper but it covers only short period loss because of crude measures. Hardman and bolot proposed low bit rate analysis using synthesis codecs and full rate GSM coding which is computationally demanding and can cover the loss periods on internet[7]. If the primary encoding has low bandwidth and covers considerable power then the secondary encoding will be same as primary Eg: International Telecommunication Union (ITU) G.723.1 [6] codec which consumes a considerable fraction of today's desktop processing power, but has a low bandwidth (5.3/6.3 kb/s).

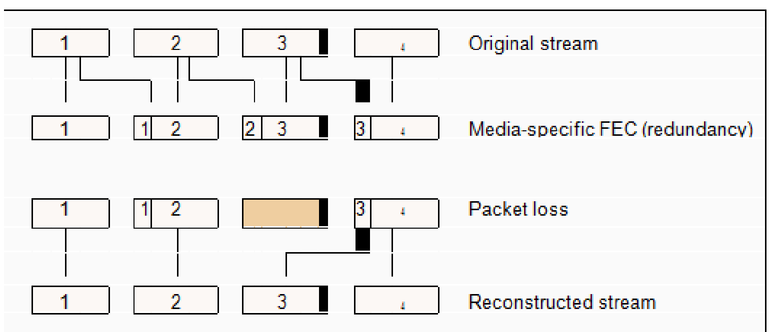

Figure 5. *Repair using media-specific FEC.*

Media specific FEC has overheads in terms of packet size and these overheads are variable like in media independent but in this the over head can be reduced without affecting the number of losses unlike in media independent but quality will vary and to reduce overhead approximate repair should be used.

Foe every packet a media specific FEC need not to be transmitted because audio signal have transient stationary state of 80ms., hence viswanathan proposed to send LPC codecs where 30% of bandwidth will be saved without any quality loss. So at the time of transmission a decision should be taken depends on the situation whether to transmit a FEC or a LPC codec

Advantages:

- Low Latency because of the addition of single packet delay which will be used in interactive applications where large end to end delay cannot be tolerated.
- Media specific FEC is supported by Mbone audio conferencing tools at the time of writing[8]

Addition of large amount of repair data to resolve the data problem will increase the network congestion resulting in packet loss but by media specific FEC congestion control can be done in this by the addition of FEC it will repair data to a media stream and will protect it from packet loss. This is more important while sending large multi cast groups because of network heterogeneity which results low capacity regions to suffer congestion while high capacity regions are being underutilized and interleaving concept is used to increase the security aspect in networks.

At the time of writing there is no particular solution for this problem but Layered encoding will be employed for long term encoding and FEC for short term encoding in Layered encoding data will be sent at different rates over multiple and multicast groups with receivers leaving and joining the groups in response to the congestion it is expected to provide solution for streaming audio congestion but this work is not yet complete [9].

III. CONCLUSION AND FUTURE SCOPE

In this paper we discussed Forward Error Correction has great potential in recovering the packet losses caused due to congestion in a packet-switched network, provided that the coding rate and other coding parameters are chosen appropriately. Future work is the analysis of the additional delay caused by the FEC coding, perhaps combined with new Interleaving concept. Even though there are many codes like Reed Solomon for coding and algorithms like Berlekamp-Massey algorithm ,Back propagation still there is a need to improve the efficacy of FEC coding combined with interleaving in the combating packet losses in IP networks.

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BIOGRAPHY

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