
WE TRIP THE LIGHT
FANTASTIC

CONTRIBUTION AND DISTRIBUTION OVER IP NETWORKS

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ABSTRACT

The move to IP networks for contribution and distribution provides broadcasters with flexible bandwidth pipes and greater flexibility than existing cell based networks such as ATM. With the move to IP networks broadcasters are faced with the challenge of providing the reliability and quality of service associated with traditional SDH and PDH networks over new IP networks. The transmissions of real-time broadcast video over IP networks must be protected from network congestion and packet loss as these degrade the received service. The Pro-MPEG Forward Error Correction (FEC) specification addresses the issue of reliability and defines a flexible protection scheme which can be tailored to the characteristics of the transmission network. When combined with IP quality of service protocols such as MPLS, IP networks are able to meet the demanding requirements for the transmission of real-time broadcast quality services.

INTRODUCTION

Within the contribution and distribution arena traditionally telco networks have been used for the transmission of real-time content due to their deterministic characteristics. The carriage of real-time content places stringent requirements on the underlying network. A real-time broadcast requires the following transmission characteristics, constant bandwidth, constant latency, and maximum defined error rate. These are important characteristics and are affected by the transmission network. Traditional networks such as ATM are cell-based, and provide deterministic characteristics which are optimal for broadcast transmissions, whereas Ethernet network are inherently un-deterministic.

In real-time applications all packets must be delivered on time and without loss since retransmission is not possible. Broadcasters have traditionally used networks, such as ATM which provide the required quality of service. With developments in IP networks to address quality of service broadcasters are looking at IP for their next generation networks. In order to understand the demands placed on the network the transmission characteristics for streaming real-time content must first be understood.

Required transmission characteristics for real-time content.

- 100% network availability
- Guaranteed bandwidth
- Constant network latency
- No packet loss or defined loss allowing real-time recovery through FEC schemes

If any of these characteristics are not meet then service disruption can occur. These characteristic can be summarised as deterministic and are inherent in ATM networks. The affects of not meeting these characteristics vary. For a real-time stream a fixed bandwidth is required for the duration of the transmission, if the network is not able to provide sustained bandwidth then service disruption will occur. If the network introduces packet loss or variable latency the receive devices will not be able to replace missing or delayed packets without affecting the service. Although traditional networks running SDH, PDH provide these characteristics they impose restrictions on the broadcaster.

- Restricted network access (dependent on termination access)
- High equipment costs for network termination
- Complex network management
- Manual bandwidth management (inflexible)

The introduction of IP networks offers the broadcaster savings and enables flexible carriage of mixed data types over a common network while lowering the cost of access. The move to IP networks provides the following benefits.

- Lower equipment cost (network adaptors, switches, cost per port)
- Improved access (point of presence)
- Dynamic bandwidth allocation improving core network utilization

Moving to IP is not a simple replacement since the network characteristics are different to those provided by traditional networks. Networks such as ATM were designed to provide high availability, constant bandwidth and fixed transmission latency were as IP networks were designed for random occasional traffic utilising a ‘best effort’ approach. We are not talking about using standard IP network for the transmission of broadcast quality video but broadcast IP network designed for the distribution of broadcast content. With the demand on IP networks to provide a quality of service consistent with traditional networks, a number of protocols can be applied to provide the required quality of services.

In order to protect real-time broadcasts over an IP network the use of quality of service protocols such as Multi Protocol Label Switching (MPLS) are required to ensure that the broadcast traffic is given a constant level of service. Combining network Quality of Service (QoS) protocols alongside error protection schemes such as Pro-MPEG Forward Error Correction (FEC) for lost packet recovery, enable reliable transmission of broadcast traffic over IP networks.

VIDEO OVER IP

The default transport medium for real-time compressed MPEG content is the MPEG-2 Transport Stream (TS) packet. The carriage of MPEG-2 compressed content over traditional ATM networks involved the encapsulation of MPEG-2 TS packets in ATM cells.

In the IP domain the transport stream packets must be encapsulated within IP packets to allow transmission across the network. The IP packet is normally restricted to 1500 bytes as this fits a standard 1500 byte Ethernet frame without fragmentation. A 1500 byte frame is chosen assuming most streams will pass over an Ethernet at some point. If a frame size greater than 1500 bytes is used then at some point the frame may need to be spilt into two frames of 1500 bytes, this will introduce jitter by varying the inter arrival time of the packets at the receive point. A 1500 byte packet allows a maximum of seven 188 byte MPEG-2 TS packets per an IP packet. A single IP packet containing 7 MPEG-2 TS packets is shown in Figure 0 - MPEG-2 TS packet encapsulation.

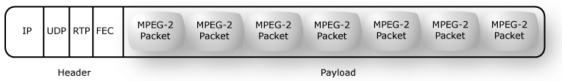


Figure 0 - MPEG-2 TS packet encapsulation

The encapsulation of 7 MPEG-2 TS packets into a single IP packet adds a 5% overhead providing optimum efficiency and minimising encapsulation jitter. Therefore a single lost IP packet would result in 7 lost MPEG-2 packets. If a packing ratio of 1:1 were used then minimal loss would occur but the encapsulation overhead would be 35% lowering efficiency and increasing packet jitter. Figure 0 – Transmission overhead vs. packing ratio shows transmission overhead as a percentage, against packing density.

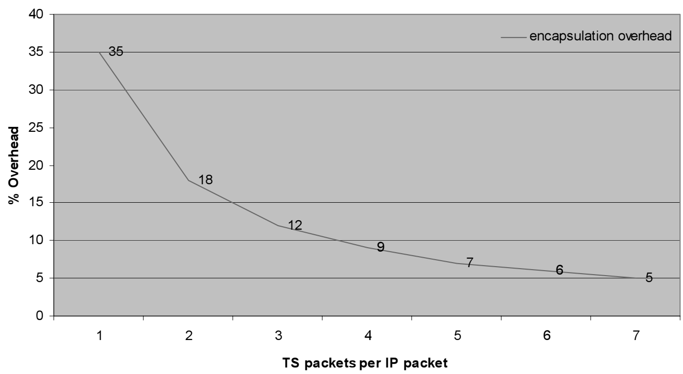


Figure 0 – Transmission overhead vs. packing ratio

Following encapsulation the IP packets are streamed onto the network. The process of encapsulation and streaming happens in real-time. The IP packets are now routed to the receive points by the network were the MPEG-2 TS packets will be de-encapsulated and presented to the end device.

Transmission Impairment

The IP packet flow must now transverse the network to the destination device or devices. The process of transmission across the network may introduce impairment into the received MPEG-2 stream. The following impairments may be introduced during transmission.

- Packet loss due to random or burst errors on the network
- Jitter from queuing within network devices
- Sequence errors (out of order packets)

Impairments will result in service disruption unless corrected. Depending on the level of impairment, correctional techniques can be applied to recover and provide a stable transmission. Once such technique is the application of Forward Error Correction (FEC) which can allow the receive point to recover missing packets. In addition network protocols such as traffic classification and prioritization can be used to apply edge-to-edge quality of service based in traffic type.

PRO-MPEG FORWARD ERROR CORRECTION

The IP layer provides mechanisms to recover missing packets using the TCP protocol through re-transmission. Unfortunately this is not possible in a real-time transmission. The use of a real-time error correcting scheme such as Pro-MPEG Code of Practice #3 FEC is required to protect real-time business critical transmissions from packet loss. For streaming either User Datagram Protocol (UDP) or Real Time Protocol (RTP) can be used for encapsulation. Often UDP is used as this adds only 4% overhead and requires less processing at the receive device. The use of RTP adds 5% overhead but provides packet sequence detection allowing a receiver to identify out of sequence, discarded or reordered packets. The Pro-MPEG FEC scheme uses the RTP transport protocol as the building block for providing packet recovery techniques to ensure reliable real-time transport.

The Pro-MPEG standard defined the transmission of FEC data streams alongside the unicast or multicast media streams. These additional FEC streams are used by FEC enabled receivers to recover missing packets. The use of separate IP streams allows non FEC enabled receiver's access to the media streams. The use of separate streams also provides flexibility in insertion of the FEC data relative to the media stream it is protecting, so that burst errors do not result in media and FEC packet loss.

The RTP protocol does not specify the payload format allowing carriage of MPEG-2, MPEG-4 or SDI transparently. The media and FEC streams utilise RTP encapsulation. In RFC2733 the RTP format for FEC data was defined. The FEC RTP format restricts the number of consecutive packets that can be used to generate the error-correcting packets to 24 limiting the burst correction capabilities of any FEC scheme based on RFC2733. The Pro-MPEG standard defines an extension to RFC2733 allowing non-consecutive packets to be used for generating FEC packets removing the 24 burst correction limit. In this scheme periodically selected media packets are used to generate the FEC packet.

FEC packet generation is based on the use of a matrix defined by two parameters L and D, Were L is the spacing between non-consecutive packets used to calculate the FEC and D is the depth of the matrix. The scheme defines two FEC packet types, column FEC and row FEC. The FEC scheme is based on the exclusive OR mathematical function which allows correction of any single packet loss from the series of packets used to form the FEC packet. The transmitting device must calculate the FEC packets in real-time and stream these alongside the media stream.

Column FEC Generation

Column FEC provides correction for consecutive burst packet loss of up to L packets. The FEC packets are generated per a column within the matrix allowing recovery of any single media packet within a column or burst of errors within a row to be corrected. Column FEC is ideal for correcting packet burst and random errors. In column FEC the XOR function is applied to every L^{th} media packet with the next L packet for D packets to generate the FEC packet. In an L=D=4 matrix the first column FEC packet would be calculated as shown in the following equation.

$$FECpacket\,(0) = M_1 \oplus M_5 \oplus M_9 \oplus M_{13}$$

Row FEC Generation

Row FEC provides correction of non-consecutive packet loss, allowing correction of any single lost packet within a row. The FEC packets are generated per a row allowing recovery of any single packet. Row FEC is ideal for correcting random packet errors. In row FEC the XOR function is applied to L consecutive media packets to generate the FEC packet. In an L=D=4 matrix the first row FEC packet would be calculated as shown in the following equation.

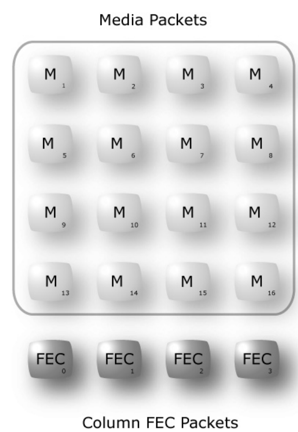
$$FECpacket\,(0) = M_1 \oplus M_2 \oplus M_3 \oplus M_4$$

The standard defined two transmission modes for the FEC packets. Either column FEC or column and row FEC packets maybe transmitted alongside the media stream. The application of column FEC is often referred to as 1 dimensional FEC whereas column and row FEC is termed 2 dimensional FEC.

In Figure 0 – FEC Matrix, Column FEC packet generation for a 4x4 matrix is shown. The combination of column and row FEC provides a robust error protection scheme capable of dealing with random and burst errors, able to correct more errors than either column or column and row schemes can independently.

The matrix size is user defined and only restricted by values of L and D. To promote interoperability a range of matrix sizes which must be supported are defined. The matrix range is defined by L and D with limits of LxD≤100, 1≤L≤20 and 4≤D≤20 and L≥4 for column and row FEC.

The matrix size defines the additional overhead in FEC bit rate required. A large matrix reduces the ratio of FEC to media packets while a small matrix increases the ratio. The matrix size also affects the latency at the receive device in addition to any IP receive buffer for the removal of jitter.



FEC Transmission

Once calculated the FEC packets are transmitted alongside the media stream. The media and FEC packets are transmitted on separate IP streams using different UDP ports. Figure 0 – Media and FEC Transmission, the separate IP streams for the media and column and row FEC packets are shown.

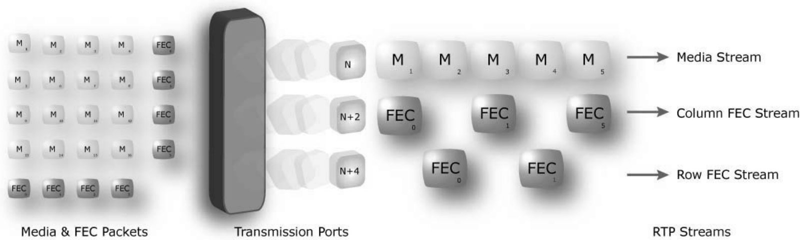


Figure 0 – Media and FEC Transmission

The FEC packets are transmitted on UDP port numbers offset from the media stream by +2 for column and +4 for row FEC as defined in the standard. This enables Pro-MPEG enabled receivers to automatically locate the FEC streams.

The standard allows definition of user defined UDP ports. The receiver should allow manual entry of non standard ports as port translation may occur during transmission. The use of separate streams enables reception by enabled and non-enabled FEC receivers, were FEC enabled receivers utilise the additional FEC streams to correct missing or corrupted packets, while non-enabled receivers are unaffected by the FEC packets.

FEC Reception and Processing

Once the IP packets reach the receiver they are first placed in a de-jitter buffer which removes jitter introduced by the transmission. The size of the de-jitter buffer will determine the amount of IP jitter that can be tolerated and additional latency. After de-jittering the receiver must determine if the FEC packets are needed. By using the RTP sequence number missing or out of order packets can be detected and missing packets can be corrected using the relevant FEC packets if available.

The packets are then placed in a receive matrix were the FEC packets are applied to generate missing packets, this process maybe iterative and introduces a delay dependent on the matrix size and the chosen FEC scheme. Figure 0 – FEC Processing shows the recovery processing of missing media packets using column and row FEC.

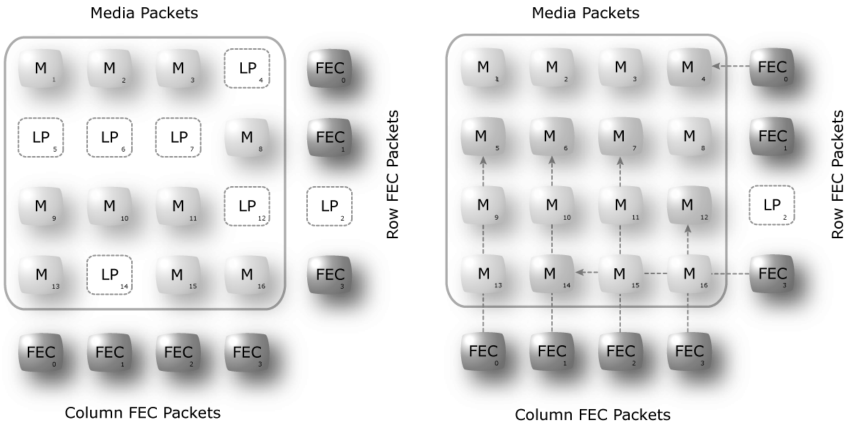


Figure 0 – FEC Processing

Following the recovery of missing packets the media stream maybe placed in a smoothing buffer to ensure a compliant ASI stream containing MPEG-2 TS packets is generated. We have now seen the complete cycle starting with encapsulation, FEC packet generation, transmission and packet recovery. The protection offered by the FEC scheme comes at a cost in terms of additional transmission latency and bandwidth overhead. The FEC scheme can be tuned to meet latency, transmission bandwidth and packet recovery requirements. The additional sources of latency added as a result of transmission over IP and Pro-MPEG FEC is listed below.

- MPEG-2 TS packing ratio per an IP packet, with 1:1 giving the lowest latency
- FEC matrix dimension, a large matrix will introduce more latency than a small matrix

A choice is made between latency, FEC overhead, matrix size and number of correctable packets that best suite the network and application. Figure 0 – Parameter Trade, show the trades possible.

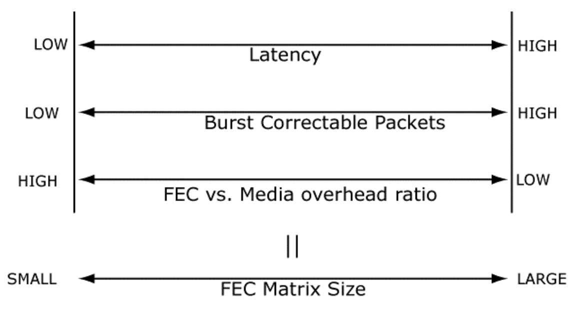


Figure 0 – Parameter Trade

TRANSMISION OVERHEAD

We have now examined the complete end-to-end process for delivery over IP networks from encapsulation through to FEC correction and the potential affects of IP transmission. The IP encapsulation and addition of FEC data to protect against packet loss add overhead to the transmission bandwidth. In ATM networks MPEG-2 TS packets are encapsulated within ATM cells and in IP network they are encapsulated in IP packets (Ethernet frames). All networks add overhead for encapsulation and error protection. We shall now examine the differences MPEG-4 makes and compare this to MPEG-2 in delivery over IP networks.

With the increase in demand for High Definition (HD) content, demand on distribution networks has increased. The industry has moved towards Advanced Video Coding (AVC) for HD content and traditional broadcast have moved to adopt MPEG-4 AVC as the chosen encoding format. Moving to MPEG-4 AVC typical provides a 40% saving in bandwidth over MPEG-2 encoded content, enabling broadcaster to offer HD services to the home. The savings don't stop with the gains in video bit-rate, additional saving when considering distribution over IP networks are gained. The 40% bit rate saving applies to the IP encapsulation and FEC data overhead reducing the total transmission bit-rate. Table 1 – Transmission overhead, show an example for a typical Standard Definition (SD) and HD transmission.

Table 1 – Transmission overhead

	SD		**HD**	
	MPEG-2	MPEG-4	MPEG-2	MPEG-4
Service MPEG-2 TS bandwidth (Mbits/sec)	4	2.4	17	10.2
IP Encapsulation overhead (Mbits/sec)	0.2	0.12	0.85	0.51
Service TX bandwidth + Column FEC (Mbits/sec)	1.05	0.63	4.46	2.68
Service TX bandwidth + Column Row FEC (Mbits/sec)	1.26	0.76	5.37	3.22
Total MPEG-4 saving over MPEG-2		2.1		8.95
Percentage MPEG-4 saving		40%		40%

Taking the HD example we can see a 6.8Mbits/s saving on the service through video bit rate reduction and an additional saving of 2.15Mbits/s in encapsulation and FEC data. This gives a total saving of 8.95Mbits/s which could be used to provide additional services. The move to MPEG-4 makes IP delivery more attractive as the overall transmission bandwidth demand drops which in turn reduces the potential for packet loss and delay within the network. We have now introduced the Pro-MPEG FEC scheme for protection against packet loss and looked at the impact of MPEG-4 AVC encoding on the transmission rates. We shall now focus on the IP network and the use of quality of service protocols in providing services levels dependent on traffic classification and priority.

QUALITY OF SERVICE

Through the application of quality of service protocols IP networks behave with deterministic characteristics that are suited to high quality transmissions rather than in a best effort un-deterministic manner. An Ethernet network operating at low utilization will have characteristics suitable for the transmission of real-time video services, providing sufficient bandwidth, low or no packet loss and constant latency.

As the network utilization increases towards congestion its performance will deteriorate unlike an ATM network, which provides constant performance regardless of utilization.

In Figure 1– Low utilization, two packet flows through a network device such as a switch or router are illustrated. The network device is not under load and the egress link has sufficient bandwidth to service both traffic flows without packet loss. This shows that whilst a non-QoS enabled network is capable of delivering real-time broadcast quality content, it is dependent on the networks loading which is variable.

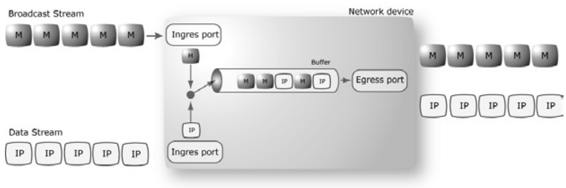


Figure 1– Low utilization

As the network loading increases the network will become congested resulting in packet loss as the network devices discard packets due to insufficient link capacity. At this point packets will be randomly discarded as the internal queues overfill. This means that the networks performance will vary dependent on loading. Figure 1 – High Utilization (no QoS), shows that not all packets arriving at the ingress ports are present at the egress port.

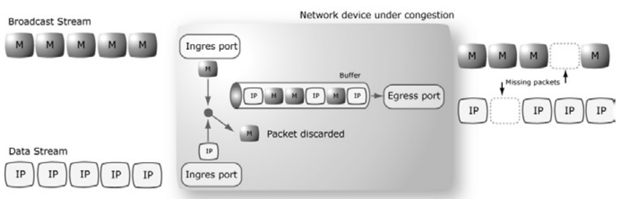


Figure 1 – High Utilization (no QoS)

This also affects the latency of the transmission due to variable queuing delay resulting in inter-packet arrival time variance impacting on MPEG jitter. If packets are lost from the broadcast stream, service disruption will occur since all packets are required to reconstruct the service at the receiver unless some form of error protection is implemented. The use of quality of service protocols provides deterministic network characteristics for specific traffic types, allowing network operator to guarantee bandwidth, latency and packet loss rate for different content types. This allows operators to use a single network to distribute a mix of real and non-real time content making the network more efficient and driving down operating cost. Quality of service works by classifying traffic flows entering a network, once classified policies are applied determining the transmission priority within the network. This allows network providers to segregate traffic flow within the network giving priority for real-time transmission of video and audio services over non-real-time critical traffic.

The application of QoS within the network enables the network operator to guarantee service levels required for service level agreements and allows the network to reserve capacity for the broadcast traffic when required without losing the inherent flexibility gained in using IP networks. Figure 2 - Congestion with QoS shows a QoS enabled network device under congestion, now the broadcast service is guaranteed a bandwidth agreed under a service level agreement. The broadcast service is unaffected by the congestion while the un-protected traffic is handled on a best effort basis resulting in potential packet loss during periods of congestion. Bandwidth management must be planned correctly when running QoS otherwise congestion can still occur on protected traffic, link capacity between routers within the network must be calculated to ensure that sufficient capacity if available.

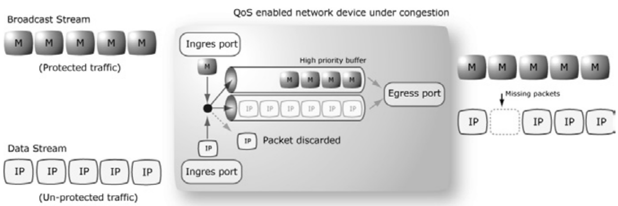


Figure 2 - Congestion with QoS

The use of quality of service protocols allows the network operator to meet the requirements for the transmission of real-time broadcast quality media. To ensure the most reliable transmission of broadcast video over IP network QoS should be used in conjunction with an error-correcting scheme such as Pro-MPEG CoP r3 FEC.

SUMMARY

With the application of QoS protocols, IP networks are now able to deliver the deterministic characteristic required for the transmission of real-time broadcast streams. The combination of network QoS and error correction schemes such as Pro-MPEG FEC within the video network adaptors provide a robust mechanism for the transport of real-time video over IP networks, allowing IP networks to provide the level of service traditionally associated with ATM networks but with the benefits of IP. With the move to MPEG-4 AVC additional bandwidth savings are made through the reduction in encapsulation and FEC overhead which are proportional to the initial service bandwidth. As a result of these technologies broadcasters now see IP networks as a viable offering and are moving towards IP enabled networks for their next generation contribution and distribution networks.

REFERENCES

In preparation of this paper the following references were used.

1. Pro-MPEG Code of Practice #3 release, 2 July 2004
2. Pro-MPEG FEC Presentation by Neil Trimboy, 12th April 2005
3. Quality of Service in IP Networks by Grenville Armitage, Pearson Education