White Paper

Experience with SDI Contribution over IP Network

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Introduction

The growth of IP based service in the national and international market has been tremendous, and the traffic volume is comparable to ordinary voice. At the same time the cost of IP technology is considerably less than that of ATM and SDH. The consequence is that telecom operators are constructing new IP networks with the intent of converging all services to IP. The result for the broadcaster is that it will be more economical to base new contribution and distribution networks on IP. The Pro-MPEG Forum has proposed two standards for transport of video over IP. Code of Practice 3 (CoP 3) covers compressed video in the form of MPEG-2 Transport Streams. This standard is well advanced and equipment is available from several vendors, as being demonstrated at this event. Code-of-Practice 4 is similar to CoP 3 and concerns transport of uncompressed video - SDI. This paper describes the experience with using CoP 4 for contribution. Moreover, the paper includes a description of the standard and configuration trade-off for SDI transport.

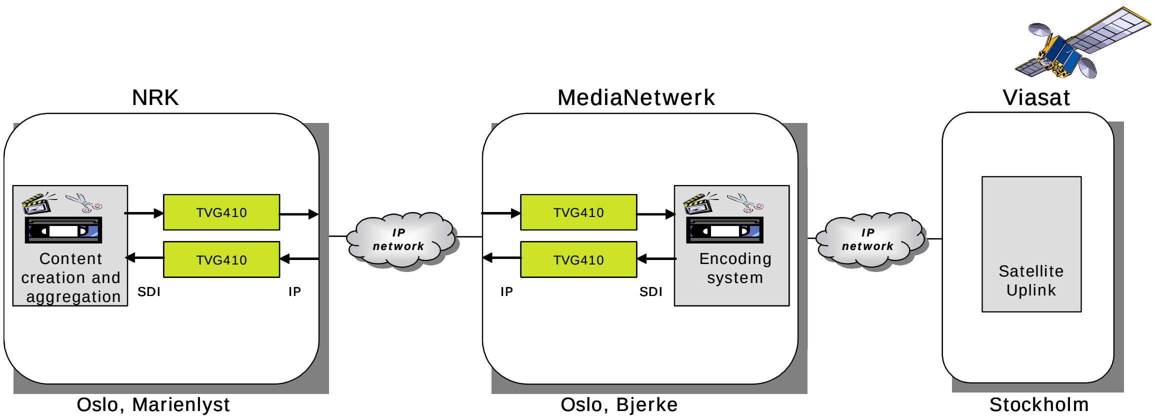
Case study: SportN and Media Netwerk

The competition in DTH is equally strong in Norway as in any other country. NRK and MTG, have decided to join forces and have launched a new sports channel in order to compete with Canal Digital and TV2 after they purchased the rights for the Norwegian Soccer League for the next three years. The partners decided that content should be compressed and transported from the NRK studio in Oslo to the satellite uplink in Stockholm. The network operator responsible for video transportation, Media Netwerk, decided to carry out the MPEG-2 encoding at its own studio as this provided some operational advantages, for instance adding live commentary to programs. The major benefits of selecting SDI over IP are no compression distortion, no compression delay and 10 bit video resolution.

Media Netwerk is one of the pioneers in Scandinavia in using IP for Broadcasting following its decision to replace satellite SNG with IP-based contribution back in 2001. Its network consists of fiber as well as Nera's Citylink (microwave radio links with Ethernet interfaces). The purpose was for backhaul of horse racing from race tracks in Norway and this network is still used every day.

For transport of the new, "*SportN",* channel T-VIPS supplied 2 pairs of TVG410 SDI to IP Gateways. The main transmitter/receiver pair is used for the forward path from NRK to Media Netwerk while the second is used to send a return signal back to NRK. As the TVG410 gateway can be configured either as a transmitter or receiver the second pair also serves as backup for the main pair. The IP network involves three Cisco Catalyst 3750. The connection uses layer 2 with VLAN trunking. In addition to the SDI signal there a number of other services going over the same VLAN. The equipment was installed in the middle of

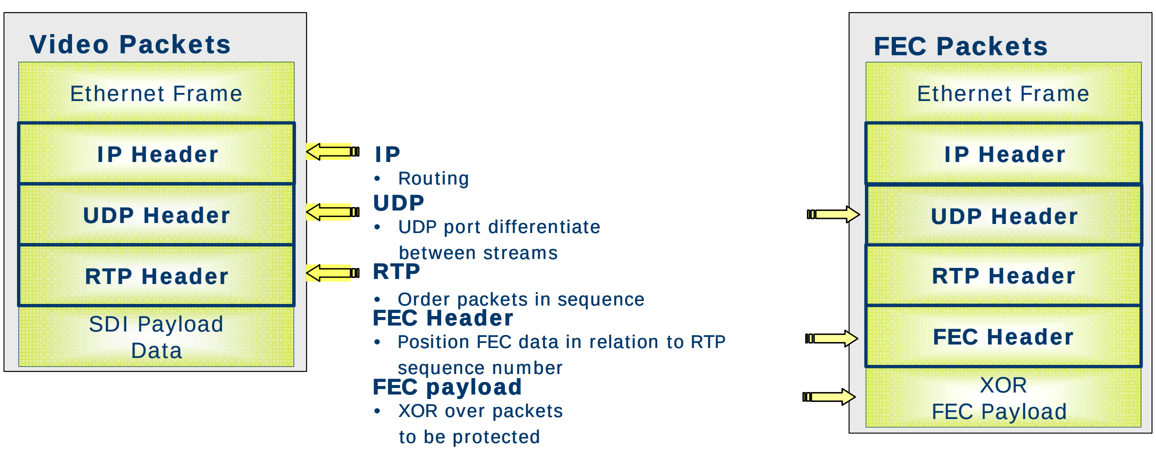
November 2005 and put into live service for launch of SportN on November 29th 2005.



Figur 1. SportN System Architecture

Pro-MPEG CoP 4 protocol

Pro-MPEG CoP 4 uses RTP/UDP/IP protocol layers. UDP does not include retransmission of lost packets as that would likely create congestion for CBR streams at this rate. In order to be suitable for broadcast television a Forward Error Correction (FEC) layer is included in CoP4. The maximum length of an Ethernet frame is 1500 bytes. This fits well with splitting a video line into two parts and leads to an IP packet rate of 31250 packets/s for 25 Hz frame rate. The error correction is derived from RFC 2733, and extends to protect against burst loss. The connection between video packets and error protection packets are by RTP sequence number. As the sequence number is incremented for every packet it is possible to re-order out-of-order packets as well as to remove duplicated packets.



Figur 2: Pro-MPEG CoP 4 protocol stack. The maximum size of a normal Ethernet frame is 1500 bytes. As PAL TV line is 2160 bytes a video line is split into two parts and leads to an IP packet rate of 31250 packets/s for 25 Hz frame rate.

The payload packets are ordered in a two-dimensional matrix with D row and L Columns where packets are entered by rows. The default FEC mode is to add a FEC packet for each column. It is then possible to regenerate 1 lost packet in each column. Optionally one may also add a FEC packet for each row. By performing an iterative regeneration of lost packets this mode makes it possible to correct multiple errors in a row or column.

Adapting implementation to network performance

An IP network is connectionless and the default mode of operation is by best effort forwarding. Variable traffic load will cause delay variations in the routers and overflow will result in packet drop. ITU has produced Y.1541 standards for “Network Performance Objectives for IP-based Services”. The highest performance was class 0 intended for voice services. In the latest draft for revision a new class 6 for video services has been added. The requirements for delay time variations are up to 50ms at the 10^{-3} quantile for class 0 and 50ms at the 10^{-5} quantile for class 6. For packet loss ratio the limits are set to 10^{-3} for and 10^{-5} respectively. The short time jitter will normally be around 1 ms, but rerouting may cause immediate changes in delay in the range of 5 to 20 ms. Broadcasters, on the other hand, cannot accept these amounts of loss and need to apply error correction. Considering 1 packet lost per day as an acceptable limit, the packet loss ratio after lost packet reconstruction must be less than $4\ 10^{-10}$.
The network performance can be improved by using MPLS or, as in this case, by setting up a VLAN in order to get isolation between the broadcast services and other services going through the same IP routers or switches. Alternatively or complimentary, the Type of Service (TOS) byte may be set to high priority in order to be expediently forwarded through the routers. Similarly the Class of Service byte may be applied if the Gateway is configured in VLAN mode.

Selection of FEC mode

The operator may select between the default mode using column FEC only or the optional dual mode with both column and row FEC. With the same matrix sizes the FEC computation delays are similar. The dual mode requires some more time in order to do the iterative correction, but this added time is short compared to the overall FEC delay. The FEC delay is linked to the time it takes to wait for the last packet in the column before being able to compute the FEC checksum + the time a FEC packet needs to be held back before it can be transmitted. The shortest delay is achieved by using what is referred to as CoP3 Annex A. In contrast to the annex B mode the FEC matrix is skewed and the FEC packets can be transmitted as the checksum is computed. With a rectangular matrix as in annex B it is required to hold back FEC packets in order to get a constant rate.

The selection between column and dual FEC mode is a trade off between overhead and FEC performance. Consider an overhead of 10% which in most cases is quite acceptable, the operator can select between a 10x10 matrix with column FEC or a 20x20 matrix with dual FEC. The FEC delay for the 10x10 matrix is about 3.5 ms, while 14 ms for a 20x20 matrix. At these settings a network packet loss ratio of up to 4 10^{-6} for column mode or 7 10^{-4} for dual FEC can be reduced to less than 1 error per day. As both modes have acceptable delay, it is advisable to use the dual FEC mode due to considerable better performance for error correction.

A curious effect of the dual mode is that the performance only lightly depends on the network packet loss ratio when this ratio is less than $10^{-4.}$ This is due to the fact that for low loss rates the dominant mode is one payload packet and the corresponding FEC column and row packets are lost. Of all combinations with 3 packets lost there are exactly LxD combinations that cannot be corrected. As the packet loss ratio is the ratio of uncorrectable packets divided by the number of packets in a matrix, the matrix size does not influence the share of uncorrectable packets for this particular mode. For higher loss ratios other combinations start to increase in influence and the improvement decreases, so that as the row lengths increase, the performance will get close to the column FEC mode with the same number of rows.

Selection of receiver latency

Receiver latency is the time that elapses from the moment an IP packet enters the receiver until it is being delivered as an SDI line on the output port. Delay is required to compensate for long time delay variations caused by, for instance, rerouting and short time jitter caused by variable time spent in the buffers of the routers on the way from transmitter to receiver. In addition it is required to add the time required for the FEC process. A possible setting of these parameters may be to allocate 15 ms for delay and jitter. The total delay then ends up being 18ms for a 10x10 matrix and 28ms for a 20x20 matrix. Packets that have a longer delay than the allocated value will result in packet loss.
If the receiver latency is set too small the buffer will go below the matrix size and error correction can no longer be performed.

Selection of SDI output clock mode
The SDI data is clocked from he receiver output buffer by a stable 27 MHz clock. The clock tolerance for a studio signal is in the range of 3 to 0.1 ppm for various television systems. There are no real applications where it makes sense to lock the SDI output clock to a regenerated clock with this stability as the buffer filling will now be entirely linked to the network jitter. For instance, a 1 ms jitter decrease would lead to 270000 more bits in the memory. By increasing the output rate with 1 ppm the buffer will decrease with 270 bits/s and it will take 1000 s before the output is reduced to default setting. Furthermore all switching inside a studio requires that the signal is locked to the house sync. The total performance is achieved by integration of a framestore inside the gateway as the frame skip and frame repeat can then be done in the best position. A hysteresis

needs to take the jitter into account in order to avoid short interval skip/repeat sessions. Media Netwerk decided to lock the SDI output to their studio sync.

Receiver error concealment.

Even with FEC running, some packets containing video data may be lost. As the output signal needs to be correct at all times, the receiver will recover lost packets by inserting data from the previous line or frame. In most cases this will only create small if any distortions to the picture. This is in contrast to MPEG-2 transport streams, where an error is likely to cause distortion over several blocks for a number of video frames.

Experience achieved

The SDI connection was put into service on November 29th 2005 and has been running with only minor problems. Currently FEC has not been used and lost packets have been compensated by error concealment. For the initial 45 days of operation the number of lost packets reported is 52 for the main link from NRK and 236 for the return link. The packet loss ratio for the main link is acceptable, but for the return link FEC is required in order to be within a target of less than 1 error per day. During the same period there have been 6 instances of sequence count errors. These are caused by packet loss indicating that average burst loss is in the range of 9 and 26 respectively for the forward and return path.

Using VLAN, as in this case, is a simple way to set up permanent fixed routes between studios. We also got a demonstration of one of the drawbacks of using layer 2 when the SDI signal was broadcast on the complete network as the SDI transmitter was trying to locate the receiver. The 300 Mbit/s signal put a very effective stop to the 100BT links in the system. The SDI link is now placed on a separate VLAN. The system will be converted to layer 3 routing as this reduces the risk of flooding and looping

About T-VIPS

Headquartered in Oslo, T-VIPS AS is a Norwegian technology company with new products and solutions for the growing professional Video over IP transport market. T-VIPS provides solutions for broadcast contribution, studio-to-studio media exchange, in-house signal distribution and routing, post-production, live event coverage and primary distribution. The company is funded through investments from the leading Scandinavian VC funds Northzone Ventures and Selvaag Venture Capital.

For further information, please visit: www.t-vips.com

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