Interoperability of FM Composite Multiplex Signals in an IP based STL

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Abstract - *The emergence of high bandwidth IP network connections is an enabler for the transport of the FM composite multiplex (MPX) signal in a Studio-to-Transmitter Link (STL). When architecting a STL network topology, transporting a MPX signal vs baseband FM components has advantages in terms centralized distribution and control. Two methods of MPX interconnection are digital and analog. Each present a tradeoff in terms of required network bandwidth, signal quality, and compatibility. Digital MPX over AES offers the possibility of an all-digital processing chain, while analog MPX can offer greater flexibility and compatibility with legacy equipment. With the recent emergence of digital MPX, there is a need for bridging and interoperability between newer digital MPX equipment and older analog FM plant MPX infrastructure. We review various MPX STL topologies, examine where various elements can be located, and what the networking requirements are for robust signal transport. Use cases are presented to illustrate differences between all analog, all digital, and hybridized analog/digital FM MPX STLs and the tradeoffs they present with respect to compatibility and network bandwidth, scalability, and delay.*

BACKGROUND

In FM broadcasting, the multiplexed signal (MPX) contains multiple components such as the left plus right audio, the left minus right audio, a 19 kHz stereo pilot tone, and a Radio Data System (RDS) signal. The left minus right audio is modulated onto a 38 kHz subcarrier (locked to the 2nd harmonic of the stereo pilot tone). RDS data is digitally modulated onto a 57 kHz subcarrier (locked to the third harmonic of the stereo pilot tone) and used to carry low bit rate (1187.5 bps) RDS metadata. In addition, Subsidiary Communications Authorization (SCA) channels may be generated which are modulated onto higher subcarriers, 67 kHz and 92 kHz. These services are low bandwidth (less than 8 kHz) and typically are audio services. At the receiver, the stereo pilot tone is used as a timing reference to generate the 38 kHz sub-carrier for the stereo demodulation process.

The MPX composite signal frequency spectrum is shown in Figure 1. The frequency spectrum bandwidth varies depending on components carried, but at a minimum, with RDS, it is 60 kHz and can be up to 99 kHz with two SCA channels.

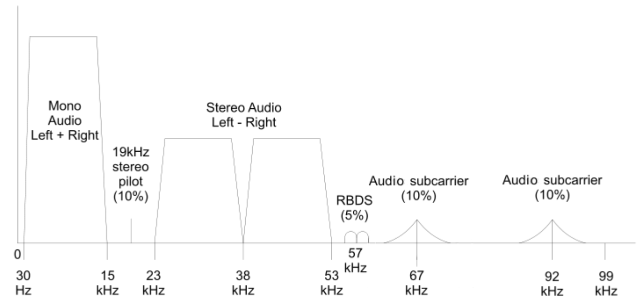

FIGURE 1 FM MPX FREQUENCY SPECTRUM

THE FM TRANSMISSION CHAIN

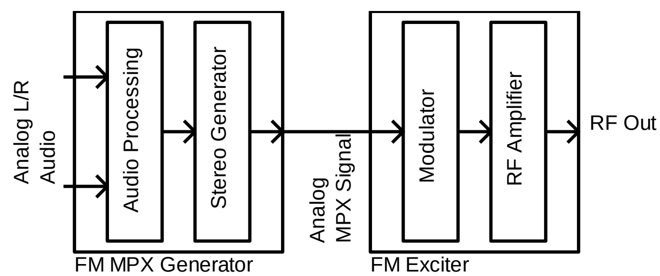

FIGURE 2 FM SIGNAL PROCESSING CHAIN

Figure 2 shows a common FM processing chain. The audio channels undergo audio processing which performs functions such as over-shoot limiting, frequency limiting (typically to less than 15 kHz), and pre-emphasis. After audio processing, the stereo generation or MPX encoding process occurs. Any of these processes can be performed in the digital or analog domain. The resultant analog MPX baseband signal feeds into an FM exciter.

To reduce analog processing, the stereo generation process can be performed entirely in the digital domain. Such digital MPX processing can produce a digital discrete time representation of an MPX signal formatted for interconnection using AES3 (also known as AES/EBU). AES3 is a standard for interchange of digital audio information and is well-established and well-supported in the broadcast industry. AES3 encapsulates two 24-bit channel audio samples into two 32-bit sub-frames. The remaining 8-bits of each sub-frame are used for metadata, parity checking, and synchronization information. The two sub-frames are combined into one 64-bit frame. MPX over AES is carried on one of the two AES3 channels, typically the left channel, at a sample rate of 192 kHz. Due to the Nyquist frequency, MPX over AES is band-limited to approximately less than 80 kHz so the FM sub-carrier at 92 kHz is not supported.

To support the full FM composite spectrum of 99 kHz – needed when using the 92 kHz subcarrier or second SCA channel – MPX over AES3 can utilize both the left and right channel to implement a 384 kHz sampling rate. In this case, samples are multiplexed between the left and right channels in an odd-even sequential arrangement. If the 384 kHz sampled composite spectrum is less than 96 kHz, then either the left or right channel can be used alone. In which case, they are compatible with 192 kHz MPX over AES sampling. If the 384 kHz sampled spectrum contains energy from 96 to 99 kHz, then aliasing will occur in both the 192 kHz left and right channels between 93 and 96 kHz.

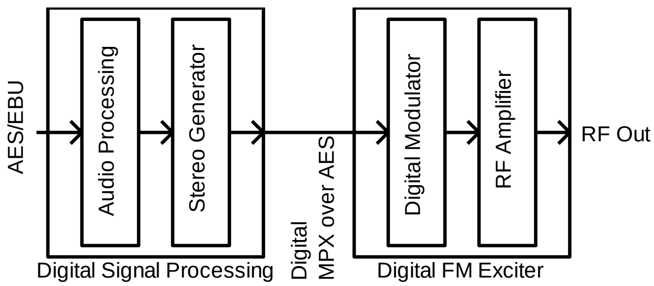

FIGURE 3 ALL DIGITAL FM SIGNAL PROCESSING CHAIN

Figure 3 shows an all-digital FM processing chain with baseband audio in the digital domain (AES3) feeding into an audio processor and MPX over AES feeding the FM exciter. The MPX signal is in the digital domain and embodies the components necessary for FM broadcast. Digital MPX offers higher RFI immunity, simpler connections and signal distribution than analog MPX.

MPX STL

A FM Studio-Transmitter Link (STL) using digital telecommunications such as T1/E1 or IP can have several topologies as shown in Figure 4. One common topology is where left and right audio are transported from the studio to far-end transmitter site using an audio codec. The stereo generation is done at the transmitter site. Other FM baseband information such as RDS data and SCA audio can be transported in the same audio codec assuming it supports all those additional data interfaces. Another possible STL topology is transport of the analog MPX signal itself using an MPX codec. This codec performs analog to digital conversion (ADC) at the studio and digital to analog conversion (DAC) at the transmitter.

The MPX over AES signal can also be transported on some audio codecs. In this case, the codec needs to support 192 kHz sampling rate and transports the information transparently (no sample rate conversion or audio compression).

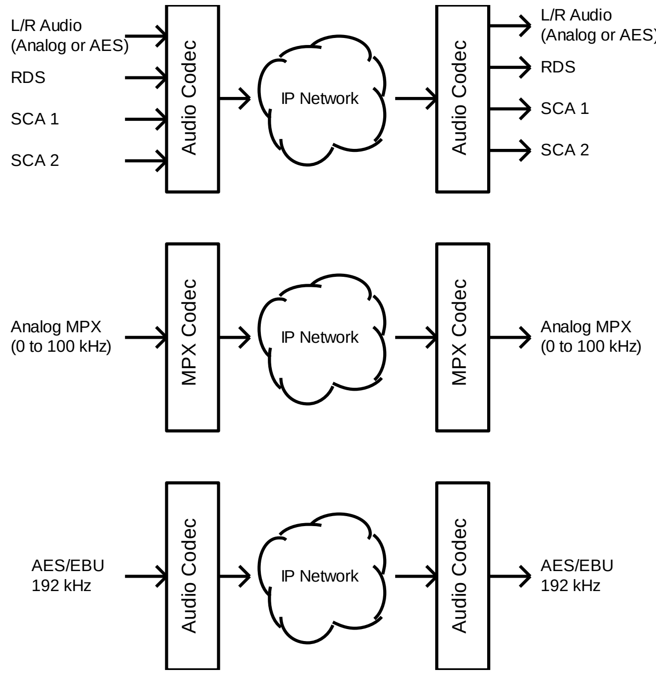

FIGURE 4 STL TOPOLOGIES

STL transport of MPX can have several advantages over baseband audio transport. The MPX generation process (stereo generation, RDS, and SCA modulation) is centralized and controlled at the studio site. In the case of multiple transmitter sites, the MPX generation is done once rather than being distributed out to the transmitter sites. With MPX over AES, a complete digital processing chain is preserved with no additional analog processes required.

The analog MPX signal can also be transported with a codec where the analog MPX signal is digitized and processed by the codec. Such a codec can support different sampling rates and sampling word sizes (16, 20, and 24-bit). The sampling rate can be adjusted to transport the stereo audio plus RDS, or stereo audio plus RDS plus one or two SCA channels. This type of flexibility can adapt the codec to best suit the bandwidth of the STL network connection.

DIGITAL MPX OVER A STL

The digital MPX (MPX over AES) signal can be transported over an STL in a transparent mode (Figure 5). In this method, the STL system performs an end-to-end, bit-by-bit transport of the signal with no alterations. It is only necessary to transport the AES3 24-bit left channel sample word across the STL. Other AES3 data such as parity, synchronization, and metadata can be regenerated at transmitter site to save network bandwidth. One channel of 192 kHz, 24-bit words has a data rate of 4.608 Mb/s.

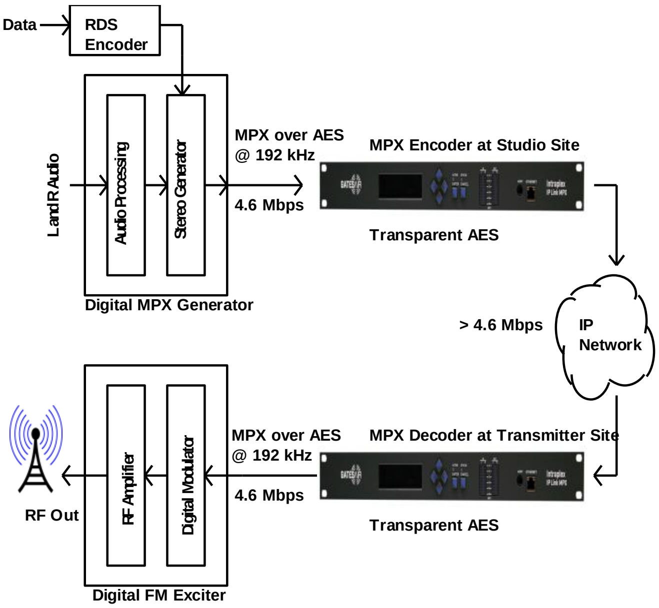

FIGURE 5 DIGITAL MPX STL USING TRANSPARENT TRANSPORT

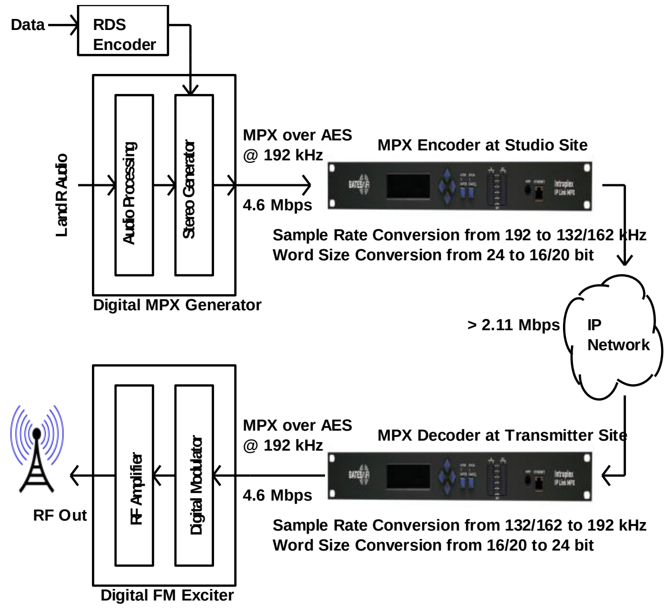

FIGURE 6 DIGITAL MPX STL WITH REDUCED BANDWIDTH

To reduce the network bandwidth requirements for MPX over AES STL transport, techniques such as sample rate conversion and word size reduction can be used. Sample rate conversion is the digital signal process of changing the sampling rate of a discrete time signal to obtain a new discrete representation of the underlying continuous signal. The 192 kHz sampled AES signal can embody an MPX signal of up to 96 kHz. Oftentimes the MPX signal does not contain information up to this frequency. For example, if the MPX signal contains stereo audio and RDS only, then the frequency content is up to 60 kHz. If the MPX signal contains in addition to this a single SCA channel (sub-carrier at 67 kHz) then its frequency content is up to 75 kHz. In these cases, the AES3 signal can be sample rate converted to a lower sample rate such as 132 kHz (stereo audio plus RDS) or 162 kHz (sub-carrier at 67 kHz) without loss of information. Modern state-of-the-art sample rate converters have excellent performance. With 24-bit samples, the THD and dynamic range can be greater than 125 dB with a near constant group delay and an amplitude vs frequency response characteristic close to the Nyquist rate. Using sample rate conversion in an STL, the AES over MPX signal can be sample rate converted down at the studio site and then sample rate converted back up to 192 kHz at the transmitter site. Figure 6 shows this process in an STL system.

Another useful effect of sample rate conversion when down sampling is frequency filtering during the decimation process. When using 384 kHz MPX over AES sampling, the FM composite energy from 96 to 99 kHz is aliased into the 192 kHz left and right channels from 93 to 96 kHz. If the 192 kHz left or right channel is down sampled below 186 kHz, then the aliased energy will be effectively removed – a useful property when interoperating between 384 kHz and 192 kHz AES over MPX systems.

AES3 defines a 24-bit word size. The theoretical maximum achievable dynamic range for a digital representation of an analog signal using uniform quantization is 6 dB per bit. So 24-bit quantization can provide 144 dB of dynamic range. As a practical matter, this is more dynamic range than can effectively be generated by the FM stereo generator or utilized by an FM exciter. In most instances, the word size can be reduced to 16-bits without any loss of quality. With 16-bit sampling, the dynamic range is still a robust 96 dB. Further reduction in word size beyond this is possible with some tradeoff in quality. Some commercial MPX codecs have sampling word sizes as low as 12-bits. These smaller word sizes can allow a MPX codec to keep the network bandwidth below 1.984 Mb/s which is a requirement for E1 type telecommunication transmission. Whenever reducing sample word size, rounding must be used on the two's complement PCM sample, otherwise a ½ bit DC offset will result.

Using sample rate conversion and sample word size reduction, the AES over MPX STL network bandwidth can be considerably reduced. Using 132 kHz sampling and 16-bit word conversion the network payload bandwidth is 2.11 Mb/s vs 4.6 Mb/s for AES3 transparent mode.

MPX BRIDGING

Analog MPX is compatible with most existing FM plant infrastructure while digital MPX (MPX over AES) is a relatively new operating standard first introduced in 2013 and requiring acquisition of compatible equipment. For interoperability between the two, the MPX signal can be bridged from analog to digital or from digital to analog. This is useful when interoperating between older FM equipment not supporting digital MPX and newer FM equipment which does. For example, an older FM stereo generator at the studio can interoperate with a new FM exciter through a MPX bridging device. Conversely, a new FM stereo generator can interoperate with an old FM exciter through a MPX bridge. This hybridized bridging device sits between digital and analog domains of operation. The bridging can be a device between co-located FM equipment or the bridging function can span an STL when it is integrated into a codec. In the later the MPX codec at both the studio or transmitter site must support dual analog and digital MPX interfaces. Figure 7 shows the MPX STL bridging between a digital MPX stereo generator and an analog FM exciter. Figure 8 show the MPX STL bridging between an analog MPX stereo generator and a digital FM exciter.

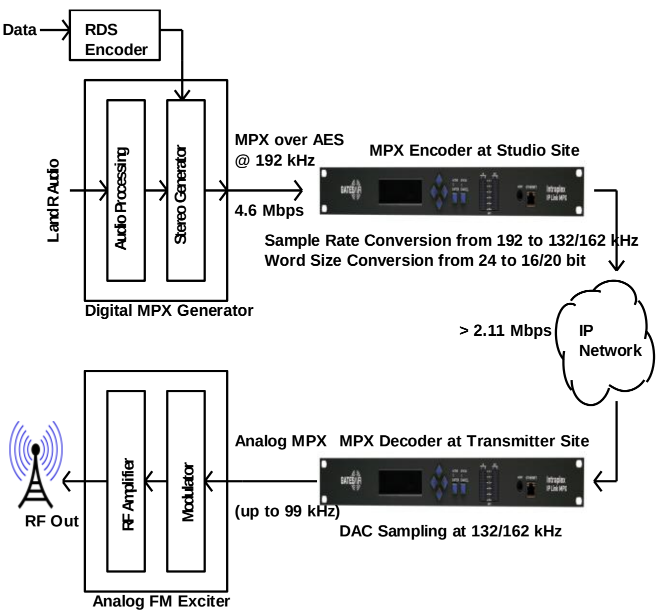

FIGURE 7 MPX STL BRIDGING BETWEEN DIGITAL AND ANALOG

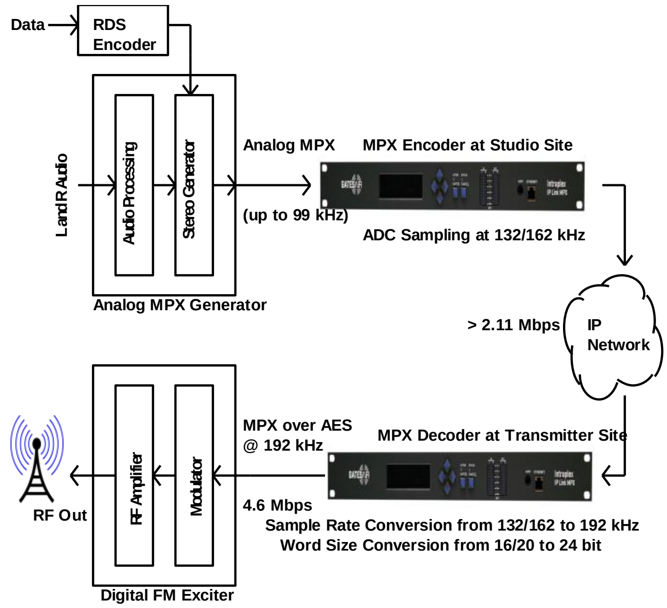

FIGURE 8 MPX STL BRIDGING BETWEEN ANALOG AND DIGITAL

RF SINGLE FREQUENCY SIMULCASTING

RF single frequency simulcasting uses multiple, geographically disperse RF transmitters operating on the same carrier frequency, modulating the same program material. By using multiple transmitters, geographic RF coverage area is expanded. The region where a RF receiver can pick up multiple signals feeds from multiple transmitters is the overlap region. In an audio broadcasting application, a RF receiver in this region will simultaneously demodulate audio programming carried on multiple RF carriers. In this region, the modulation should be closely phase aligned from the multiple transmitters to provide the best receive quality.

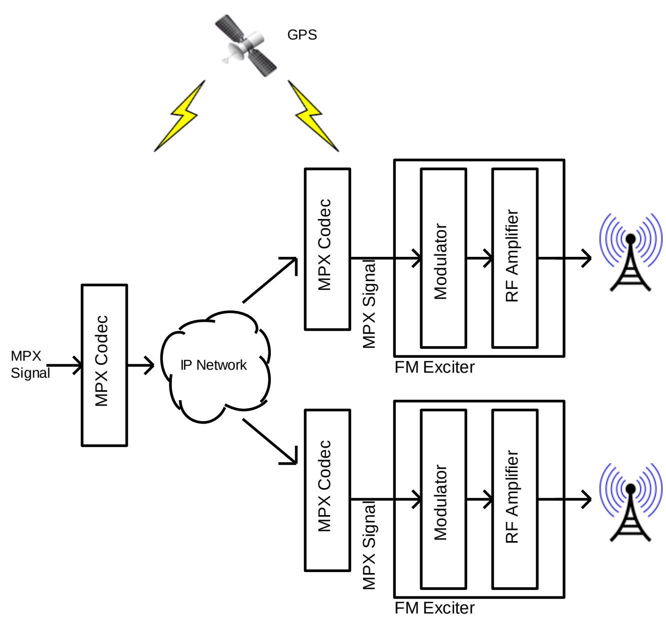

FIGURE 9 FM SIMULCAST SYSTEM

MPX vs baseband audio transport over an STL can offer advantages when deploying a RF simulcasting system (Figure 9). In a baseband audio transport STL scenario, the audio codec process undergoes a precision delay process unique to each geographically diverse transmitter site so that each FM stereo generator is presented the audio “in-phase” with respect to all the other FM generators in the simulcast system [1]. However, other MPX sub-carrier components (RDS and SCA) may not undergo the same precision delay process resulting in out-of-phase RF transmission for these components. In an MPX transport STL scenario, the entire MPX can undergo a precision delay process so all components are in-phase with each other.

When using transparent AES transport over an STL, consideration must be made toward timing compatibility. In transparent mode, the data clock rate is preserved in the entire encode and decode process. For the precision delay process to function properly, the data clock must be derived from a global source such as GPS. In most cases, transparent transport is unnecessary, and the AES transport can be “re-timed” using sample rate conversion in which case GPS can serve as the re-timing clock source.

MPX STL Networking Considerations

The wide area network (WAN) payload bandwidth requirements for transporting an MPX signal varies based on sampling rate and word size. MPX over AES has a data rate of 4.602 Mb/s if AES3 overhead information such as synchronization, metadata, parity, etc. is not transported. This signal contains stereo audio, RDS, and one SCA channel. This data rate can be reduced when down sampling from 192 kHz to a lower sampling rate using sample rate conversion. Further data rate reduction can be achieved by reducing the word size from 24-bits to 20 or 16 bits.

Sample Rate, kHz	Sample Size, bits	Bandwidth, kb/s
132	16	2112
132	24	3168
162	16	2592
162	24	3888
192	16	3072
192	24	4608
216	16	3456
216	24	5184

Table 1 MPX Data Rates

Transport of an analog MPX signal has flexibility in the payload data rate because of settable options for sampling rate and sample word size. The sampling rate selection is made based on the services needed to be carried across the WAN. For example, 132 kHz sampling can carry stereo audio and RDS data, 162 kHz sampling can carry an additional SCA channel, and 216 kHz sampling can carry the entire 99 kHz MPX spectrum. Sample word size defines the minimum resolution and dynamic range of the signal. In most cases, 16-bits can be used and gives good performance with 96 dB of dynamic range. Higher resolution sample sizes like 20 and 24 bits may also be used. Depending on the sampling rate and the sample size, the payload bandwidth will vary, Table 1 provides a summary for some possible combinations.

Usage of IP based WAN in broadcast application is rapidly proliferating. IP based networks provide advantages in both reduced operational cost and flexibility for site to site interconnection. The packetized transport of the MPX signal does add additional overhead for packet headers as well as delay associated with the packetization process. The packetization process needs to accumulate samples in a packet buffer prior to transmission. A higher number of samples in a packet results in lower overhead, lower packet rate and higher associated packetization delay. Conversely, the lower number of samples per packet translates to higher packet rate, higher overhead, and lower delay.

The overhead in a packet is associated with the packet header. Real Time Protocol (RTP) over UDP is the most common encapsulation method for transport of real-time media. RTP provides both time stamping and sequence numbering to detect duplicate packets and re-order out-of-order packets. The overhead associated with RTP/UDP headers when transported over IPv4 networks is 40 bytes per packet. Taking the per packet overhead in to account, Figure 10 shows IP network bandwidth required for several MPX transport options based on packet interval (1 and 3 mS), sampling rate (128 to 216 ksps) and sample size (16 to 24 bit). For the most bandwidth efficient option, the packet size should be just under the Maximum Transmission Unit (MTU) of the network, for Ethernet this is 1500 bytes.

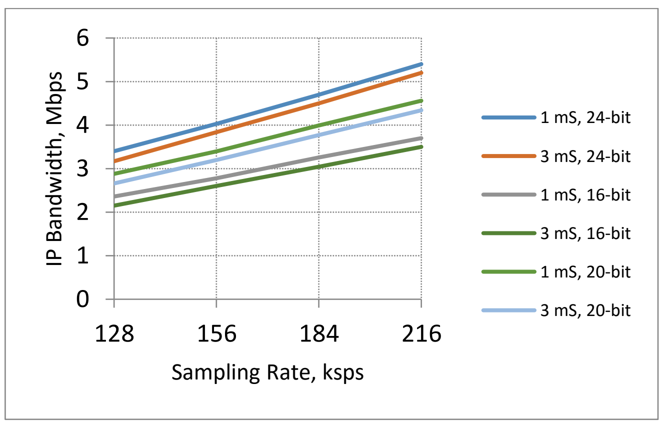

Figure 10 IP Bandwidth vs Sampling Rate

There is tradeoff between delay and bandwidth efficiency, while keeping the packet size to be less than the MTU of the network path to avoid fragmentation. Fragmentation of IP packets can cause additional delays for re-assembly and traversing Network Address Translation (NAT) devices. In summary, you want to have high packet interval without causing IP fragmentation. This will ensure bandwidth efficiency and reduces the packet rates which helps in avoiding congestion within the IP networking nodes [2].

CHALLENGES WITH IP TRANSPORT

The IP based WAN pose various impairments such as packet loss, jitter, loss of network connection, etc. A MPX stream requires constant WAN bandwidth. This contrasts with audio using lossy compression (AAC, MPEG) in which case the encoder can dynamically adjust the bitrate based upon the congestion state of the WAN connection. Packet losses can also have a more pronounced effect on a MPX signal than with audio. For example, with compressed audio, concealment techniques can work effectively. The audio codec keeps an ongoing measurement of the spectral image of the audio. The codec already has a time-to-frequency domain transform as part of its perceptual coding function. When a packet loss is detected, a synthetic replacement can be created by using the spectral values in the preceding and subsequent packets [3]. A MPX stream is a lossless PCM encoding method, so no spectral information is computed. A lost MPX packet results in missing data for that packet interval. So when transporting MPX over IP, it is recommended to subscribe to a managed Internet Service Provider (ISP) connection which guarantees bandwidth and keeps packet loss to a minimum. The Packet Delay Variation (PDV) or jitter is less of a concern on managed connection because it can be mitigated by adjusting the size of the receive jitter buffer at the decoder. This adjustment can be either manual or the system can dynamically measure the jitter and adapt the buffer to the appropriate size.

PACKET LOSS AND CONNECTION LOSS MITIGATION

Even when subscribed to a managed ISP connection, there may be packet losses during certain times that must be dealt with. In addition, due to the complexity of the IP networks, connections are more susceptible to complete path failure when compared to the legacy T1 connection. Packet losses can be reduced using one or more combinations of these recovery techniques: RTP level Forward Error Correction (FEC) and/or redundant streams. Both techniques are designed to work in unicast (point-to-point) and multicast (point-to- multipoint) modes of packet transport.

RTP Level FEC

The RTP level FEC uses the concept of parity packets to recover lost packets at the receiver. It uses a matrix of RTP data packets as shown in Table 2 to generate the FEC packets. FEC packets are created using XOR of rows and columns of data packets for a 2-dimensional FEC. These FEC packets are then sent along with the data packets over the network to the receiver. The ratio of the number of parity packets to data packets is the bandwidth overhead of the FEC scheme. For example, the above scheme generates 8 parity packets for 16 data packets, hence it requires 50% additional bandwidth for the stream. In addition, the receiver needs to buffer up the enough packets based on the size of the matrix to effectively recover lost packets, hence there is additional delay associated with FEC based packet loss recovery as well. The FEC schemes have shown to be extremely effective for random packet losses, however if the packet losses occur in bursts and if the burst size is greater than number of columns in the FEC matrix, the effectiveness of FEC scheme starts to deteriorate.

	Col 1	Col 2	Col 3	Col 4	F(x)
Row 1	1	2	3	4	XOR(1,2, 3,4)
Row 2	5	6	7	8	XOR(5,6, 7,8)
Row 3	9	10	11	12	XOR(9, 10,11,12)
Row 4	13	14	15	16	XOR(13, 14,15,16)
F(x)	XOR(1, 5,9,13)	XOR(2, 6,10,14)	XOR(3, 7,11,15)	XOR(4, 8,12,16)	

TABLE 2 4X4 TWO-DIMENSIONAL FEC MATRIX

Redundant Streams

Redundant streaming is another technique that can be used for packet loss protection. In this scheme, duplicate packets are sent either over independent network paths – network diversity or on the same network using time diversity as shown in the Figure 11. With time diversity, the duplicate packets are sent with a programmable delay between them. Both these techniques are effective for burst packet losses. However, the network diversity technique has shown to give the best result provided the paths are truly independent. This technique also provides “hitless” protection against complete failure of a single path. One important point to understand with network diversity is that independent network paths have different delays and jitter characteristics that can be time varying and therefore the implementation must ensure that the receive buffering is always adapted to the optimal size to account for the longest delay and largest jitter among the network paths. When figuring out packet loss protection for a single network, the choice between FEC, time diversity, or combination of the two schemes should be based on analyzing the packet loss patterns. Of course, all of these protection techniques assume that the user has adequate network bandwidth and connections along with the network protection capabilities embedded within the MPX codec.

SUMMARY

The emergence of high bandwidth IP network connections enable the transport of the FM MPX composite signal in an STL. When architecting a STL network topology, transporting a MPX signal vs baseband FM components has

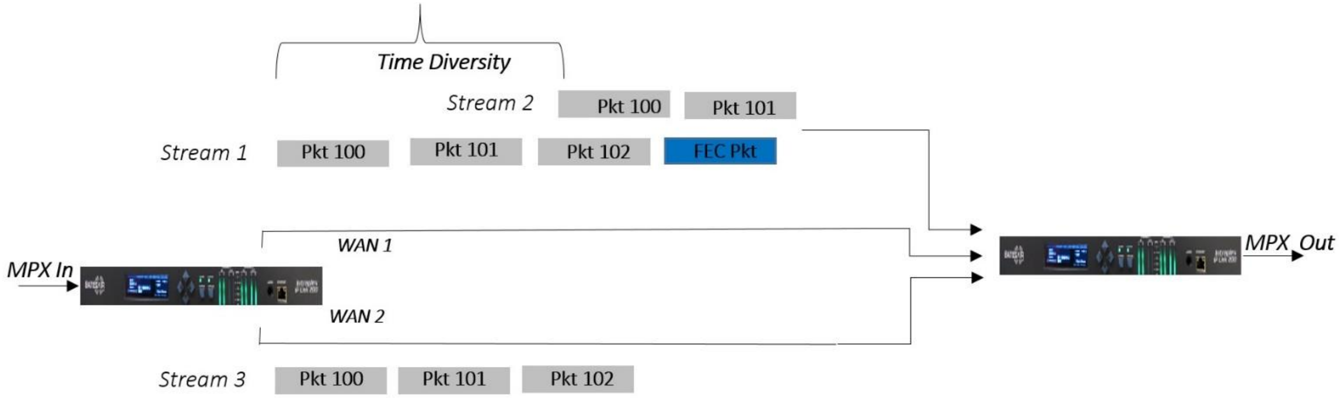

FIGURE 11 COMBINATION OF DIVERSITY AND FEC PROTECTION

advantages in terms centralized distribution and control and being able to effectively perform RF simulcasting of the RDS and SCA.

Two methods of MPX interconnection are digital and analog. Digital MPX over AES offers the possibility of an all-digital processing chain with high RFI immunity and simpler signal distribution, while analog MPX maintains compatibility with existing legacy equipment. Transport of the digital MPX signal over a STL can be performed transparently or with sample rate conversion and sample size reduction to reduce the required network bandwidth. The sample rate can be selected to support the FM subcarriers needed for transmission. Interoperability between analog and digital MPX is possible with a hybridized bridging device that interconnects between the two domains. This is useful when interoperating between older FM equipment not supporting digital MPX and newer FM equipment which does.

IP network impairments such as packet loss and jitter must be addressed for robust MPX streaming to be realized. Techniques such as jitter buffering, FEC, time and network diversity with redundant streaming can be utilized. These techniques, if applied in a systematic and intelligent manner can greatly improve the performance of MPX streaming over impaired IP networks.

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