



Distribution of the Analog or Digital FM Composite Multiplex Signal across IP Networks

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GatesAir's



Junius Kim
Hardware Engineer



Keyur Parikh
Architect / Software Lead

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Distribution of the Analog or Digital FM Composite Multiplex Signal across IP Networks

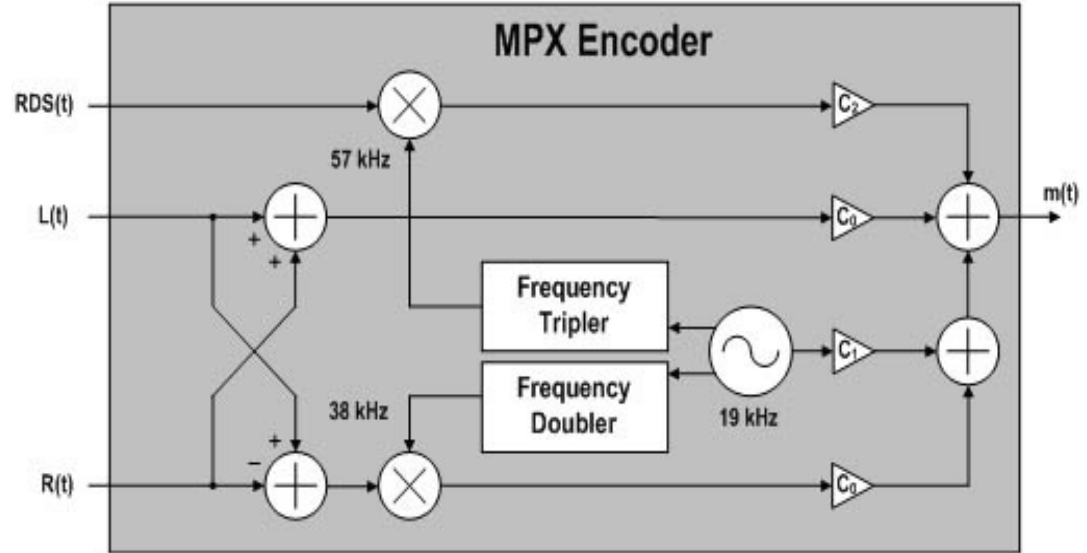
Junius Kim and Keyur Parikh

GatesAir

Mason, Ohio

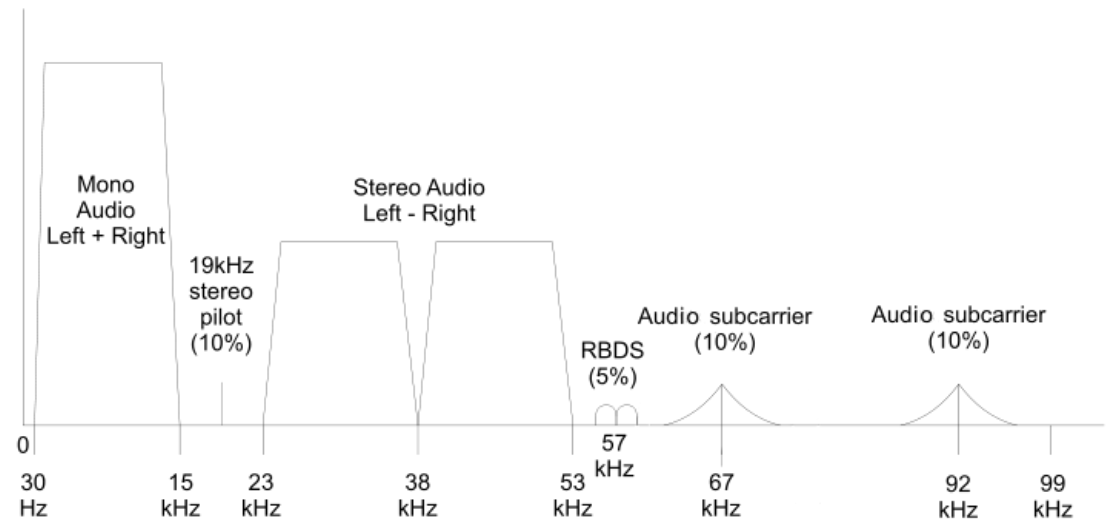
FM MPX Generation

- L+R
- 19 kHz pilot tone
- L-R – 38 kHz subcarrier
- Radio Data System (RDS) - low bit rate (1187 bps) digital data – 57 kHz subcarrier



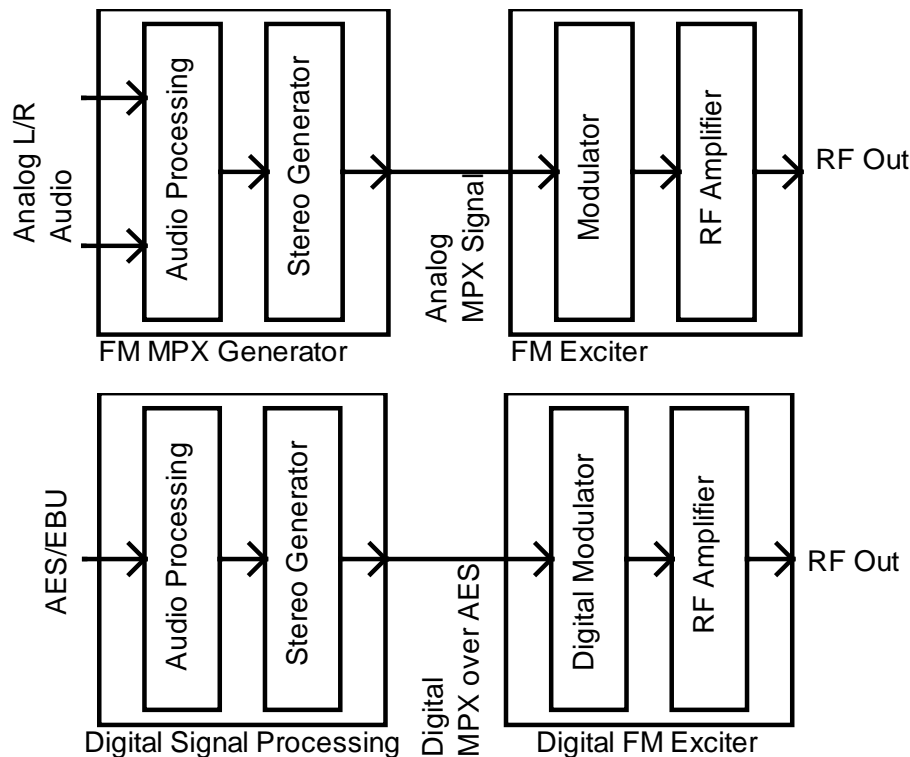
FM MPX Frequency Spectrum

- Pilot tone decoded at receiver
- Subsidiary Communication Authorization – Low BW audio
- 67 and 92 kHz Subcarriers
- FM MPX BW
 - 53 kHz: L and R audio
 - 60 kHz: plus RDS
 - 75 kHz: plus one SCA
 - 99 kHz: plus one SCA



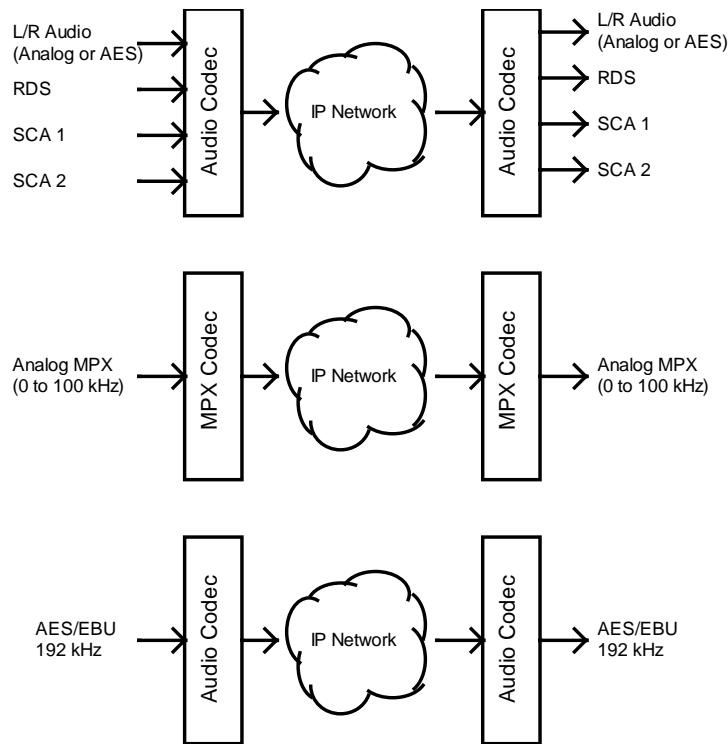
FM Transmission Chain

- FM Audio Processing
 - Frequency Limiting (< 15 kHz)
 - Overshoot Limiting
 - Pre-emphasis
- Stereo Generation
- FM Exciter
 - Analog MPX
 - Digital MPX over AES/EBU @ 192 ksps



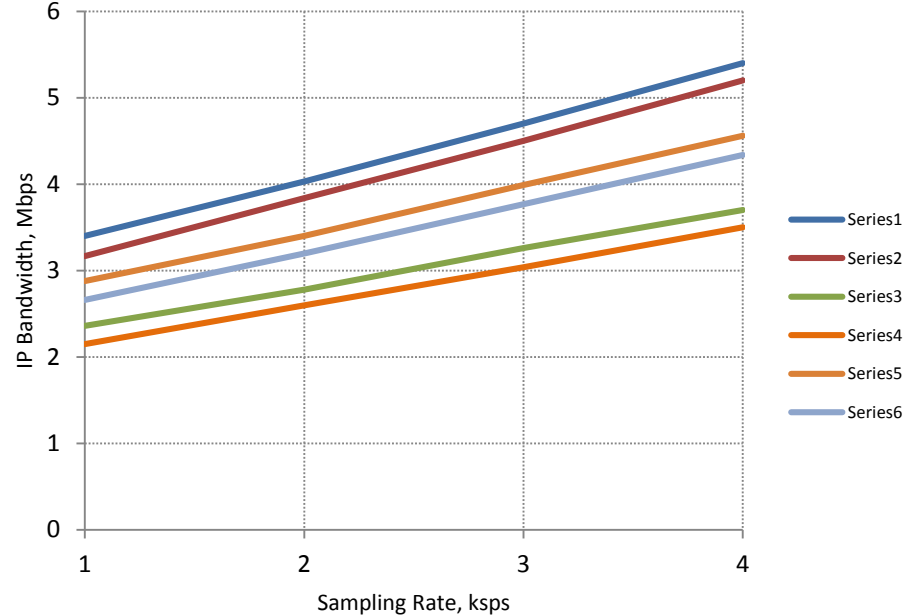
STL Topologies

- STL Transport
 - Baseband Audio
 - Analog MPX
 - Digital MPX

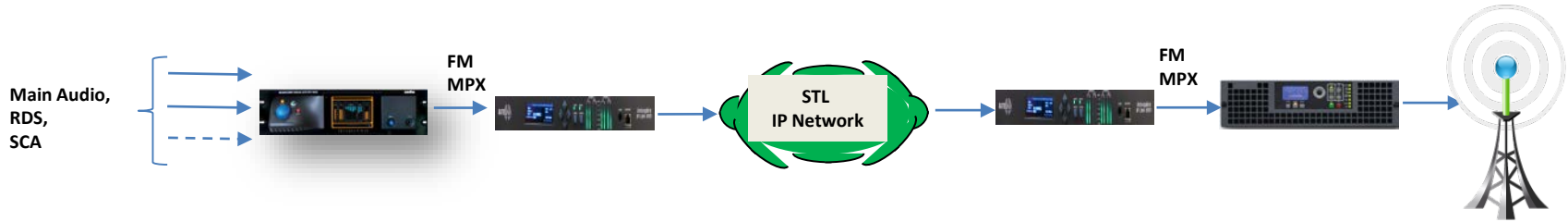


MPX STL Bandwidth

- MPX is linear PCM – uncompressed
- MPX over AES/EBU is 192 ksps @ 24 bit sampling, one channel – 4.6 Mbps
- Analog MPX sampling is 132 to 216 ksps
- MPX IP transport uses RTP
- IP RTP/UDP header overhead is 40 bytes
- Tradeoff between delay and packing efficiency



FM MPX over IP



- Benefits of FM MPX over IP vs Audio over IP
 - Enables baseband equipment (audio processor, stereo generator, RDS generator) to be located at the studio side
 - Reduces CapEx when distributing the same signal to multiple transmit sites
 - Simplifies operation for FM SFN
- However – MPX requires higher STL capacity than audio only transport
 - Audio is amiable to lossy compression – AAC, MPEG, opus, etc

MPX over AES IP STL

- Digital FM MPX – AES 192
 - AES/EBU interface with 192 ksps sampling rate to carry up to one SCA subcarrier
 - Uses one channel of AES/EBU interface
 - All digital chain

Data Rate Requirements:

Sample Rate/ Sample Size	Services	Packet Rate, pps	Packet Payload, bytes	Data Rate
192 ksps/24 bits	Audio + RDS + one SCA	500	1152	4.7 Mbps
192 ksps/16 bits	Audio + RDS + one SCA	333	1153	3.2 Mbps

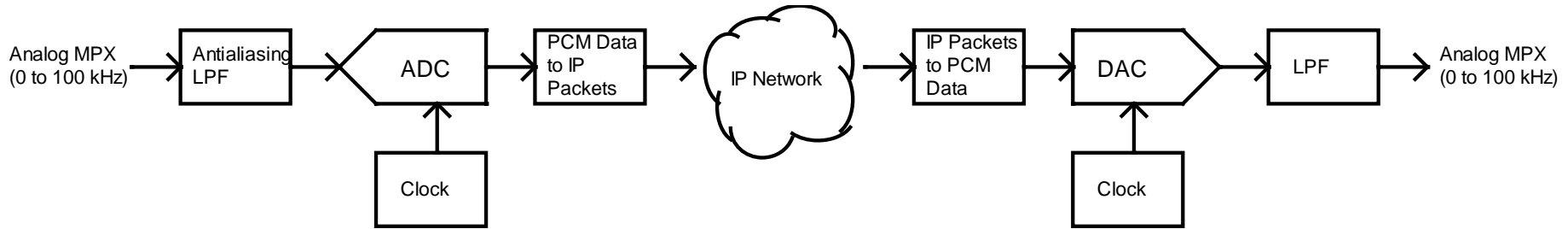
Analog MPX over IP STL

- Analog MPX
 - Flexible sample rate configuration based on FM MPX components to transport
 - Reduced data rate to carry audio + RDS vs MPX over AES @ 192 ksps
 - Flexible sample sizes: 16, 20 and 24 bits
 - Compatibility with existing infrastructure

Data Rate Requirements:

Sample Rate	Services	Packet Rate Range, pps	Data Rate Range
132 ksps	Audio + RDS	200 - 333	2.2 – 3.2 Mbps
162 ksps	Audio + RDS + SCA 1	333 - 500	2.7 – 4.2 Mbps
216 ksps	Audio + RDS + SCA 1 + SCA 2	500	3.6 – 5.4 Mbps

Analog MPX Codec STL



Studio Site

Transmitter Site

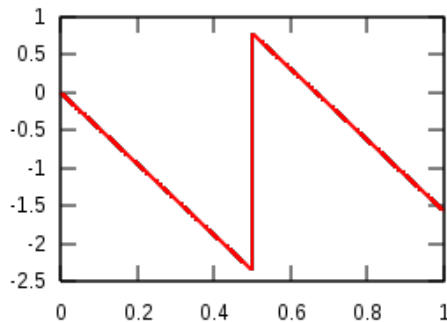
Analog MPX Codec Requirements

- Band from 0 to 53 kHz contains stereo audio
- Left: $2L = (L+R) + (L-R)$
- Right: $2R = (L+R) - (L-R)$
- Gain flatness of 0.05 dB across 0 to 53 kHz for >50 dB stereo separation
- Linear phase response

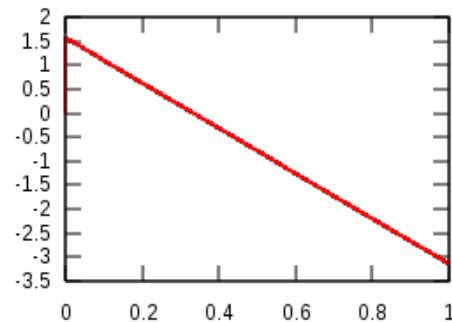
Linear Phase

- Linear phase = constant group delay
$$\tau_g(\omega) = -\frac{d\phi(\omega)}{d\omega}$$
- Use FIR filters for linear phase
- Use over-sampling at ADC to minimize effect non-linear phase response analog filters
- Use interpolation at DAC to minimize effect non-linear phase response analog filter

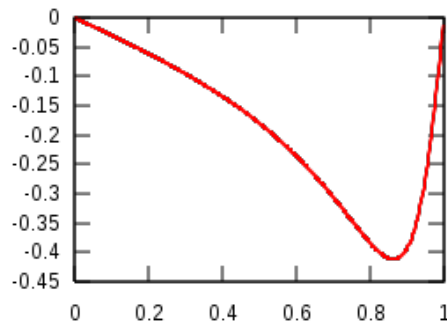
a) FIR Filter (Type II) having Linear Phase



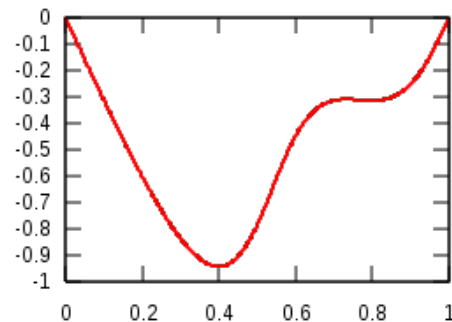
b) FIR Filter (Type IV) having Linear Phase



c) IIR Filter having Non-Linear Phase

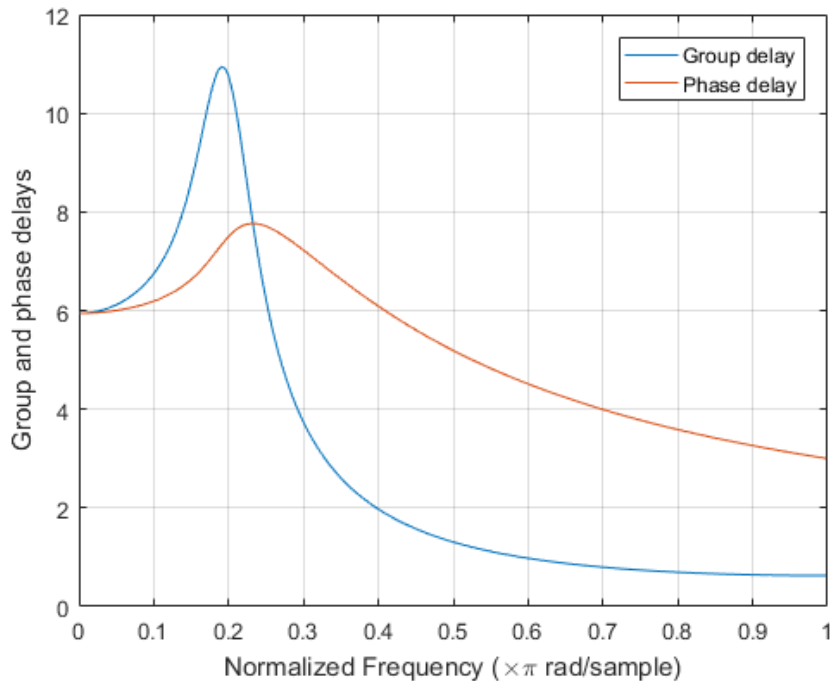


d) FIR Filter having Non-Linear Phase



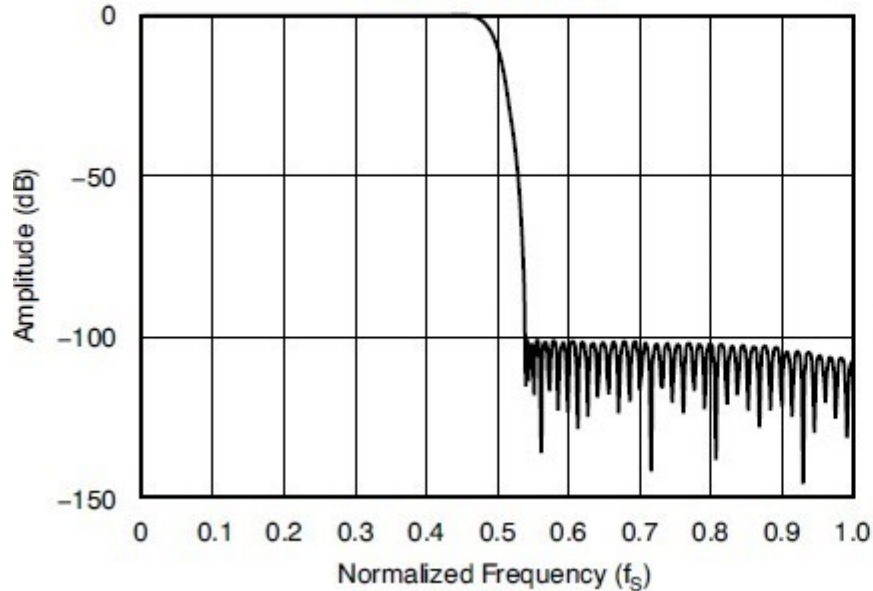
Butterworth Filter Bode Plot

- Flat gain response in the passband
- Non linear phase response – Moderate phase distortion

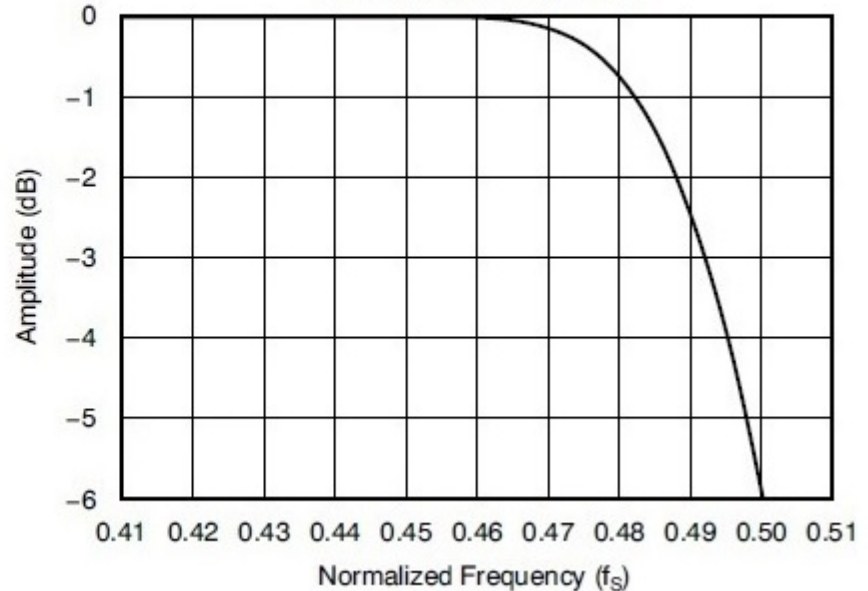


Example of ADC Characteristics

DIGITAL DECIMATION FILTER
Stop Band Detail



DIGITAL DECIMATION FILTER
Transition Band Detail

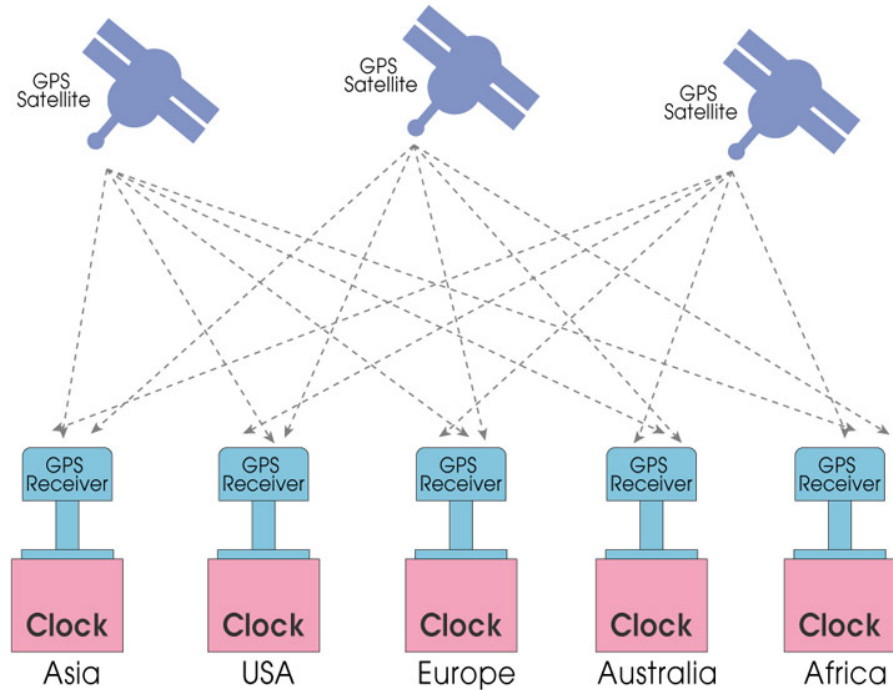


MPX STL Timing

- Ideally...
 - Analog MPX – DAC clock is same as ADC clock
 - MPX over AES – AES/EBU clock at transmitter is same AES/EBU clock at the studio
- In AES/EBU, clock is embedded with data
- MPX over AES across the WAN – tight coupling of data/clock is lost

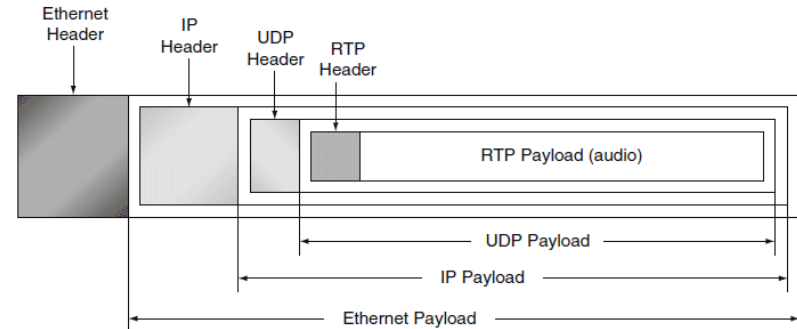
GPS Based Timing

- Precise common timing at geographically disperse locations
- GPS based receivers with 1 PPS and 10 MHz output
- GPS timing delivery using IEEE 1588 or PTP
 - Low phase “noise”, low jitter clock reference
- Generate AES/EBU or ADC/DAC clock using PLL



Timing Recovery

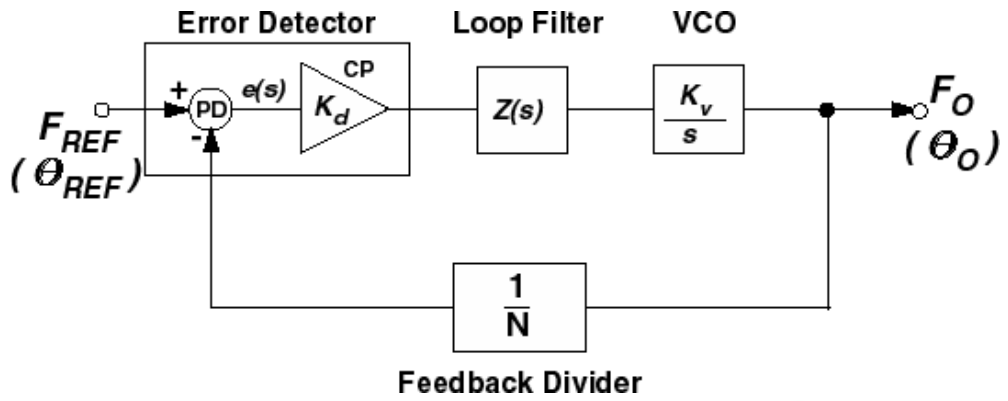
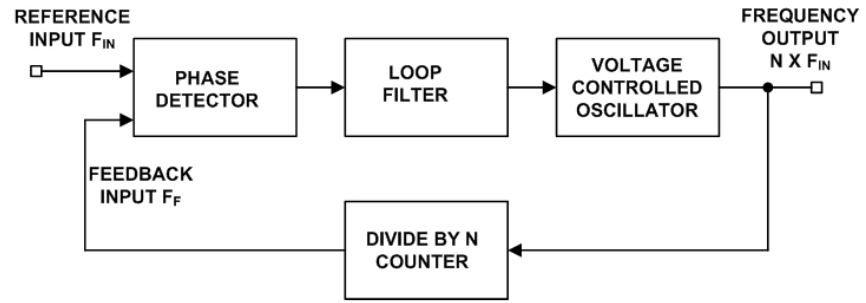
- If no common timing reference, then timing recovery from the RTP stream must be used
- Use RTP packet as time reference
 - Loose coupling of data/clock
 - High jitter and phase noise
- In FM, 19 kHz pilot tone tolerance is +/-100 PPM



- In AES/EBU - 1:1 ratio between clock and data
- In RTP – 1:10,000 ratio between clock and data

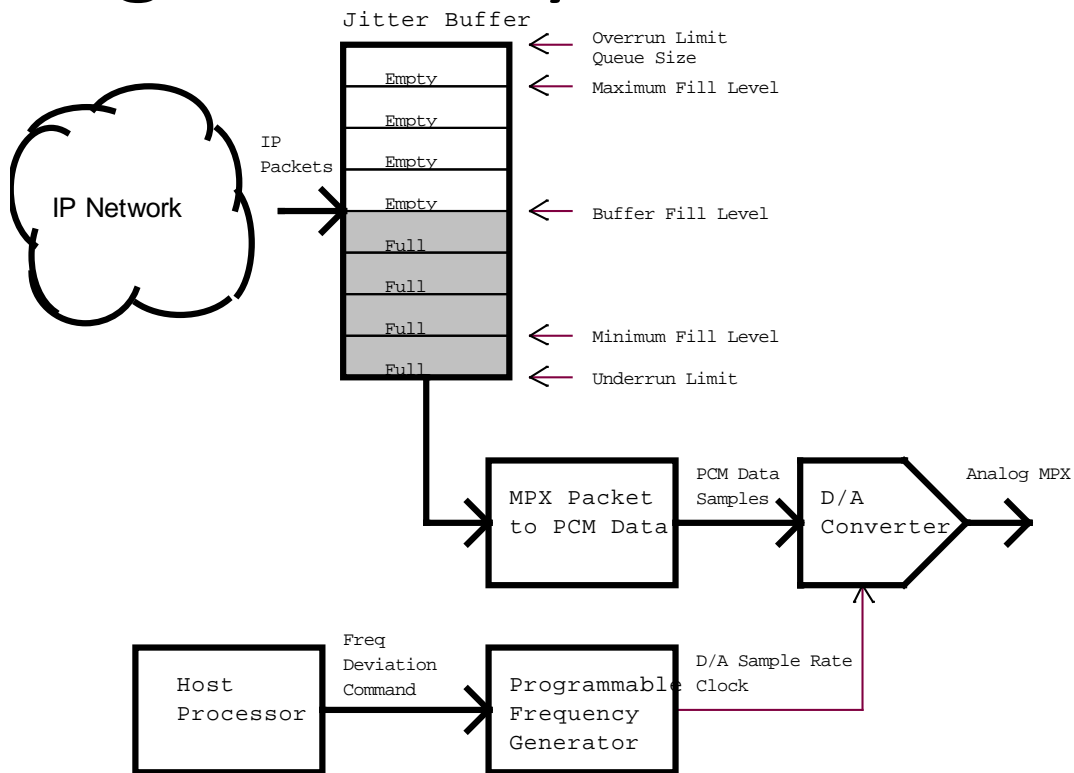
Phase Lock Loop

- Loop stability
- Loop filter response < reference frequency response



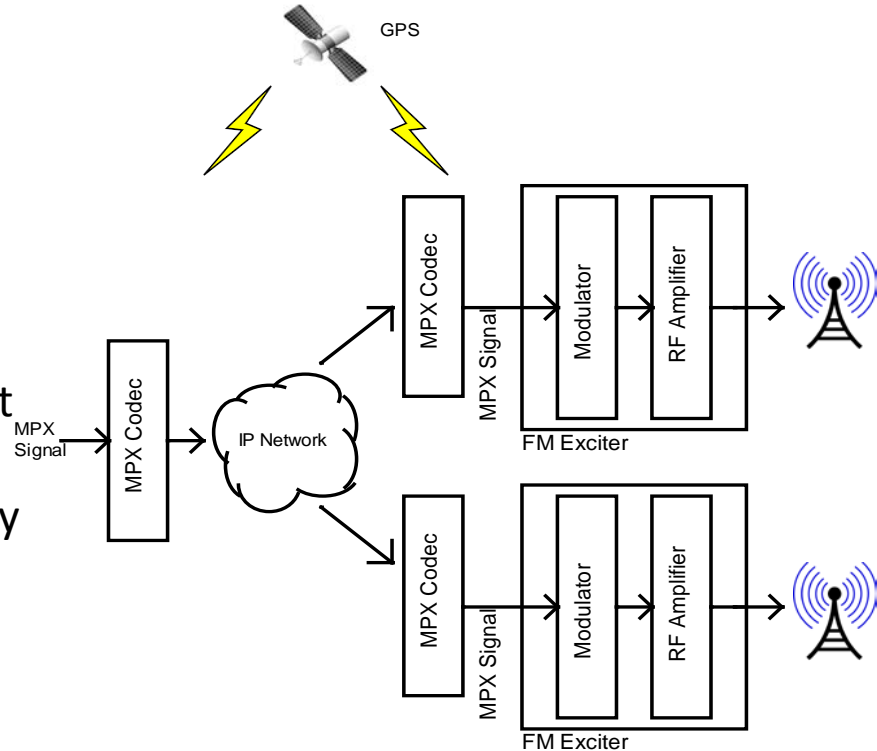
Timing Recovery

- RTP jitter buffer
- Timing recovery algorithm
- Adjust DAC clock to keep constant jitter buffer fill level



Single Frequency Simulcasting

- RF single frequency simulcasting uses multiple, geographically disperse RF transmitters operating on the same carrier frequency
- In simulcast, baseband signal undergoes a precision delay process
- MPX advantage vs audio only transport over STL
 - In MPX all components are equally delayed

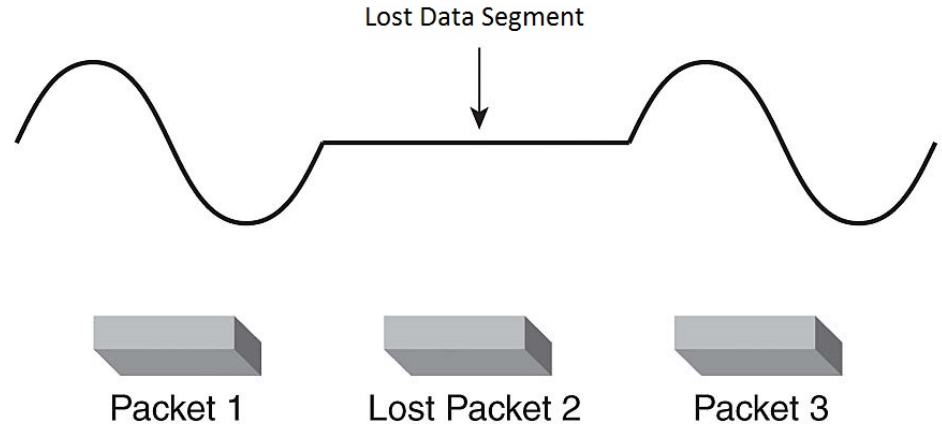


IP Packet Loss

- Causes of IP packet loss: route flapping, transmission errors, congestion
- Unmanaged vs. managed network services
- In audio - packet loss concealment methods: frequency interpolation, replaying previous frame
- In MPX – no standardized concealment methods
- For MPX, use correction techniques for packet loss mitigation

Packet Loss Effects

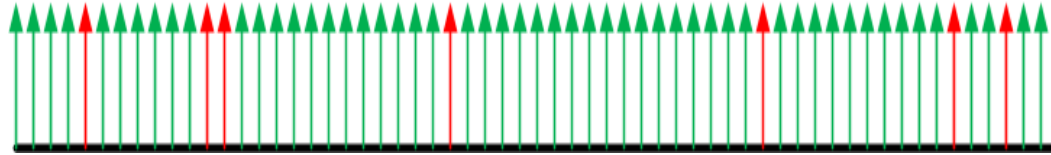
- Audio compression algorithms keep spectral information – fill in missing data segment from previous data – error concealment
- MPX codec method is PCM coding – no spectral information is computed
- MPX codec – no error concealment



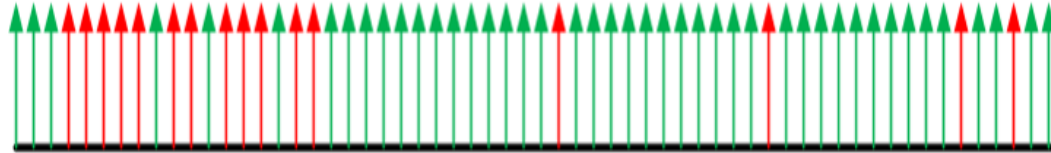
IP Packet Loss

- Random vs. Burst Packet Loss
- Random Losses
 - Uncorrelated
 - Appear to be spread out

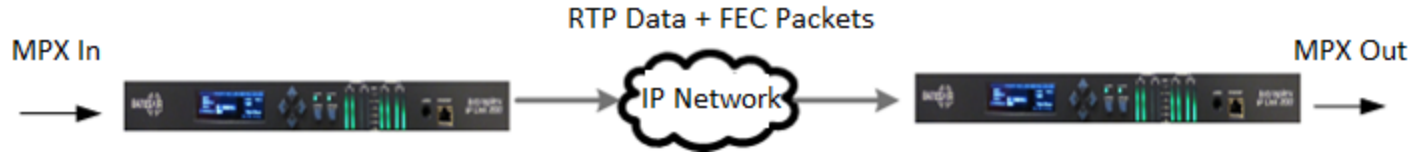
Random Loss



Burst Loss



RTP Forward Error Correction

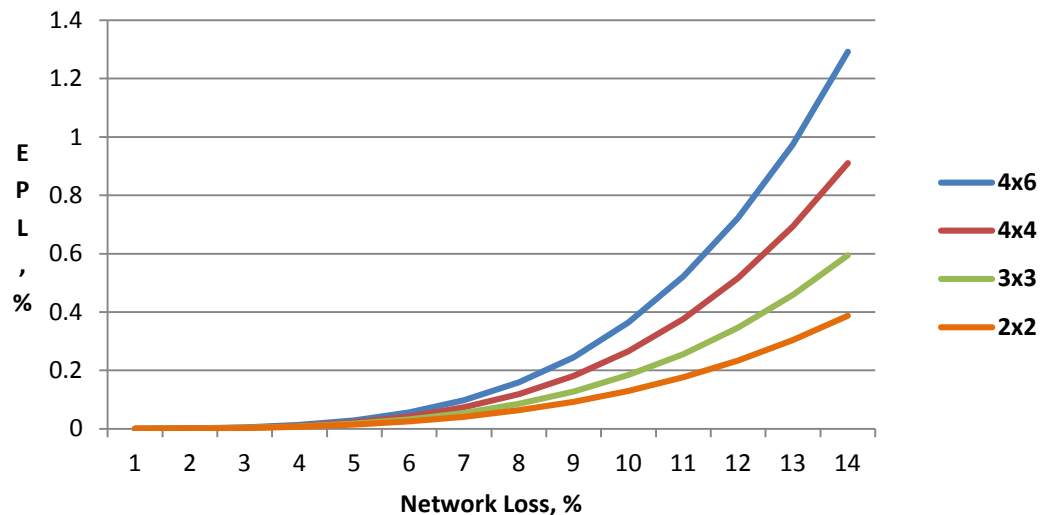


- FEC packets are generated from a matrix of RTP data packets
- Both RTP data and FEC packets are sent to the receiver
- FEC attempts recovery of lost data packets at the receiver
- Effectiveness of recovery depends on type of packet loss

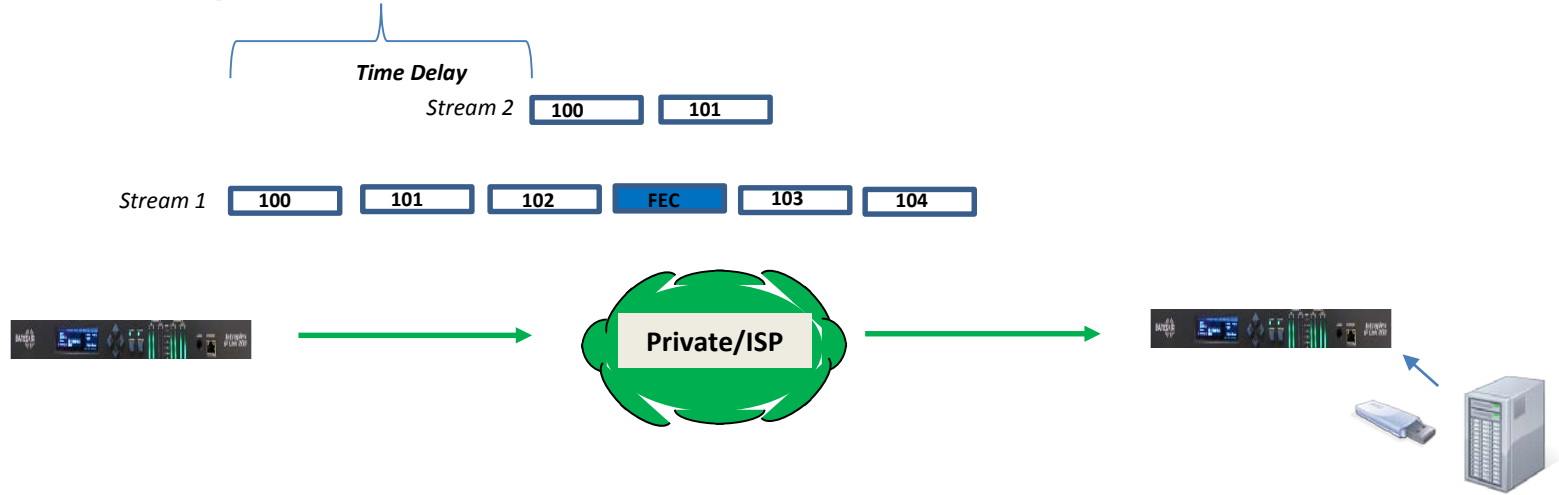
FEC Matrix

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

FEC Correction for Random Loss

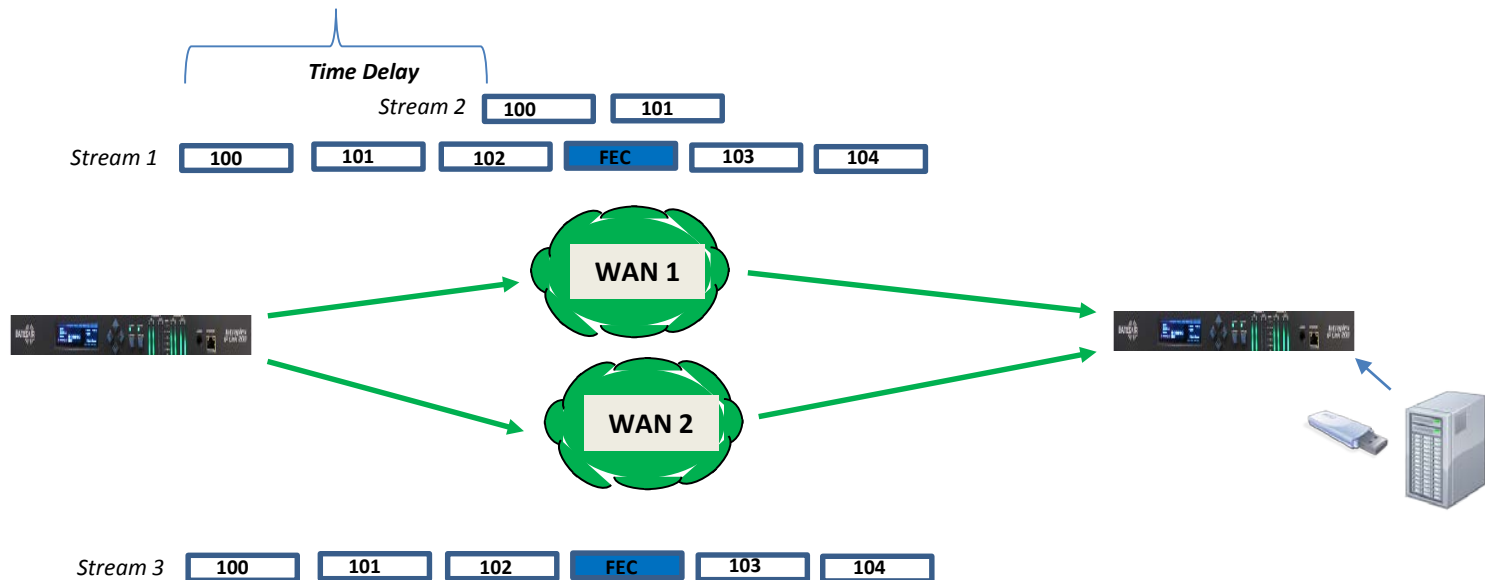


Single Network Packet Protection



- For burst loss, packet level FEC with interleaving– adds delay
- Or, add redundant streams in a group with programmable time delay. Very effective for burst packet losses which are typically seen on public ISP connections
- Time delay value based on network analytics

Multiple Network Packet Protection



- Network diversity
- Grouped streams sent across diverse network paths
- Scalable protection per network based on capacity
- “Hitless” operation with packet and network losses

Summary

- High bandwidth IP connections is an enabler for MPX transport
- MPX STL - advantage of centralization at studio and simulcasting
- Two methods of MPX – MPX over AES and analog MPX
- Analog MPX with DSP signal processing can enable high quality transport
- MPX STL timing – common timing or timing recovery
- For high quality MPX STL, IP packet loss must be mitigated
 - FEC
 - Redundant streaming
 - Network diversity

Thank You