# Analyzing Requirements for IP STL

April 17, 2016

GatesAir Connect | NAB Show 2016

Featuring GatesAir's

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GATESAIRCONNECT

Connecting What's Next

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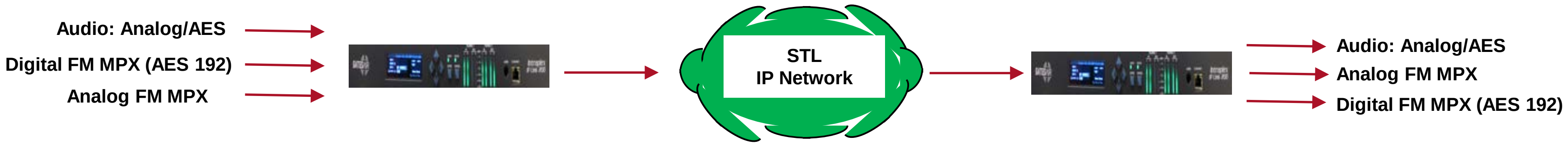
# Overview




- Analyzing capacity requirements on IP STL networks for transporting different signal types – Audio (uncompressed, compressed), MPX (Digital, Analog)
- Understanding the load generated by the encoder in terms of both data rate and packet rate
- Deciding between the use of unmanaged versus managed ISP connections based on the load generated by the encoder
- Overcoming network impairments: packet losses, jitter and delay. Relation between packet rate and techniques for reliability
- STL IP connection topologies and monitoring network performance using analytics tool

# Understanding Network Load

- Real-time media applications use RTP/UDP protocol encapsulation for transporting unicast and multicast streams
- Constant bit-rate with constant packet rate and packet size
- Total Data Rate for a stream = Packet Rate x Packet Size (bps)
- The Packet Rate has an impact on the load on networking devices. It is less taxing to process bigger packets with lower rates than smaller packets with higher rates due to lower packet interrupts
- Packet Rate is inversely proportional to Packet Time (a.k.a Packet Interval or Packetization) and delay. Packet Rate = 1/Packet Time
- Packet Time is the duration of audio segment contained in each packet
- A higher Packet Rate requires a higher Data Rate because of the packet overhead in each packet

# Packet Time Example

- Consider a Linear audio stream at 44.1 kHz sent with 2 different Packet Times below. Each packet has 40 Bytes of overhead (RTP+UDP+IP) and contains 16 bit samples for left and right channels

| Packet Time (msec) | Packet Rate (secs) | Packet size (Bytes) | Packet Overhead | Data Rate (mbps) |
|---|---|---|---|---|
| 4 msec | 250 | 745 | 5.1% | 1.5 |
| 8 msec | 125 | 1451 | 2.7% | 1.45 |

4 msec Packet Time generates an additional 125 Packets/Sec which need to be handled by all devices in the middle

# Selecting Packet Time For Linear Audio

- For Linear/Uncompressed audio, Intraplex® IP Link allows users to select Packet Time from 1 msec up to some value without causing the packet to be fragmented.
- With 1 msec packet time, end to end delay of less than 10 msec is achievable
- When selecting low packet time, network design must be considered carefully.
- If there is a flexibility in playout delay, our recommendation is to select the highest value without fragmenting the packet

| Sample Rate | Max Packet Time | Max Buffer Delay * |
|---|---|---|
| 48 kHz | 5 msec | 2.5 secs |
| 44.1 kHz | 8 msec | 4 secs |
| 32 kHz | 8 msec | 4 secs |

** Based on 512 packet buffers per stream*

# Compressed Audio

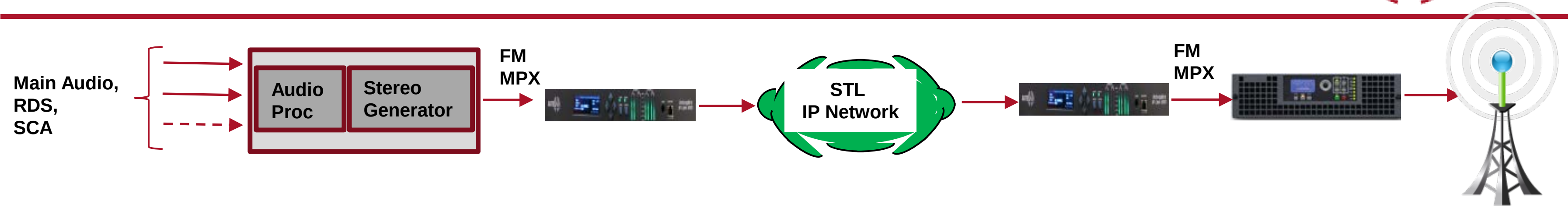
| Codec | Packet Time (msec) | Packet Rate (secs) | Data Rate (kbps) |
|---|---|---|---|
| Linear@48kHz | 5 | 200 | 1601 |
| Opus<br>96 - 160kbps | 20 | 50 | 120 - 180 |
| AAC LC<br>128 – 192kbps | 21 | 48 | 144 - 210 |
| AAC-He<br>80 – 128kbps | 42 | 24 | 96 - 134 |
| AAC-Hev2<br>40 - 64 kbps | 21 | 24 | 56 - 72 |

- Opus:
  - Open source, used for both voice and wideband audio
  - Backed by Google, Microsoft and Mozilla
- AAC
  - LC is the core codec
  - HE and HEv2 uses SBR and PS tools to reduce coding rate
  - HE/HEv2 are widely used for both streaming and broadcast applications (DAB+, DRM, XM Radio)
- Selection should be based on:
  - Source content
  - Network capacity and condition

# Compressed Audio

- Packet Rates for AAC codecs can be further reduced by packing multiple frames in a RTP payload. This increases the packet time of audio. As low as 5 packets per second is possible (200 msec Packet Time).
- Higher Packet Time also provides better tolerance of jitter by having a longer playout buffer delay. Intraplex IP Link can provide up to several minutes of buffering depending on the codec configuration
- With a higher packet time, there could be audible gaps because the Packet Loss Concealment techniques may not be as effective
- Compressed audio is best suited for unmanaged ISP connections:
  - Packet Rate and Coding Rate can be adjusted based on the network conditions
  - Lower packet rate has less impact on NAT and VPN devices.
  - Lower data rate allows more capacity to be used for packet loss protection.

# FM MPX Over IP




- Benefits of FM MPX 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 Tx sites
  - Simplifies the SynchroCast® operation for FM SFN. The baseband processing and stereo generation delay is common for all sites
- Similar to uncompressed audio, MPX also requires higher sustained STL capacity
- High network jitter can impact the performance of the adaptive stream clock recovery. No proven method to conceal lost packets
- Requires private or high speed ISP connection with guaranteed SLA

# FM MPX Over IP

- Digital FM MPX – AES 192
  - AES/EBU interface with 192 kHz sampling rate to carry up to SCA 1 subcarrier
  - Uses left channel of AES/EBU interface
  - End-to-End digital path provides better audio quality

Data Rate Requirements:

| Sample Rate/ Sample Size | Services | Packet Rate (secs) | Data Rate (mbps) |
|---|---|---|---|
| 192 kHz/24 bits | Audio + RDS+SCA1 | 500 | 4.7 |
| 192 kHz/16 bits | Audio+RDS+SCA1 | 333 | 3.2 |

# FM MPX Over IP

- Analog MPX
  - Flexible data rate configuration based on FM subcarriers to transport.
  - Reduced data rate and packet rate to carry Audio + RDS Vs AES 192. Sample Rate of 132 kHz Vs 192 kHz for AES 192
  - Data rate and packet rate for each sample rate varies based on sample size: 16, 20 and 24 bits

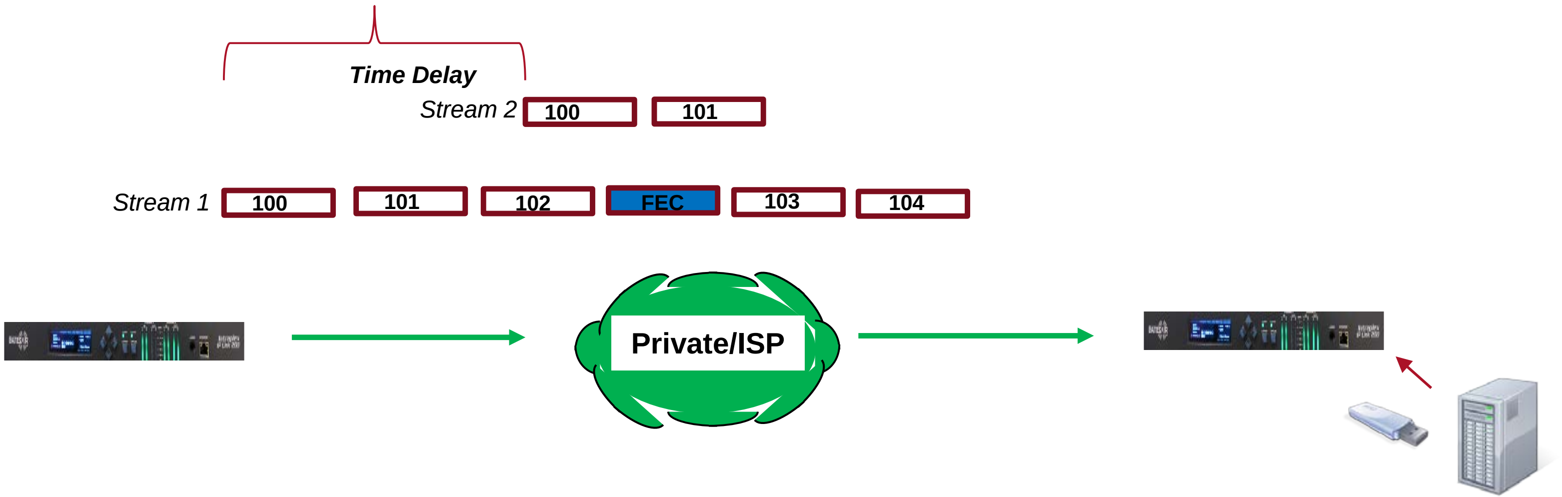
Data Rate Requirements:

| Sample Rate | Services | Packet Rate Range(secs) | Data Rate Range (mbps) |
|---|---|---|---|
| 132 kHz | Audio + RDS | 200 - 333 | 2.2 – 3.2 |
| 164 kHz | Audio + RDS + SCA1 | 333 - 500 | 2.7 – 4.2 |
| 216 kHz | Audio + RDS + SCA 1 + SCA 2 | 500 | 3.6 – 5.4 |

# Protecting Against Packet Losses

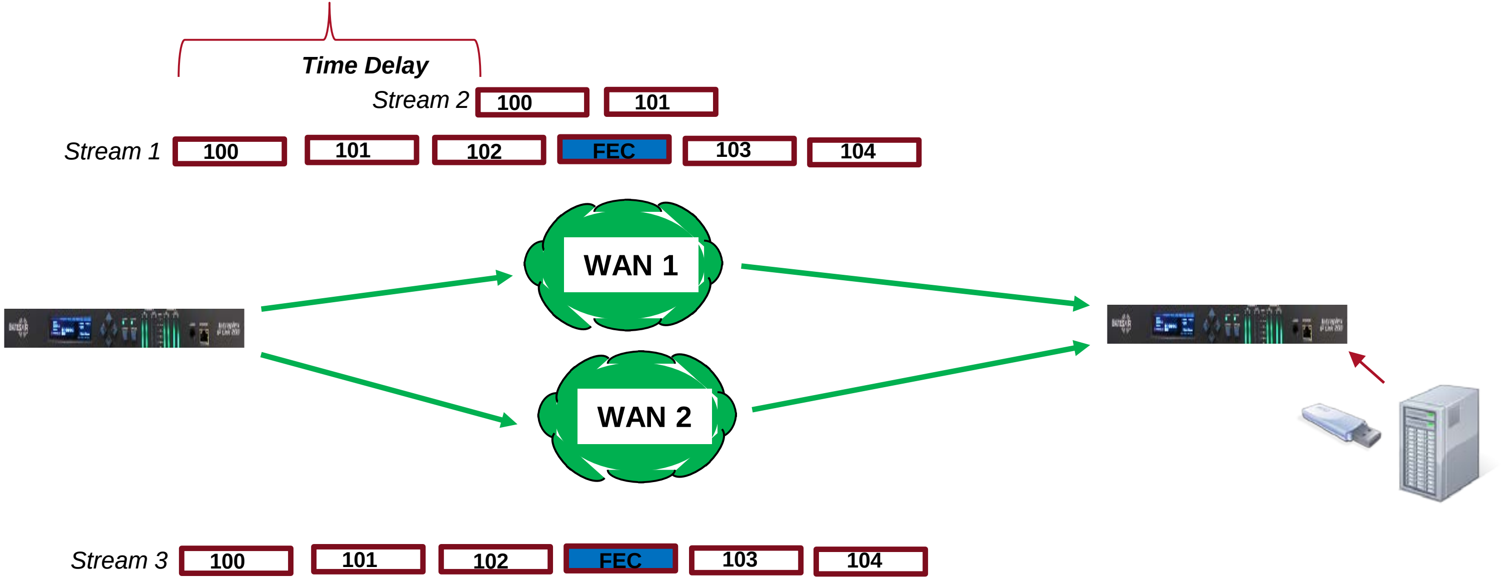
- There are 2 main methods to protecting against packet losses:
  - RTP Level FEC – uses parity packets to recover lost packet. Effectiveness is limited for burst packet losses
  - Stream Splicing technique of Intraplex® - Sends duplicate packet with time and network diversity. Effective for burst packet losses
- Packet Time implication on FEC effectiveness
  - Smaller Packet Times will require more columns of a 2-dimensional matrix.
  - For example, protection for a 100 msec network error will require 25 columns for a 4 msec packet time and 4 columns for a 25 msec packet time
  - The bigger the matrix size the greater the CPU resource requirements
  - Before utilizing FEC as a sole technique to recover lost packets, the users need to consider the matrix size, Packet Time and patterns of packet losses using network analytics
- Combining Stream Splicing and FEC can be provide effective protection against wide range of network conditions

# Single Network Packet Protection




- Packet level (FEC) Forward Error Correction: effective and efficient for isolated losses – typically seen on well managed connections
- 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

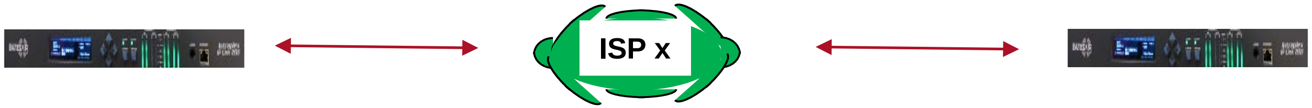
# Multiple Networks Packet Protection



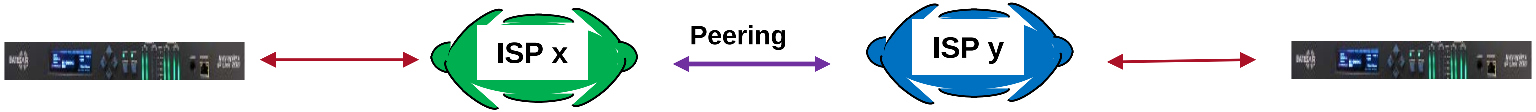

- Grouped streams sent across diverse network paths
- Scalable protection per network based on capacity
- “Hitless” operation with packet and network losses
- USB or local source as backup source

# IP Topology And Network Performance Monitoring

# ISP Topology – Intra Vs Inter ISP Connections




***Intra-ISP***

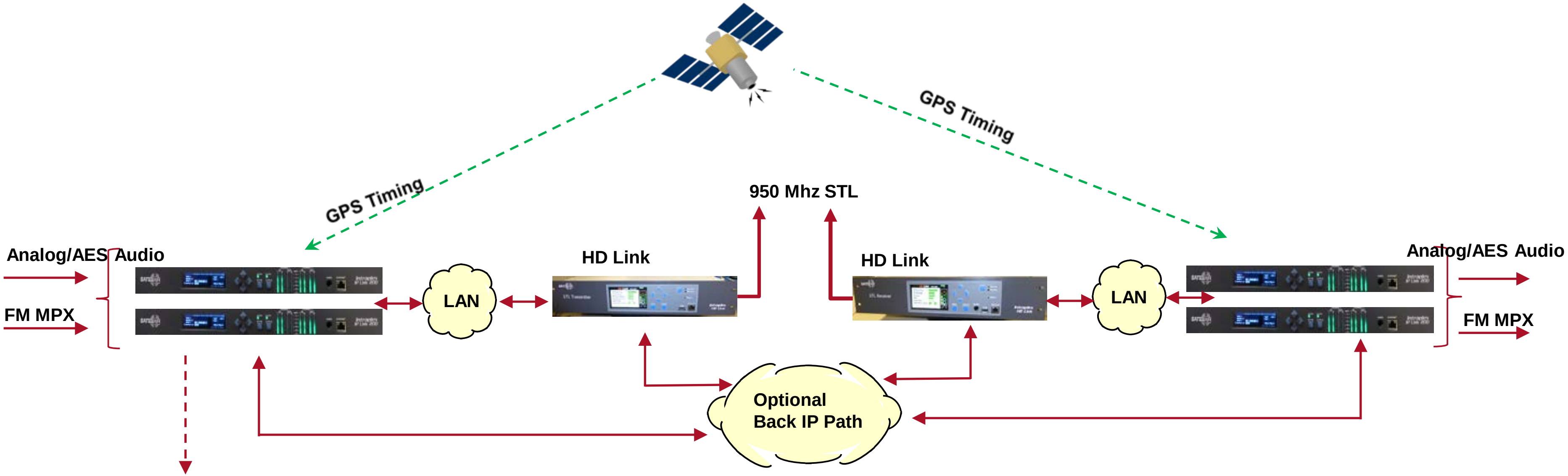


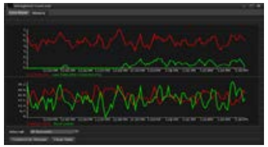

***Inter-ISP (Peering)***


***Inter-ISP (Transit)***

- For Inter-ISP connections, it is important to understand the traffic route. Use the Trace Route tool
- Peering and Transit connections points can cause congestion

# RF STL Based IP Connection



Intraplex LiveLook

Intraplex® HD Link provides up to 3.2 mbps of IP throughput

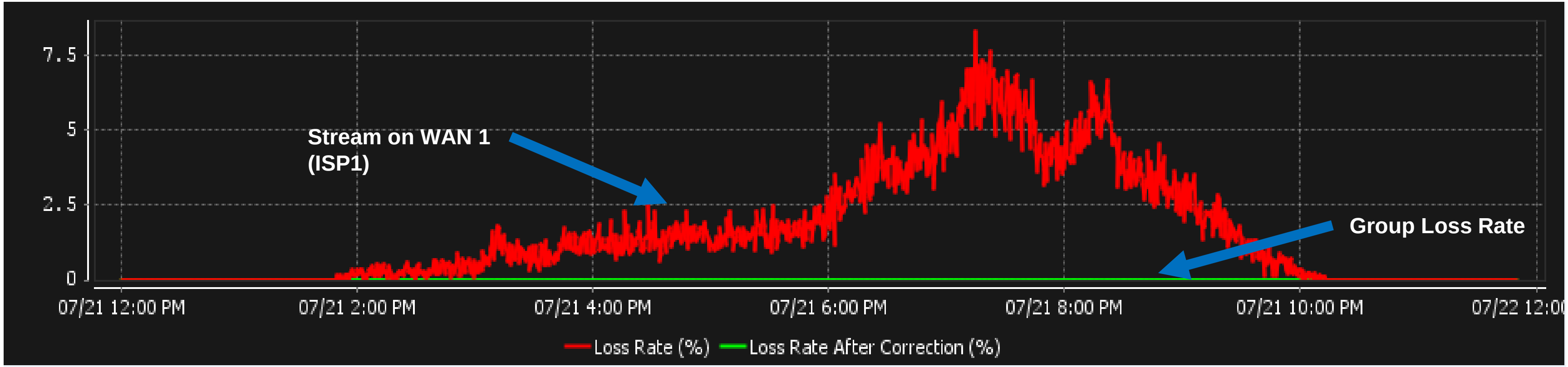
# Intraplex® LiveLook – Network Analytics Tool

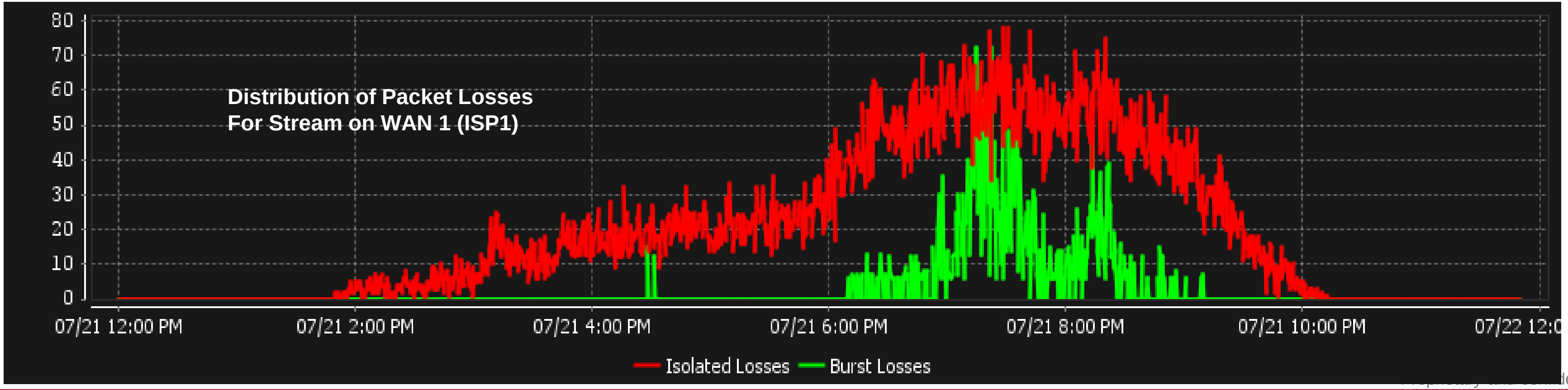

**Statistics and Status from IP Link**

**PC Application**


- Graphical network analytics and monitoring tool
- Recommends which packet loss technique will be most effective
- Logging of data helps with trouble shooting and SLA monitoring
- Dash board view of monitored audio streams with Email notification

# Use Case of LiveLook – KCLM Los Angeles



# Use Case: Inter-ISP Performance

# Thank You!