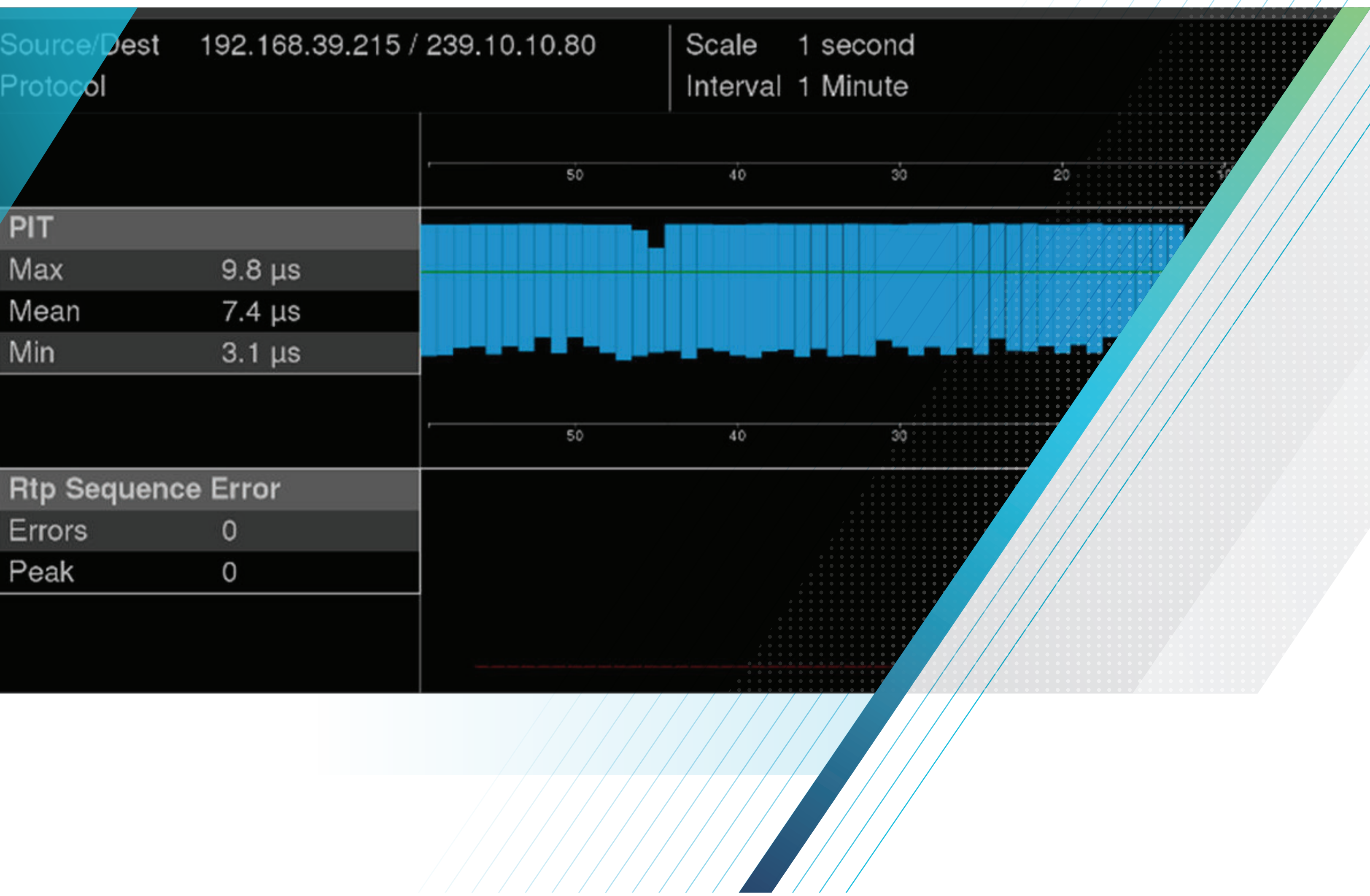
Telestream
Whitepaper

# Diagnosing and Resolving Faults in an Operational IP Video Network




telestream

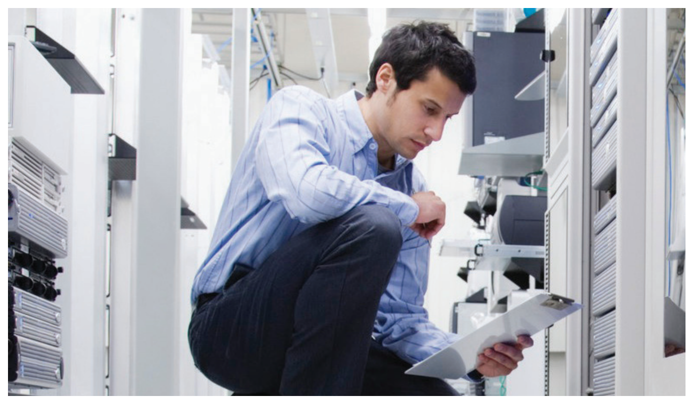
## Introduction

Deployment of IP Video networks in production and other operational applications exploits the ability to use Commercial Off-The-Shelf (COTS) IT-based infrastructure, which takes advantage of the economies of scale of the IT industry when compared with the relatively small broadcast industry. Additional advantages of reducing cabling cost and weight along with the much greater routing flexibility mean that in many parts of the World, trials, proofs of concept and deployments of IP Video are already in place. Having said this, IP does bring with it, technical challenges, including jitter; latency; the risk of dropped packets, an inherent lack of synchronicity along with asymmetry which results in different path delays upstream and downstream. Also, IP is a complex set of bi-directional protocols requiring a knowledge of both the source and destination before deployment.

Another issue is that deploying IP for video production applications is effectively the collision of the two Worlds of video engineering and network engineering. Video engineers are used to and comfortable with the use of SDI, coax, patch panels, black burst and tri-level for timing and above all, signal quality. The challenge for the video engineer is to understand IT technology and impact of an IT infrastructure on the video. On the other hand, network engineers are familiar and comfortable with, IP Flows, Protocols, Network traffic, Router Configuration and Precision Time Protocol (PTP) and Network Time Protocol (NTP) for timing. The biggest difference however is that in most data center applications, lost data can be re-sent – this is not the case with high bitrate video. The challenge for the network engineer is in understanding video technology and its impact on IT infrastructure. It is clear that there is a need for diagnostic monitoring and analysis tools that are usable by both video engineers and network engineers.

**Video Engineer**

- SDI, Analog, Audio and Patch Panels
- Black Burst and Tri Level Sync
- Importance of signal quality
- Challenge in understanding the IT technology and its impact on Video



**Network Engineer**

- IP Flows, Protocols, Network traffic, Router Configuration
- Precision Time Protocol
- Data can be resent - not the case with high bitrate Video
- Challenge in understanding video technology and its impact on IT infrastructure

## What Problems Can Occur in IP Networks?

A lot of the issues that can cause problems in IP networks can be traced back to packet jitter. As will be explained in the next section, excessive packet jitter can lead to buffer overflows and underflows causing dropped packets and stalled data flows. Other problems that can be experienced are associated with the timing delay and asymmetry of PTP packet flows. In hybrid SDI and IP workflows, it is also necessary to ensure that the relationship between the SDI and IP video is consistent to enable seamless frame accurate switching. This can be achieved by measuring the relationship between the Black Burst/Tri-Level Sync and the PTP clock and making any necessary correction by skewing the SDI syncs with reference to the PTP clock.

### WHAT CAUSES IP PACKET JITTER?

In any digital system, Jitter is any deviation from, or displacement of, the periodicity of the signal. In IP networks carrying constant bitrate data, jitter is the deviation from the periodicity of the packet arrival interval at a receiver. This can be caused by incorrect queueing or configuration issues, but assuming that the routers and switches are all configured and operating correctly, the most common cause of jitter is network congestion at router/switch interfaces.

A degree of jitter is inherent in any IP network due to its asynchronous nature. Obviously, the application within a network element will likely require the data to be received in a non-bursty form and as a result, receiving devices adopt a de-jitter buffer. The application then receives the packets from the output of this buffer rather than directly, with packets flowing out of the buffer at a regular rate, smoothing out the variations in the timing of the packets flowing into the buffer.

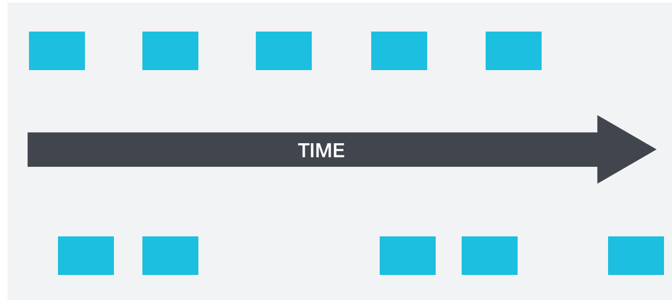



FIGURE 1. Packet jitter is deviation from the periodicity of the packet arrival interval.

### WHAT CAN BE THE IMPACT OF EXCESSIVE JITTER?

In the previous section, we saw that packets flow out of a receiver's buffer at a steady rate. This is known as the "drain rate" of the buffer. Conversely the rate at which a buffer receives data is known as the "fill rate". Selecting the size of the buffer is important as if the buffer size is too small then if the drain rate exceeds the fill rate, then it is possible that too small a buffer could underflow, resulting in stalled packet flow. If the sink rate exceeds the drain rate, then at some point the buffer will overflow, resulting in packet loss. However, if the buffer size is too large, then the network element will introduce excessive latency. As can been seen in the above diagram, network jitter causes the packets to become non-periodic and as such the buffer fill rate will no longer be constant. As the jitter becomes greater, the aperiodicity becomes larger. At some point this aperiodicity will lead to the condition where the buffer's fill and drain rates become so uneven that the buffer will either underflow, leading to stalling or overflow, leading to packet loss.

With the case of high bitrate video, either buffer underflow or buffer overflow will likely lead to impaired video. It should also be noted that port over subscription will of course also lead to packet loss.


FIGURE 2.

## IP Senders and Receivers

Initially ST2022-6 did not define specification for Senders or Receivers. Typically, 2022-6 IP gateway have to encapsulate the whole SDI data stream into IP packets so the rate of conversion was relatively constant, however server and software encoding devices could have wider distribution between packets. In the SMPTE standard 2110-21 Traffic Shaping and Delivery Timing for Video was defined. This document characterized the transmission of video RTP streams from a Sender and regulates the burstiness of the IP packets across the network. At the Receiver a virtual buffer model is defined to help regulate the flow of packets and define the buffer sizes. In ST2110-20 only the video image is packetized, and this can leave a gap between frames when no packets are sent. Alternatively, the device can decide to present the data more linearly over time. In an ideal case these packets would be presented at regular intervals, however the rate at which these IP packets are produce by the Sender can vary depending on how the device obtains the video frames and how it is able to convert these video images into IP packets that can then be sent out from the device.

A Narrow sender should be able to send out the packets at fairly regular intervals but will leave a gap at the end of the image until the start of the next image when no packets will be sent during this period. Typically, this would be done by a gateway card that maybe part of a camera or other video processing device. Alternatively, a Narrow Linear sender should be able to send out the packets at regular intervals throughout the time interval from the end of one field/frame to the next. This type of sender maybe a graphics device that provides a Logo or Ident.  A Wide Sender is a device that can take a longer time to process the image this can lead to a wider variation between when packets are sent and can result in a more bursty distribution of the packets. This type of sender maybe a PC based server or software based implementation.

### SENDER MODEL

To ensure a device passes the Network Compatibility Model a leaky bucket of infinite capacity is defined, and packets drain from this bucket at a number of packets per second. The number of packets in the bucket at a time is defined as CINST and there is a defined maximum number of packets CMAX based on the type of Sender and the format of the signal.

### TYPES OF SENDERS

- Narrow (N)
  - $C_{MAX} = 4$
  - $VRX_{FULL} = 8$
- Narrow Linear (NL)
  - $C_{MAX} = 4$
  - $VRX_{FULL} = 8$
- Wide Sender (W)
  - $C_{MAX} = 16$
  - $VRX_{FULL} = 720$

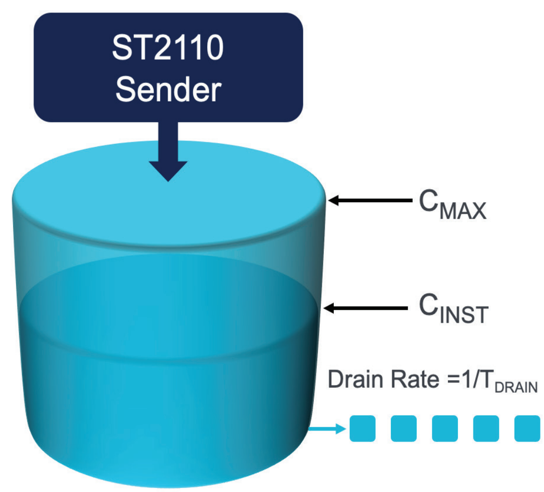



FIGURE 3. ST2110 Sender Model.

## RECEIVER MODEL

Variations across the network mean that each packet may take a different path from sender to receiver or congestion within a switch may delay a packet from being sent out of the port. These delays in packet delivery to the sender can cause out of order packets or wider time latency between packets or burstiness to packet delivery. The Virtual Receiver Buffer model defined in ST2110-21 provides characterization that the sender passes the model at the output. In this case a leaky bucket of capacity VRXFULL will allow packets to enter and exit the bucket instantaneously. Packets from the bucket will drain such that the jth packet will be removed from the Virtual Receiver Buffer a time TPRj

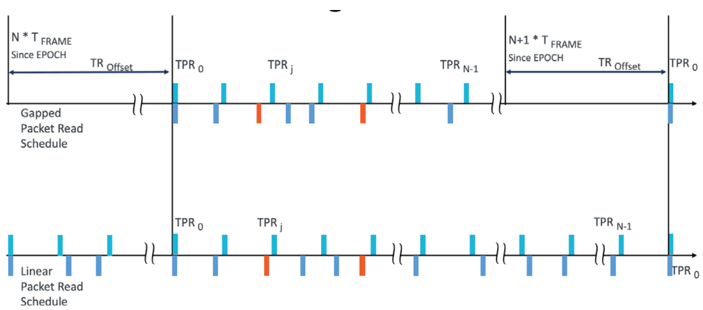



FIGURE 4. ST2110-21: Scheduled Read Timing & Video Entrance Timing

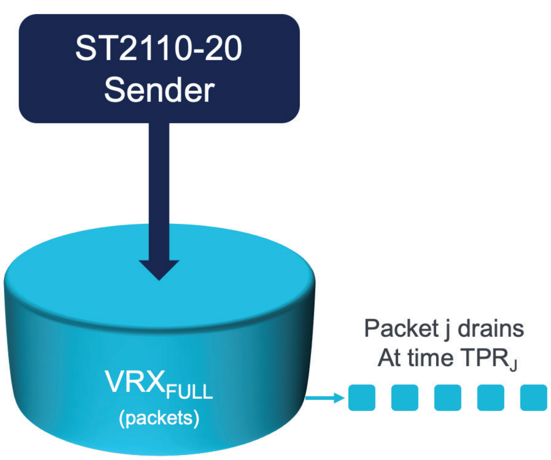



FIGURE 5. Virtual Receiver Model

## MEASURING JITTER IN REAL TIME PROTOCOL (RTP) NETWORKS

We have already seen that in networks carrying constant bitrate data, jitter is the deviation from periodicity at a receiver and as such, given an accurate clock in the receiver, jitter can be measured simply by measuring the timestamps of the packet arrival times and plotting the inter-arrival intervals versus time.

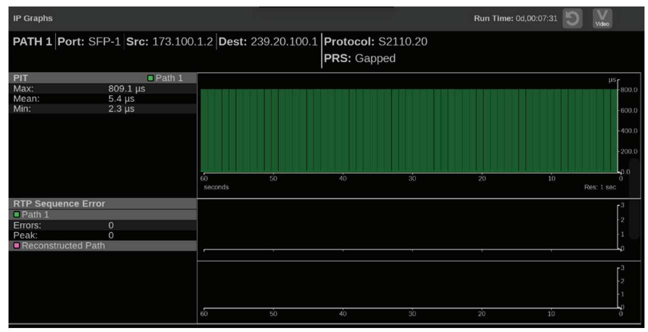



FIGURE 6. Packet inter-arrival intervals plotted versus time.

This method is useful to identify variances in jitter over time, but it is also useful to be able to plot the distribution of packet inter-arrival intervals versus frequency of occurrence as a histogram. We have also already seen that If the jitter value is so large that it causes packets to be received out of the range of the de- jitter buffer, then the out-of-range packets are dropped. Being able to identify outliers is an aid in identifying if the network jitter performance is either likely to or already the cause of packet loss.

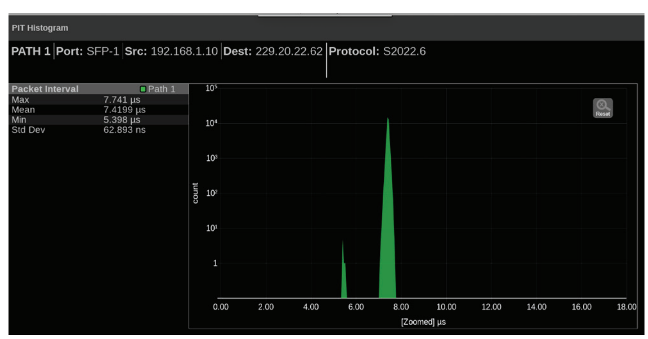



FIGURE 7. Packet inter-arrival intervals plotted versus frequency of occurrence.

## Diagnosing Issues using PIT Graph & PIT Histogram displays

The variations in packet interval arrival time and the distribution of the packets can produce a characteristic signature of a device and show the deviation of the packets due to network congestion or issues. Let's start by using the Telestream Prism connected directly to the Sender output without introducing variation across a network to show the how the packets are sent out of the device.

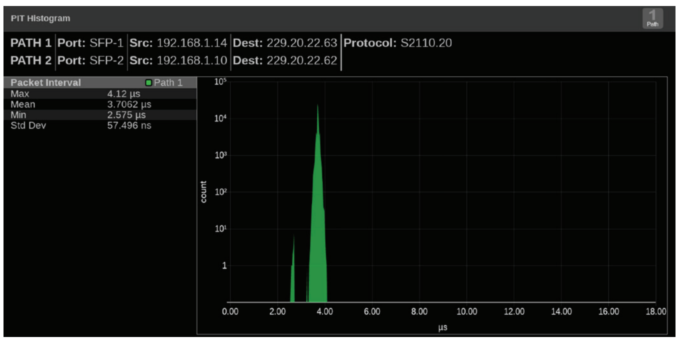



FIGURE 8. Histogram of ST2022-6 Sender.

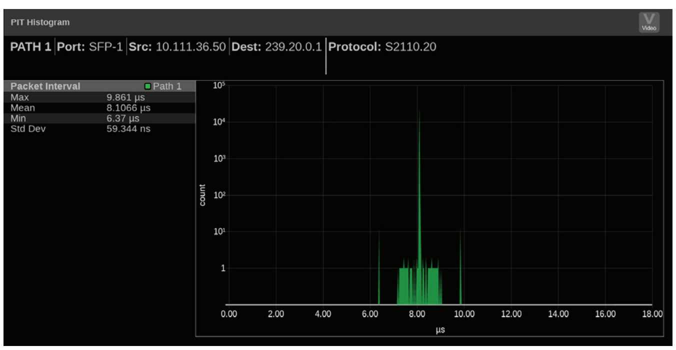



FIGURE 9. Histogram of ST2110-20 Narrow Linear Sender.

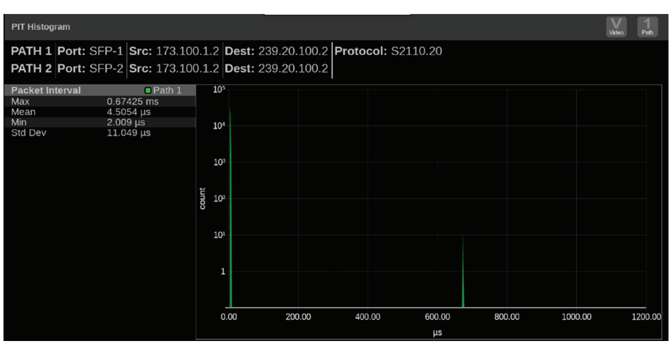



FIGURE 10. Histogram of ST2110-20 Narrow Gapped Sender.

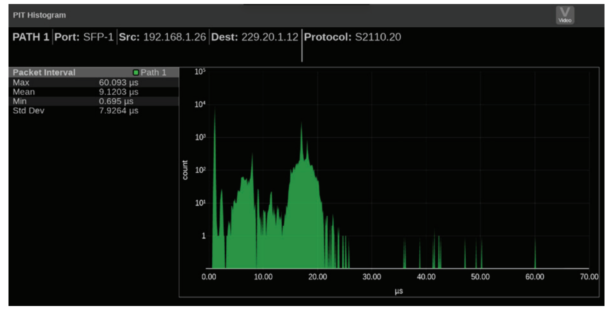
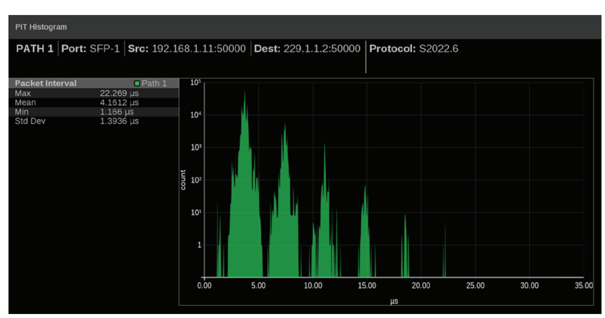



FIGURE 11. Histogram of ST2110-20 Wide Sender.

Figure 8 shows the Histogram of a ST2022-6 gateway here you can see a narrow pulse with another pulse that has slightly deviating from the main pulse. In the case of ST2110-20 Narrow Linear Sender there is also a single main pulse as the packet are sent out with similar linear spacing between packets. In Figure 10 a Narrow Gapped ST2110-20 send shows two pulse that are a distance apart. This distance is equivalent to the blank interval of the video format and is due to that fact that no packets are sent between the end of the last frame and the beginning of the next frame, so this is characteristic of a Gapped Linear sender. Note for a interlace format there will be multiple pulse showing the difference in timing between field 1 and field 2. For a Wider Sender there is a wider spread to the distribution of packets that are likely dependent on the internal process of the device for instance a server may have to gather the data from a disk array and will be dependent on the speed of access to the drive and ability to get the information across the busses of the system in order to process the data to the output. Typically, a device will have a characteristic PIT Histogram signature when directly connected to Prism. The PIT Histogram may change when the stream is passed through a series of switches and be dependent on other network traffic. Therefore, the PIT Histogram can be used to help diagnose issues with a stream. In this case in Figure 12 a loop was present in the network and this cause the same traffic to be passed across the network again and again giving a characteristic PIT Histogram display.



## Establishing de-jitter buffer size for ST2022-6

We have already seen that merely measuring the packet inter-arrival times cannot realistically be used to predict the necessary de-jitter buffer size. There is however an alternative form of jitter measurement know as Delay Factor (DF) that can be used to establish de-jitter buffer sizes. Delay Factor is a temporal measurement, which in the case of high bitrate video is represented in microseconds, that indicates how much time is required to drain a virtual buffer at a network node. At any given time, the Delay Factor (DF) represents the temporal buffer size at that network node necessary to de-jitter the traffic flow.

One such form of DF measurement takes advantage of the fact that RTP carries time stamp information which is defined by RFC 3550 as reflecting the sampling instant of the first octet in the RTP data packet (the timestamp format being the same as that of NTP). This measurement is known as Time-Stamped Delay Factor or TS-DF, as defined by EBU Tech 3337. This method is in the public domain and is well suited to high bitrate media over RTP applications. TS-DF is based on correlating arrival times of network packets with the time-stamp field in the RTP header.

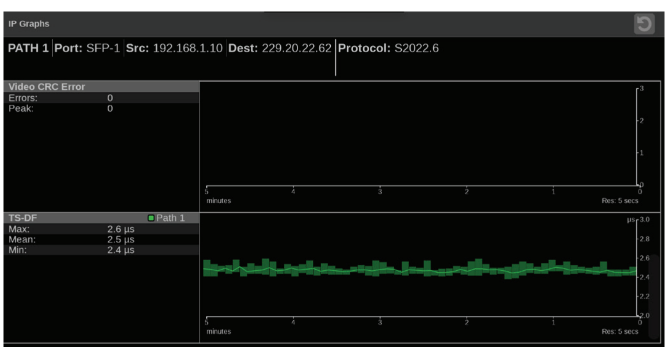



**FIGURE 13. TS-DF represents temporal buffer size in microseconds.**

The Time-stamped Delay Factor measurement is based on the Relative Transit Time defined in RFC 3550 (RTP: A Transport Protocol for Real-Time Applications). This is defined as the difference between a packet's RTP timestamp (held in the RTP header) and the receiver's clock at the time of arrival, measured in the same units. The TS-DF measurement period is 1 second. In this algorithm, the first packet at the start of the measurement period is considered to have no jitter and is used as a reference packet.

For each subsequent packet which arrives within the measurement period, the Relative Transit Time between this packet and the reference packet is calculated and at the end of the measurement period, the maximum and minimum values are extracted, and the Time-stamped Delay Factor is calculated as:

$$\text{TS-DF} = D(\text{Max}) - D(\text{Min})$$

Unlike the jitter algorithm in RFC 3550, this algorithm does not use a smoothing factor and therefore gives a very accurate instantaneous result. The hybrid SDI and IP media analysis platform implements both IP packet inter-arrival jitter measurements as well as the RTP specific TS-DF measurement.

## Network Compatibility Model for ST2110-20 using CMAX and VRXFULL displays.

Before monitor a ST2110-20 video stream to check its conformance to the Sender and Receiver buffer models, it is important to check that the Sender is locked to the Precision Time Protocol (PTP) reference. This can be done in Prism using the Timing display shown in Figure 14. Prism should be correctly configured for the PTP domain and show a green tick indicator within the status display that the device is locked to PTP. Within the Timing display Prism will show an offset from zero based on the blanking interval of the video format. For HD interlace format this will be 21 lines, for progressive formats the measurement will show 42 lines. There will be a slight wander in the Timing display around this blanking interval, but the value should remain around the blanking interval to show that the Sender is locked to PTP. Large variation in the timing display wandering through line by line is an indication that the Sender is not locked to PTP and therefore results from the CMAX and VRXFULL displays will not be valid.

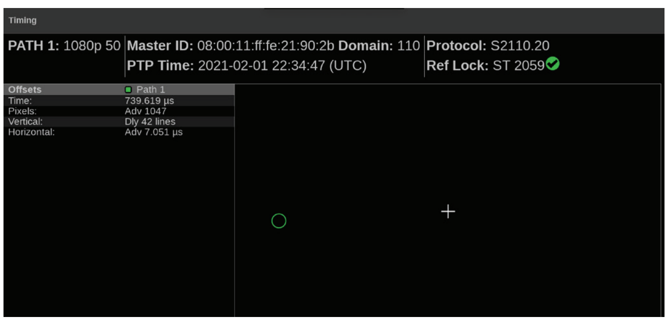



FIGURE 14. Prism Timing Display show PTP lock to a 3G 1080p50 signal.

For CMAX and VRXFULL to be measured correctly it is important to confirm the type of Sender (Gapped, Narrow Linear or Wide) this can be done using the Histogram display and then entering the appropriate type within the input configuration for Packet Read Schedule (PRS). This will ensure that the appropriate limits are applied as defined in ST2110-21 for the type of sender.


FIGURE 15. Prism CMAX and VRXFULL measurements for ST2110-20.

STREAM TIMING

To further analyze stream timing across the network Prism provides a series of Stream Timing displays. The graph of Vid-PTP Offset shows a plot of the variation of the Timing display measurement versus time (Figure 16). The graphical plot can be shown in time intervals from one minute to one day to help characterize the Sender performance. The second graph shows VID-RTP offset, in this case Prism derives the current RTP value based on the PTP time and compares against the RTP time obtained from the sender's packet. This can help characterize the delay introduce across the network that could depend on the loading of the network over time.


FIGURE 16. Prism Video Stream Timing Display

Additionally, the Stream Timing display can also be used to monitor the Aud-Vid Offset and Aud-RTP Offset to allow the user to compare the video and audio timing. Within ST2110 it should be possible to combine video and audio from multiple sources and this requires that each Sender be correctly locked to PTP and produce synchronous outputs. Variation in timing between the video and audio streams can be shown in the Audio tab of the Stream Timing display shown in Figure 17.


FIGURE 17. Prism Audio Stream Timing Display

## What is Precision Time Protocol (PTP)?

IP based networks can be considered to be asynchronous in that device clocks, at nodes distributed across the network have no inherent concept of system time. Precision Time Protocol (PTP: defined by IEEE 1588) is intended to synchronize the real- time clocks of different nodes on an Ethernet network. It should however be noted that PTP does not make the network itself synchronous (as is the case with Synchronous Ethernet also referred to as SyncE). The most recent version is IEEE 1588- 2019. However most equipment is currently operating under the IEEE 1588 -2008 also known as PTP version 2 and SMPTE has developed a standard based on PTP version 2 specifically intended for broadcast video applications, known as SMPTE ST 2059.

The adoption of video over IP along with the use of PTP to synchronize the real-time clocks of different network nodes infers that any such network requires a network time server, in order to provide the PTP genlock functionality equivalent to that delivered by a Sync Pulse Generator (SPG) in SDI networks.

Any logical grouping of clocks that are synchronized together are referred to as a PTP domain. Note that a clock in one domain may not be synchronized to clocks in another domain.

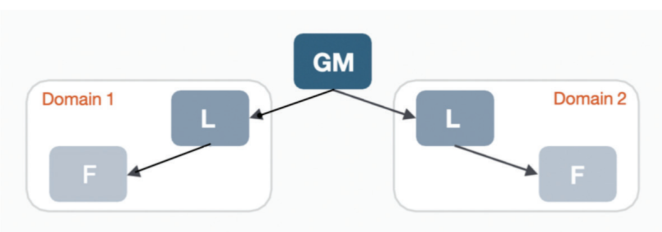



FIGURE 18. PTP Domains Synchronized to a Common Grandmaster

This PTP network time server is generally referred to as a PTP Grandmaster, with a device that derives its timing synchronization from PTP being referred to as a PTP Follower. A Leader clock is a device that provides the time in a given PTP domain and a Follower is a device that synchronizes to a Leader. A Grandmaster is a Leader that is providing the ultimate source of clock synchronization in a network. In the context of broadcast applications, PTP Grandmaster are usually synchronized to GPS, GLONASS or both.

### HOW IS TIME DERIVED IN A PTP NETWORK?

A network of Follower devices connected to a single Leader is known as a domain and within any PTP domain there are a number of message types used to establish time within that network.

Announce messages are used to establish the synchronization hierarchy and provide the clock status and clock criteria used to determine which clock becomes the Grandmaster. Sync and Follow-up messages are transmitted by the Grandmaster and are used by Followers to derive the time. Delay Request messages are a request for timing information and are sent from the Follower to the Grandmaster in order to determine the reverse path propagation delay between the Follower and the Grandmaster. A Delay Response message is sent by the Grandmaster and contains the time of receipt of the Delay Request message by the Grandmaster.

As defined, PTP is a method for distributing time over a network, with a single Grandmaster providing the source of time, to synchronize one or more Followers. The Grandmaster periodically transmits Sync and Follow up messages, which the Followers use to derive the time. In an ideal World the network delay could be programmed into each Follower which could then be offset to the time in the received packet to derive the correct time. Such symmetry can only be relied upon in point-to-point IP links. Unfortunately, the path delay in switched / routed IP networks is variable at different network nodes and can also be asymmetric, in order to take account of this, the Follower devices must periodically send Delay Request messages to the Grandmaster. The Grandmaster accurately time stamps these messages on receipt and the time of receipt is sent back to the Follower in a Delay Response message.

Using the diagram figure 19 as a reference, the Follower is now able to calculate the difference between its own clock and that of the Grandmaster using the Leader-to-Follower sync packet delay (T2-T1) and Follower-to-Leader delay request packet-delay (T4-T3). The Offset (Follower Time – Leader Time) = [(T2-T1)-(T4-T3)]/2. For the Follower time to be absolutely correct, the propagation delay in both directions must be equal.

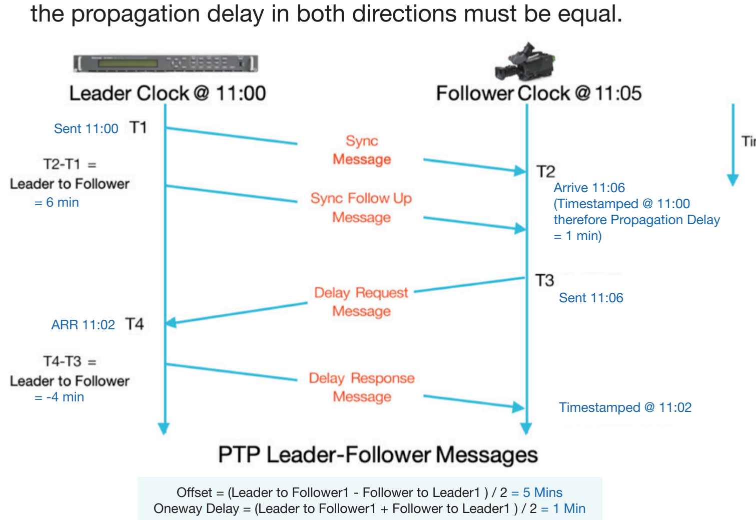



FIGURE 19. Deriving the Correct Time in a PTP Network.

## PTP MEASUREMENT AND MONITORING

The first thing to consider when monitoring the PTP network is to check the domain values present across the network. Within PRISM's IP Status the user can view the PTP traffic present and the domains that are present in the network. In this case we have a domain 110 and should configure the PTP Settings to reflect this within the instrument to ensure that device is locked to PTP.

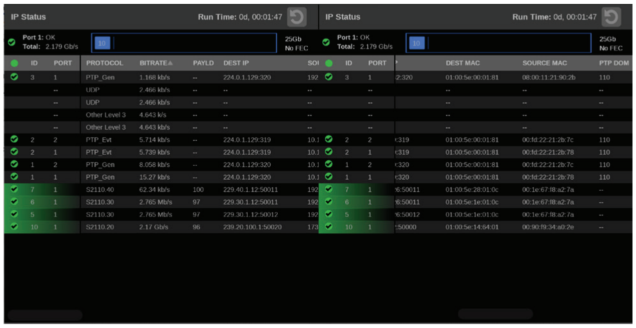


FIGURE 20. PRISM IP Status showing IP streams on the ports and PTP traffic with domains present.

The IP Session PTP tab show various syntax information including Grandmaster ID that is the MAC address of the current Grandmaster clock within the network. Additionally, the number of messages for Announce, Sync Delay Requests and Delay Responses are shown, and this should be verified to ensure that it meets system guidelines for PTP within the network. Additionally, the Priority levels, Clock Class, Clock Accuracy and Clock Variance can be verified for the current Grandmaster to ensure the appropriate clock is chosen to provide PTP to the network.

FIGURE 21. Monitoring PTP using PRISM's IP Session and PTP Graphs displays.

When considering accurate PTP performance, it is important to understand whether the network is excessively asymmetric. That is, that the packet delay from the Leader to the Follower device (L-F) is substantially different from the packet delay from the Follower to the Leader (F-L). If the propagation delay in both directions is different, then the Follower is offset to "correct" for this by adjusting its clock to a value of half the asymmetry.

The clock's control loop adjusts the Follower time to make the Leader-to-Follower and Follower-to-Leader propagation delays appear to be equal. If the asymmetry is excessive then the absolute clock value will not compensate accurately for either L-F or F-L packet delays.

FIGURE 21. Monitoring PTP performance with PTP Graphs displays.

## HYBRID BLACK BURST/TRI-LEVEL AND PTP NETWORKS

For the foreseeable future, many video networks will use a combination of SDI and IP. In these cases, it is vital that the timing of the BB/Tri-Level is synchronous with the PTP if frame accurate switching is to be achieved. This can be achieved by using the PRISM Timing display and comparing the Analog to PTP Timing against the Video Reference timing as shown in Figure 22. Once the measurements results have been obtained the timing adjusting can be made between BB/Tri-Level to PTP within the hybrid SPG/PTP Grandmaster such as the SPG8000A.

FIGURE 22. Telestream SPG8000A Sync Pulse Generator and Master Clock Reference.

FIGURE 23. Telestream SPG8000A Sync Pulse Generator and Master Clock Reference.

## Troubleshooting issues within an IP Video Network

Diagnosing and Resolving faults within an IP Video network the user needs to establish the root cause of the impairment. Whether the fault is due to IP errors or due to impairments within the video, audio or data. To be able to monitor multiple program a monitoring by exception approach is required. Focusing on each program to make sure it is present and healthy using the Telestream Inspect 2110. Notification can be triggered when the format of the video, audio or data has errors. Allowing an operator or engineer to investigate further by using the “click to view in PRISM”.

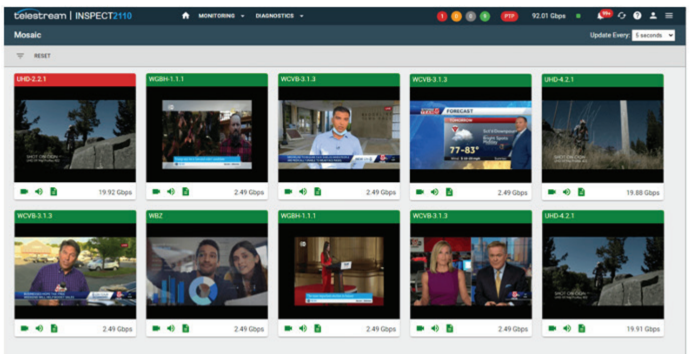


FIGURE 24. Telestream Inspect 2110 Mosaic view

Within the Telestream Prism a deeper analysis is provided, and the user can decode the IP stream of an ST2022-6 or ST2110-20 video, ST2110-30 audio, ST2110-40 data signal. The user can then check for impairments in the video or audio. The IP analysis tools within Prism can then be used to help diagnose the issue.

FIGURE 25. Prism decoding an IP stream showing waveform, picture, audio bars and event log.

Viewing the IP Status provide indication of the stream present on the IP port of Prism, simple red or green indicators provide indications of the health of the signal with metrics for Bitrate, IP address, port, Payload Type and PTP Domain. The RTP Clock should be 48kHz for audio and 90kHz for video with the RTP Marker frequency corresponding to the frame rate of the video. Further analysis of the syntax is then possible within the IP Session.

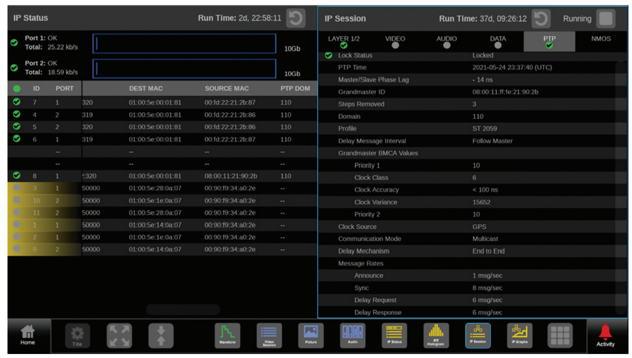


FIGURE 26. IP Status and Session of Prism showing IP stream information.

Being able to characterize when the errors have occurred is important in troubleshooting the issue and the IP graphs can show metrics of Bitrate, Packet Interval Arrival Time, RTP Sequence errors correlated to time from 60 seconds to 24 hours showing trending of the stream and allowing users to pinpoint errors. Users can also characterize the stream using the PIT histogram as discussed previous and contrast the sender histogram with that of the stream being passed through the network. Additionally, PRISM can perform a stream capture to aid in analyzing problems with the streams to diagnose the issue.

FIGURE 27. IP Graphs and PIT Histogram of Prism showing IP stream information.

## Remote Monitoring

In live production applications, network experts may not be present on the production site and networking equipment also may not necessarily be in a location that is easily accessible. With Inspect 2110 user can access the interface via a web browser application giving them the ability to monitor their system remotely. While PRISM uses noVNC application to provide remote connection to the device allowing the user to access the user interface in the same manner as they would from the front panel of the instrument. Allowing the network engineer and video engineer or operator to collaborate on resolving issues within their network.

Figure 28. PRISM Remote network monitoring.

## Summary

IP offers both opportunities and challenges for broadcasters in an environment where the worlds of Video Engineering and Network Engineering collide, it is essential that diagnostic monitoring and analysis tools are usable by both Video Engineers and Network Engineers.

Understanding the technologies of both video and IP will aid in the ability to diagnose and resolve faults within the system. The ability to monitor jitter performance over time to avoid buffer overflows and consequently packet loss will ensure an error free transmission across the network. With an ability to quickly identify network issues when they occur to determine the impact in terms of picture impairments and ensure synchronization of video, audio and data.

PTP provides the critical component in ensuring synchronization required in live production workflows and the ability to monitoring timing stability across the network is key to ensuring a correctly timed system.

Finally, the transition to IP will be a gradual “Hybrid” SDI/IP workflows and hence the need for a hybrid diagnostic monitoring and analysis toolset as well as the need for a hybrid Sync Pulse Generator/ Grandmaster will be needed within your facility.

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