An Introduction to IP Video and Precision Time Protocol

WHITE PAPER

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It’s hard to attend a broadcast industry trade show or read industry news without seeing much discussion about the enormous technological changes in the works that will impact the broadcast industry over the next few years. Some changes such as 4K/UHDTV, High Dynamic Range and High Frame Rate video could be regarded as evolutionary, but the transition to an all IP video workflow is regarded by many as a revolutionary and disruptive technology change that will demand entirely new skillsets and infrastructure.

FIGURE 1.

The migration to IP will impact everyone across the broadcast chain to varying degrees, including content producers, broadcasters, content providers, content distributors and equipment manufacturers. However, possibly the biggest impact will be to live production workflows.

Although many see IP Video as new technology, in fact in video distribution workflows, the transition from ASI to IP began almost 15 years ago and IT technology began to enable the transition to file-based workflows more than 10 years ago. In fact production is the last remaining stronghold for SDI and that is set to change. It is reasonable to ask why SDI remains in use in live production workflows. The answer is that the technology works very well indeed, giving outstanding image quality, with extremely low levels of jitter and latency as well as offering an extremely “thin” unidirectional protocol that is extremely easy to deploy and which makes frame accurate switching inherently simple. In addition SDI is an open, non-proprietary and universally supported standard.

So why would we want to move to using IP? The most commonly quoted reasoning is the ability to use Commercially 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. In addition it offers advantages of reducing cabling cost and weight. All this certainly true, but probably the biggest advantage is the much greater routing flexibility offered along with enabling new workflows such as downstream/centralized production. These new workflows in turn are likely to lead to new types of content to provide to viewers and with it new sources of revenue. One aspect of using IP for transporting video that is often overlooked is that scalability is no longer a function of port density, but instead is merely a function of bandwidth.

Having said this, IP does bring with it some 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. However all the above are surmountable, but it does not change the fact that IP is a complex set of bi-directional protocols requiring a knowledge of both the source and destination before deployment. It is often thought that transporting uncompressed or lightly compressed video is the most difficult application for IP, but in fact it could be easily argued that trading floors, where time is money demand even greater levels of performance. Switches intended for trading room applications typically offer latencies less than 250 ns, which offers more than enough performance for IP video applications, where timing accuracy is typically around 1 μs.

The Application of Standards

In general, when we refer to video over IP in the context of any video production workflow, we are referring to the distribution of either baseband or lightly compressed video over Real Time Protocol, commonly referred to as RTP. The advantage of using RTP as opposed to Universal Datagram Protocol (UDP) for the transport layer is twofold. RTP packets are time-stamped making the measurement of packet delay variation easier, but critically the packets also carry a sequence number, making the detection of dropped or out-of-order packets relatively straight forward.

In addition to carrying Video over IP, in a live production environment it is critical to consider synchronization and timing. The asynchronous nature of IP has the advantage that many different traffic types can be carried across a network without having to be concerned with synchronization, but this presents a challenge in the production environment where synchronization is critical to enable frame-accurate switching as well as synchronous video processing. To provide the necessary “genlock”, there remains the need for a precise timing standard, which for both IP and Ethernet networks is provided in the form of the IEEE 1588-2008 Precision Time Protocol, commonly referred to as PTP version 2. This is also the basis of a recently introduced SMPTE PTP standard, specifically intended for the timing and synchronization of video transmitted over RTP networks – the two part SMPTE ST 2059-1 and 2059-2. Likewise, there is an AES67-2015 PTP profile for use with audio transmitted over RTP using the AES67 format. The first part of the SMPTE ST 2059 standard refers to “the generation and alignment of interface signals to the SMPTE Epoch” (Date 1970-01-01 Time 00:00:00 TAI) and the second part refers to the definition of a “SMPTE profile for use of IEEE 1588 Precision Time Protocol in professional broadcast applications”. SMPTE ST 2059-2 is designed to enable any slave introduced into a network to become synchronized within 5 seconds and to maintain network-based time accuracy between slaves to within 1 micro second. It should be noted that while PTP provides a mechanism to synchronize the real-time clocks of devices on an Ethernet-based network to the same time, it does not make the network itself synchronous (as is the case with Synchronous Ethernet also referred to as SyncE).

Coming back to the carriage of Video over IP, there are a number of specific industry standards and proprietary methods for its distribution. SMPTE ST 2022-6 is a standard designed to transport uncompressed SDI video, embedded audio and metadata over RTP/UDP. Although it is possible to send audio on a separate flow, for example using AES67, it should be noted that the payload is always an entire SDI datagram carried at constant bitrate. When audio is distributed as a separate flow, the bitrate of this flow must be provisioned, in addition to the bitrate required by SMPTE ST 259M, 292M or 424M.

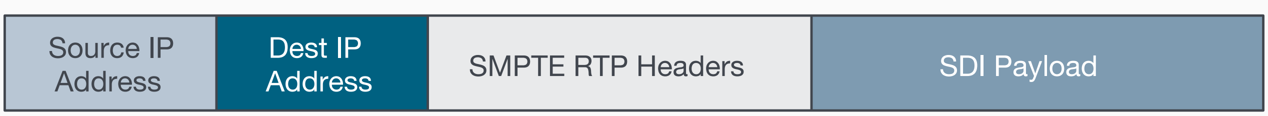


FIGURE 2. SMPTE ST 2022-6 IP Packet Format (A Single IP Flow Can Carry Video, Audio and Metadata over SDI).

SMPTE ST 2022-5 Row/Column FEC

P_1	P_2	P_3	P_4	P_5	F_{R1}
P_6	P_7	P_8	P_9	P_{10}	F_{R2}
P_{11}	P_{12}	P_{13}	P_{14}	P_{15}	F_{R3}
P_{16}	P_{17}	P_{18}	P_{19}	P_{20}	F_{R4}
F_{C1}	F_{C2}	F_{C3}	F_{C4}	F_{C5}	

P_1	Video Data Packets
F_{R1}	Row FEC Packets
F_{C1}	Column FEC Packets

FIGURE 3.

Within the SMPTE ST 2022 family of standards, there is also provision for a method for Forward Error Correction (FEC), as defined by SMPTE ST 2022-5 and also a method for seamless protection switching of two SMPTE ST 2022 datagrams in order to provide failover protection, as defined by SMPTE ST 2022-7.

SMPTE ST 2022-5 Forward Error Correction creates redundant row and column FEC packets which are used to correct errors in the video data packets. Using FEC is a trade-off between error recovery ability, extra bandwidth required, extra processing needed along with the associated receiver latency caused by the extra processing required.

SMPTE 2022-7 Seamless IP Protection Switching is enabled by IGMP multicasts and provides a method to clean switch RTP packets using frame numbers. It can tolerate the complete failure of one network path. With seamless (otherwise known as “hitless”) failover, the receiver selects packets from the main or backup streams in order to produce an error-free output, at the cost of doubling required network bandwidth. The example shown below shows an error-free output even though the main stream has suffered a total network failure.

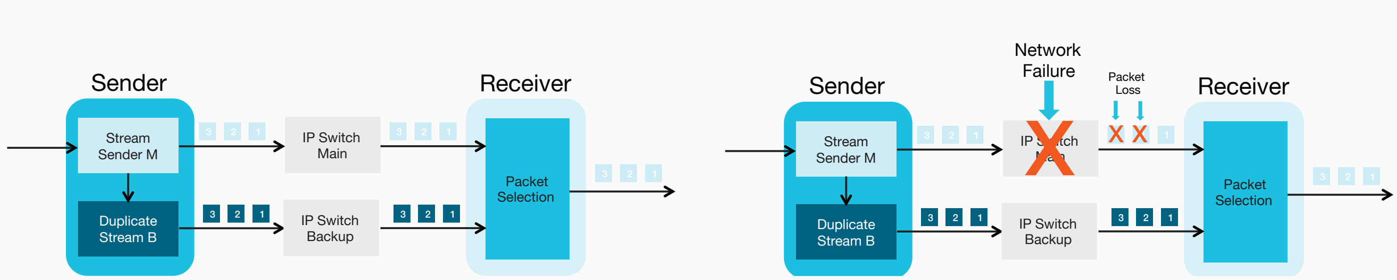


FIGURE 4. SMPTE ST 2022-7 Seamless IP Protection Switching.

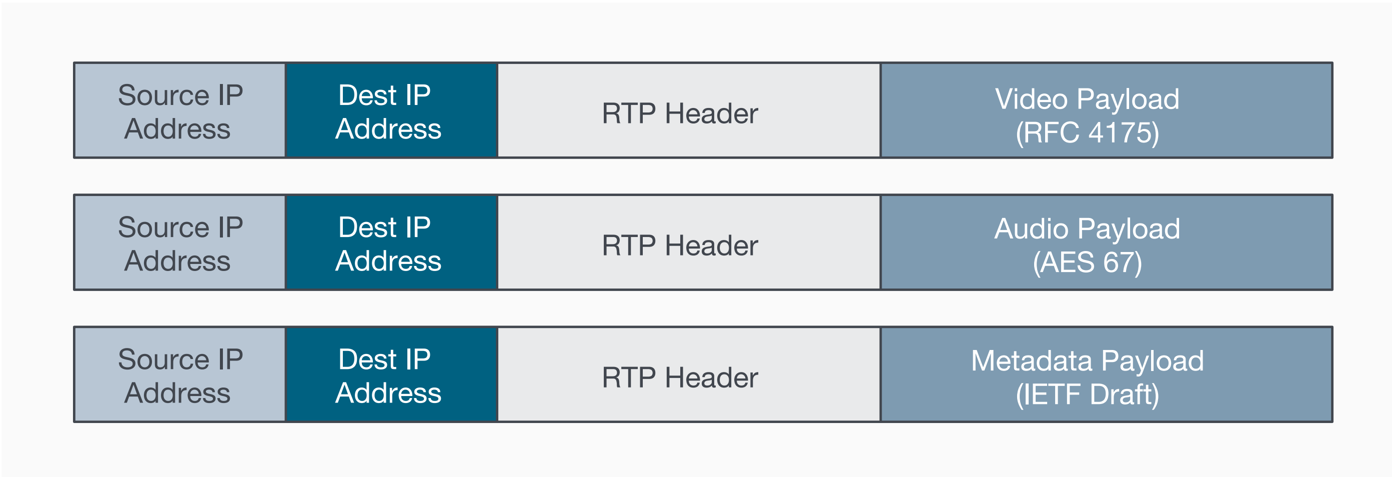


FIGURE 4. VSF TR-03 IP Packet Formats. Dedicated IP Flows Carry Video, Audio and Metadata Essence.

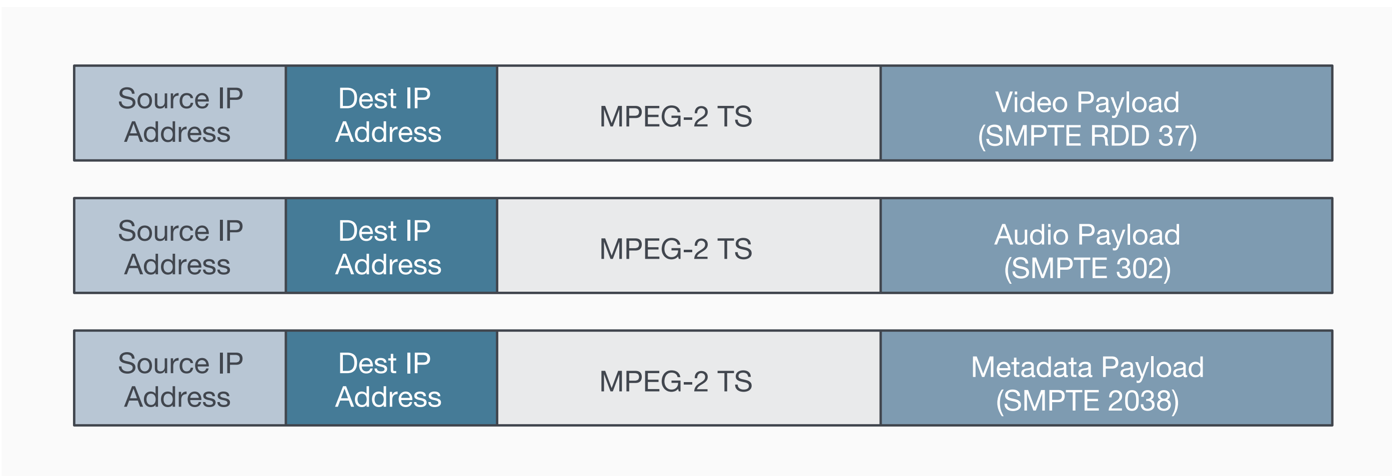


FIGURE 5. ASPEN IP Packet Formats. Dedicated IP Flows Carry Video, Audio and Metadata Essence over TS.

A second method for transporting video over IP networks is defined by the Video Services Forum as VSF TR-03. This differs from SMPTE ST 2022-6, in that it separates video, audio and metadata elements into separate IP flows using RTP/UDP. Advantages claimed for this method are inherent avoidance of audio embedding or wasted bandwidth associated with carrying only video over SDI over IP. PTP synchronization is accommodated through support of both IEEE 1588 default profile as well as the SMPTE ST 2059-2 profile. A related standard, TR-04 defines how SMPTE ST 2022-6 media flows can be used in an interoperable manner within the context of a TR-03 environment.

Another method for carrying video over IP in common usage is the Evertz ASPEN format (submitted to SMPTE as RDD 37), which has some similarity to TR-03 in that separate IP flows are dedicated to carrying video, audio and metadata elements, but in the case of ASPEN, these elements are carried on an MPEG-2 Transport Stream (TS) over RTP/UDP. Similar advantages are claimed for ASPEN when compared to SMPTE ST 2022-6. PTP synchronization is supported by offering compatibility with SMPTE ST 2059-2.

12G into 10G Won't Go – Light Compression

The practical and affordable deployment of 4K/UHD is likely to lead to the adoption of light compression methods for use with 10GigE Ethernet. Although 10-bit High Dynamic Range (HDR) has minimal impact on bitrate, the adoption of 12-bit HDR results in an approximately 20% increase in required bandwidth. It is perhaps obvious that High Frame Rate (100/120 fps) requires light compression to be used with 10GigE networks. All these new technology conspire to drive adoption of light compression methods in order to fit ever more data into a 10G pipe. All compression methods are a trade-off between latency, compression ratio and picture quality. In live production applications only low levels of compression (typically 4:1) are required, whilst conversely, latency needs to be low and the picture quality needs to be of the highest order. Although some have proposed the use of JPEG 2000 for use in production applications, others would argue that it is overly complex for the application and is better suited to its intended contribution application. Block transform codecs (MPEG-2, H.264, HEVC etc.) deliver high levels of compression at the expense of high levels of complexity and latency. The wavelet-based codecs deliver lower levels of compression for high quality or lossless applications, but with much lower levels of complexity and associated latency. There are three methods commonly proposed for use in production applications and all are relatively simple and light weight wavelet compression algorithms. They are the Sony Low Latency Video CODEC – LLVC (submitted to SMPTE as RDD 34); the Intopix Tiny CODEC – TICO (submitted to SMPTE as RDD 35); and VC-2 – also known as Dirac Pro, which developed by BBC research and development and is standardized as SMPTE ST 2042. All these wavelet CODECs are intra-coded and are designed to deliver extremely high-quality video at low levels of compression and with low latency.

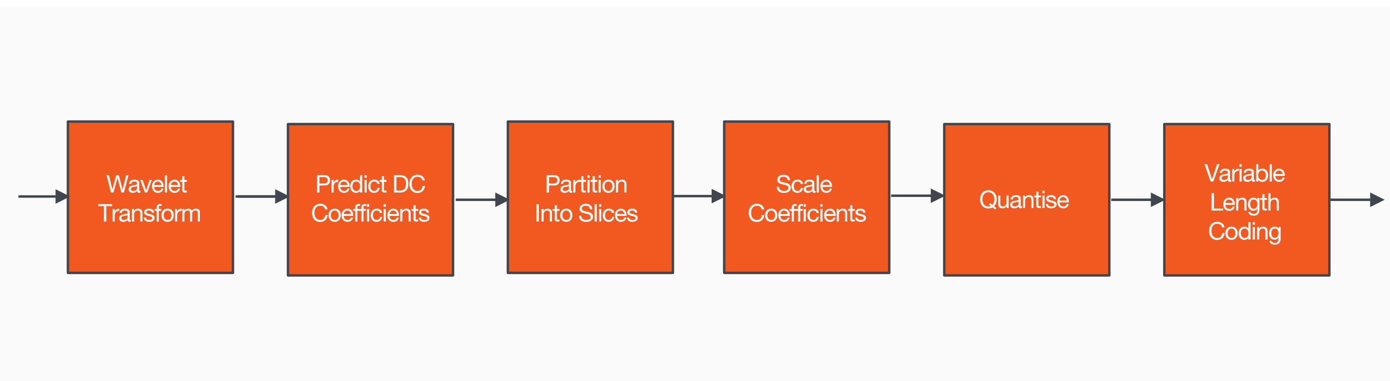


FIGURE 6. VC-2 Signal Processing Chain.

Keeping PTP Simple

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. It should be noted that clock in one domain may not be synchronized to clocks in another domain.

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 Slave. A Master is a device that provides the time in a given PTP domain and a Slave is a device that synchronizes to a Master. A Grandmaster is a Master that is providing the ultimate source of clock synchronization in a network. In the context of broadcast applications, PTP Grandmasters are usually synchronized to GPS, GLONASS or both, in order to derive accurate time-code relative to the 1970 Epoch. It should be noted that PTP Grandmasters always use the 1970 Epoch. To enable legacy equipment support, the Tektronix SPG8000A hybrid PTP Grandmaster and SDI SPG is able to phase its baseband timing outputs relative to either the 1970 or 1958 Epoch dates.

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 Slaves to derive the time. Delay Request messages are a request for timing information and are sent from the Slave

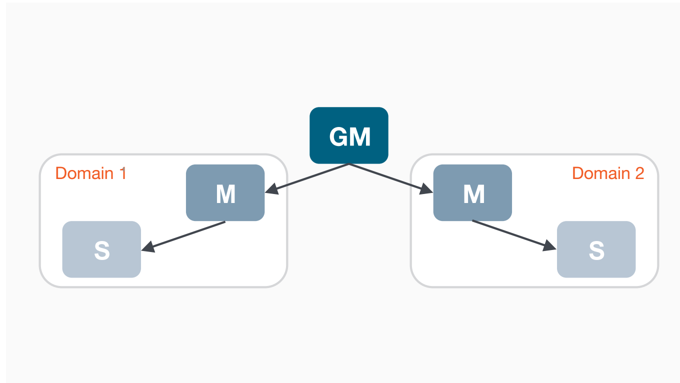
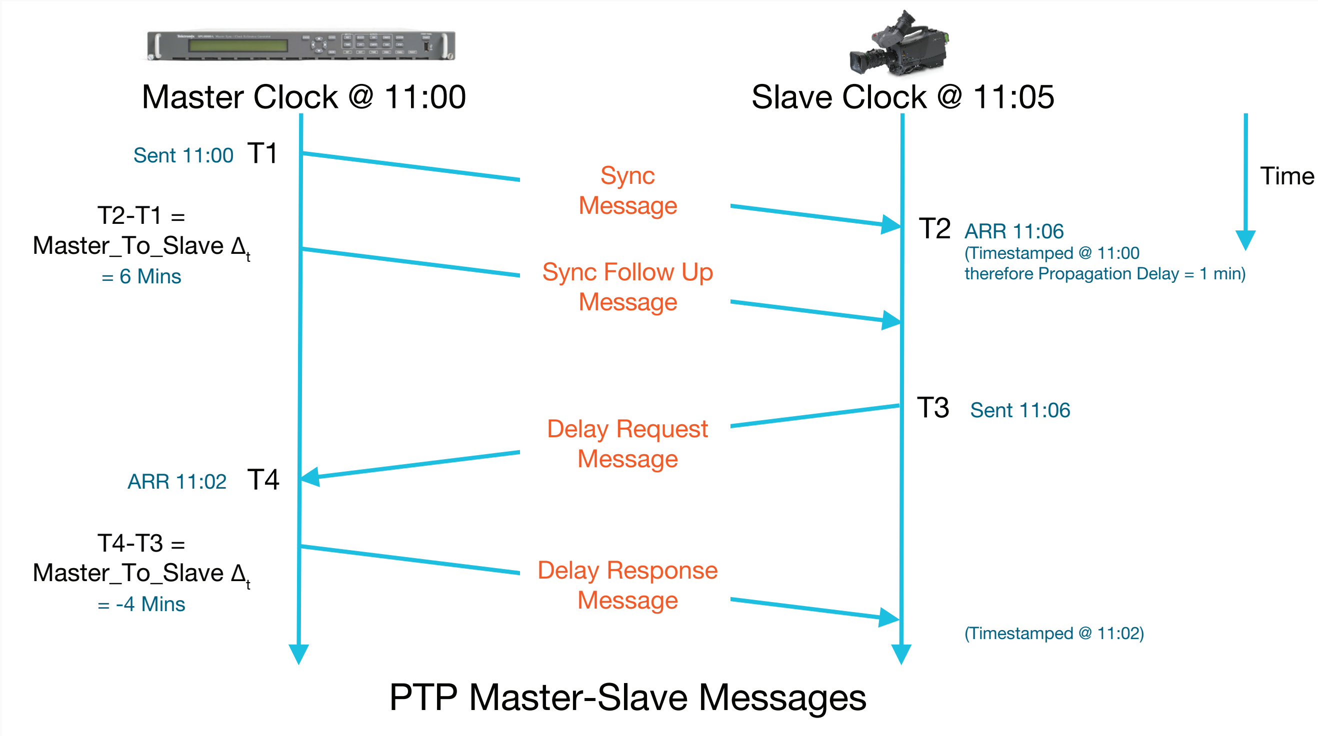


FIGURE 7. PTP Domains Synchronized to a Common Grandmaster.

to the Grandmaster in order to determine the reverse path propagation delay between the Slave 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 Slaves. The Grandmaster periodically transmits Sync and Follow-up messages, which the slaves use to derive the time. In an ideal World the network delay could be programmed into each slave 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 delay in switched/routed IP networks is both variable and asymmetric, so the Slave 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 Slave in a Delay Response message.



Offset = (Master_To_Slave Δ_t – Slave_To_Master Δ_t)/2 **= 5 Mins**

Oneway Delay = (Master_To_Slave Δ_t + Slave_To_Master Δ_t)/2 **= 1 Min**

FIGURE 8. Deriving the Correct Time in a PTP Network.

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

If the propagation delay in both directions is in fact different, then the slave is offset to “correct” for this by adjusting its clock to a value of half the asymmetry. The clock’s control loop adjusts the slave time to make the Master-to-Slave and Slave-to-Master propagation delays appear to be equal. That is, the control loop adjusts the slave time such that T2-T1 = T4-T3.

Accuracy and Reliability is Key – The BMCA

One reason for PTP's suitability to broadcast applications is the resilience provided by the use of the Best Master Clock Algorithm (BMCA). The BMCA allows the most accurate Master to automatically take over the duties of Grandmaster when the previous Grandmaster loses its GPS lock, gets disconnected from the network, or is unable to act as Grandmaster for any reason.

The BMC Algorithm runs on all clocks in a network and uses a number of criteria to determine which Master should be Grandmaster including the following in priority order:

1. User Definable Priority 1 Field (the lowest value <= 128 wins)
2. Clock Class (e.g. GPS vs free running)
3. Clock Accuracy
4. Clock Variance (jitter and wander)
5. User Definable Priority 2 Field (the lowest value <= 128 wins)
6. Clock Source Port ID (usually the Ethernet MAC Address)

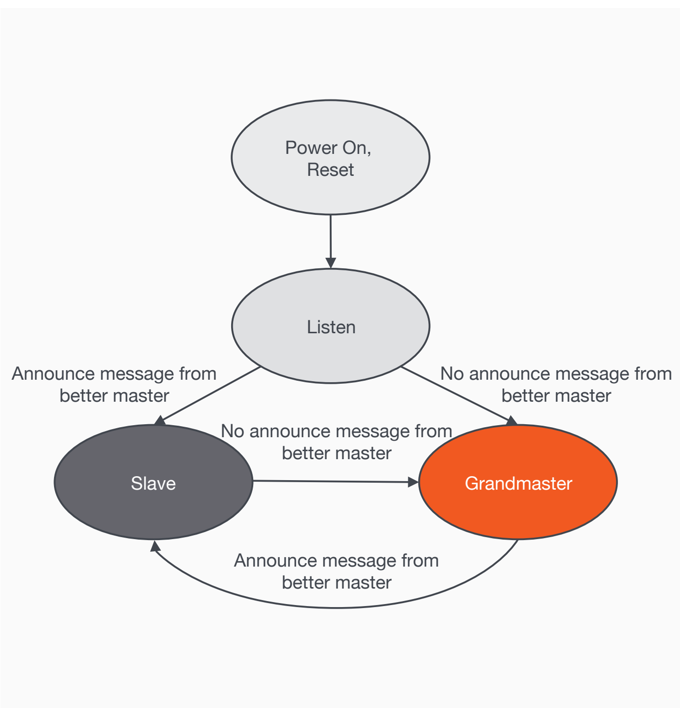


FIGURE 9. Determining Master/Slave Clock State.

Grandmaster Failover

In order to establish an automatic main and backup Grandmaster fail over the Priority 2 field is used to identify main and backup clocks between two or more otherwise identical redundant Grandmasters as follows:

- Main Grandmaster
  (Priority Field 1 = 128; Priority Field 2 = 127)
- Backup Grandmaster
  (Priority Field 1 =128; Priority Field 2 = 128)

If both identical Masters are locked to GPS, they will have the same clock quality, so the lowest Priority Two Field value will select which is the Grandmaster. If the Main clock loses GPS lock, then the Backup clock becomes the Better Master and will take over as Grandmaster.

It is worth noting that if any GPS synchronized Master loses GPS lock, it will of-course itself become free running and will be reliant upon its own internal local oscillator. However good this oscillator is, over an extended period of time it will drift, even if slightly relative to the GPS clock. Once GPS lock re-acquired, unless the Master's local oscillator phase-lock loop (PLL) is driven slowly to re-synchronize with the GPS clock, then the system can suffer from what is known as "Sync Shock" when the Master's clock frequency suddenly changes. Whilst this may be acceptable in some IT applications, this is of course highly undesirable in a video production application. In the case of the SPG8000A, the "Stay Genlock" feature is designed specifically to avoid the problem of Sync Shock through careful control of the PLL.

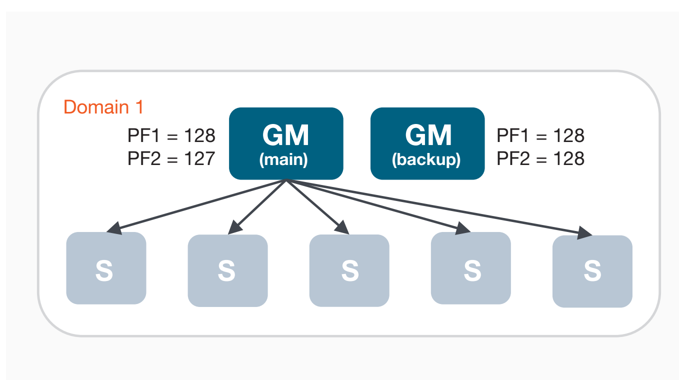


FIGURE 10. Configuration of Main/Backup Grandmasters for Automatic Failover.

Although in theory it is possible to use a Master with software-based time-stamping, in the case of live video production applications, it is highly unlikely that such a device could be devised with the necessary clock accuracy required for synchronous video processing. A hardware time-stamped Grandmaster device such as the SPG8000A is locked to GPS (or GLONASS or both to provide greater constellation resilience), with the Grandmaster's local oscillator being phase-locked to the GPS reference. This local oscillator is the reference clock used with dedicated hardware for the precise timestamp of the incoming PTP messages and PTP sync packets. A dedicated hardware approach is unaffected by operating system behavior or network traffic latency.

PTP Clock Types

Ordinary Clocks are those devices that are at either end of a network and are not switches or routers. A Slave Only clock never acts as a master, whereas a Master/Slave clock can act as either and a Preferred Grandmaster is configured to never become slave.

It is vital that switches and routers in any IP video network that relies upon PTP for synchronization are "PTP Aware". That is they are able to account for their own queuing delay, to ensure downstream timing accuracy. This can be achieved in one of two ways. The first is by the switch acting as a transparent clock which hardware time stamps Sync and Delay Request messages on arrival and departure and adds the difference to a correction field in the message.

The second way for a switch or router to account for its own queuing delay is to act as a Boundary Clock, which receives time from a Master on one slave port and provides one or more Master (not Grandmaster) ports to downstream Slaves in a PTP Domain and in doing so, removes the effect of its own queue.

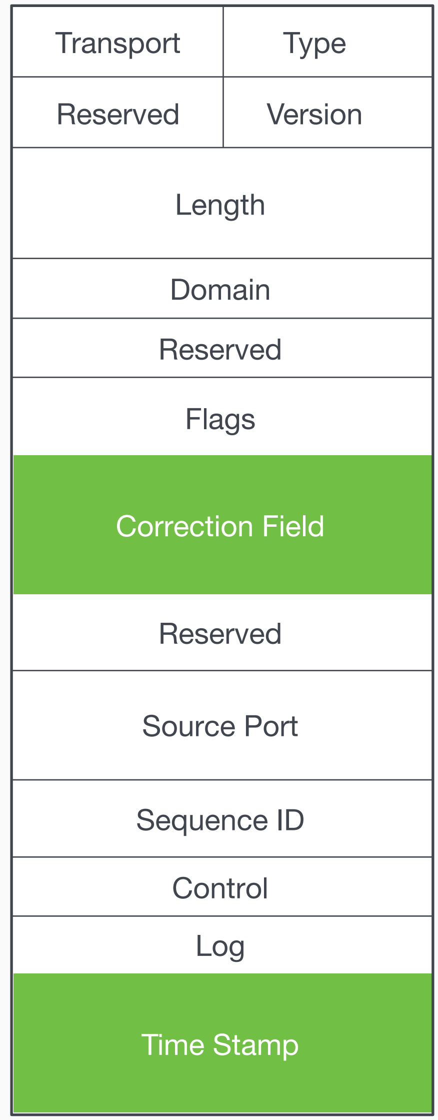


FIGURE 11. Sync/Delay Request Message Format.

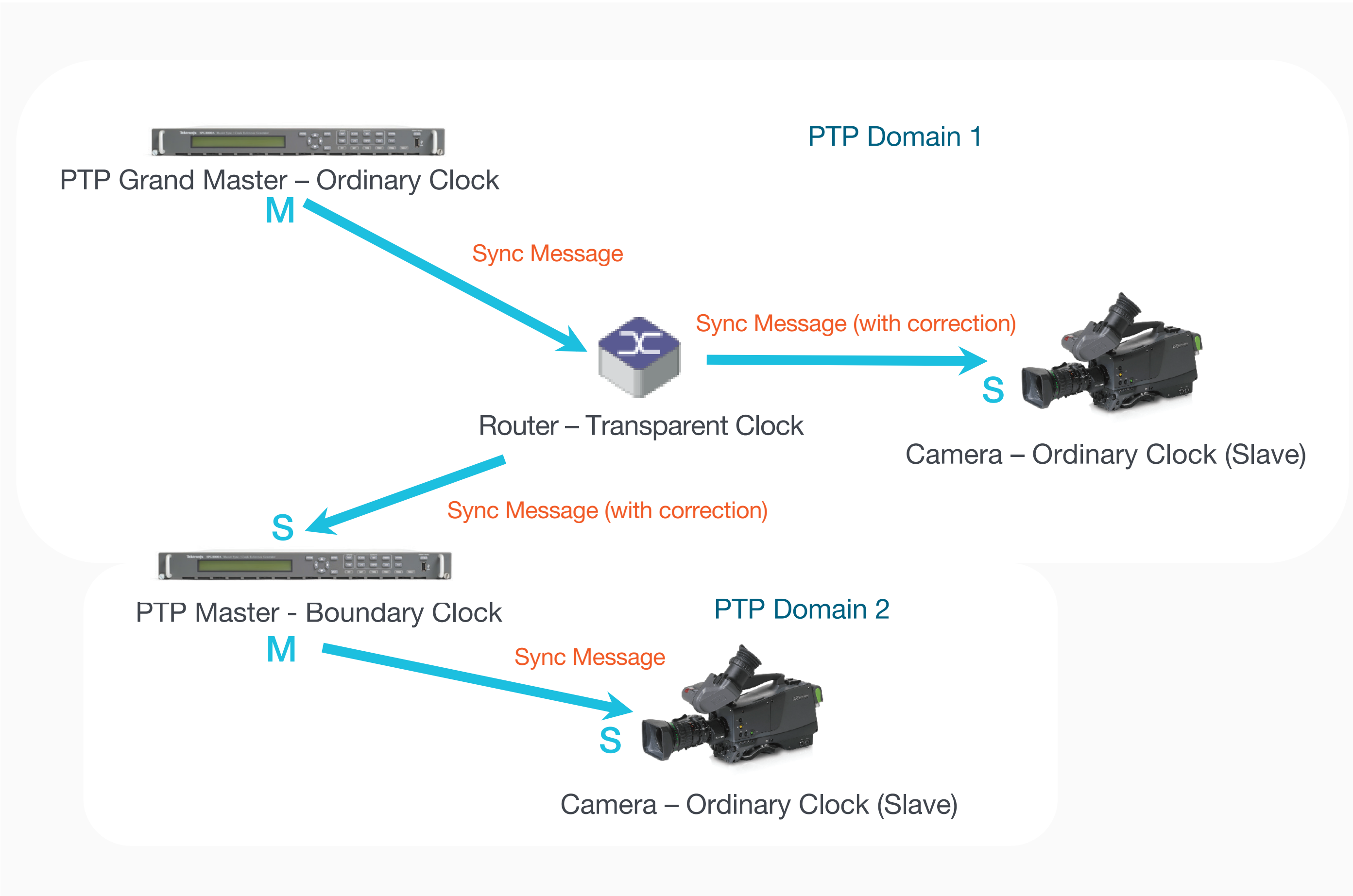


FIGURE 12. PTP Clock Types in a Network.

Summary

Although the transition to IP is seen by some as inevitable, not all equipment is available with IP interfaces. It must also be considered that the investment in SDI equipment has been so considerable, that the use of hybrid IP/SDI networks is likely for the foreseeable future. For broadcast applications, it is essential that the PTP Grandmaster such as the Tektronix SPG8000A provides support for the application specific video and audio PTP profiles, such as SMPTE 2059 and AES67, as well traditional SPG features including black burst, tri-level and SDI out. All the above protocols must be referenced to the same GPS clock, or such a hybrid IP/SDI network would be inoperable. It must also be considered that a broadcast live production network is entirely reliant on a stable reference and any timing and synchronization devices “must work”.

Although many equipment vendors have IP enabled equipment at an early stage of development, both equipment vendors developing IP video equipment and broadcasters and other content providers producing content are reliant on the availability now of an accurate and reliable timing and synchronization solution.

Although the concept of carrying uncompressed (or lightly compressed) video over IP is perceived as being very new, and indeed revolutionary, the precedent for the broad adoption of IT infrastructure for live production facilities has in fact been in place for many years. As was mentioned at the beginning of this paper, IT infrastructure began to be adopted almost fifteen years ago for compressed video distribution using MPEG-2 transport streams over IP. IT infrastructure is also in industry-wide use as the distribution and control component for file-based workflows. In both these cases, Tektronix provided technology support for early adopters; with TS over IP test equipment as well as with the first file-based QC tool on the market.

As such, Tektronix was involved with the earliest adopters of compressed video over IP and file-based QC and continues to be closely engaged with these latest developments with baseband video over IP.

We are at the beginning of a long term transition to IT-based infrastructure and those involved in the production and facility side of video have little experience with the new technology, but conversely are extremely experienced using SDI and all the issues associated with its use. This coupled with a huge investment in existing technology and workflows implies that the transition will take place gradually, making it likely that hybrid SDI/IP infrastructure will be in place for some years. Such production facilities will require equipment that is able to operate seamlessly and reliably in such a hybrid environment. The companies best placed to provide equipment that meets those requirements are those who have experience both of the challenges of the live production environment, as well as extensive experience of the challenges associated with the distribution of video over IP networks.

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