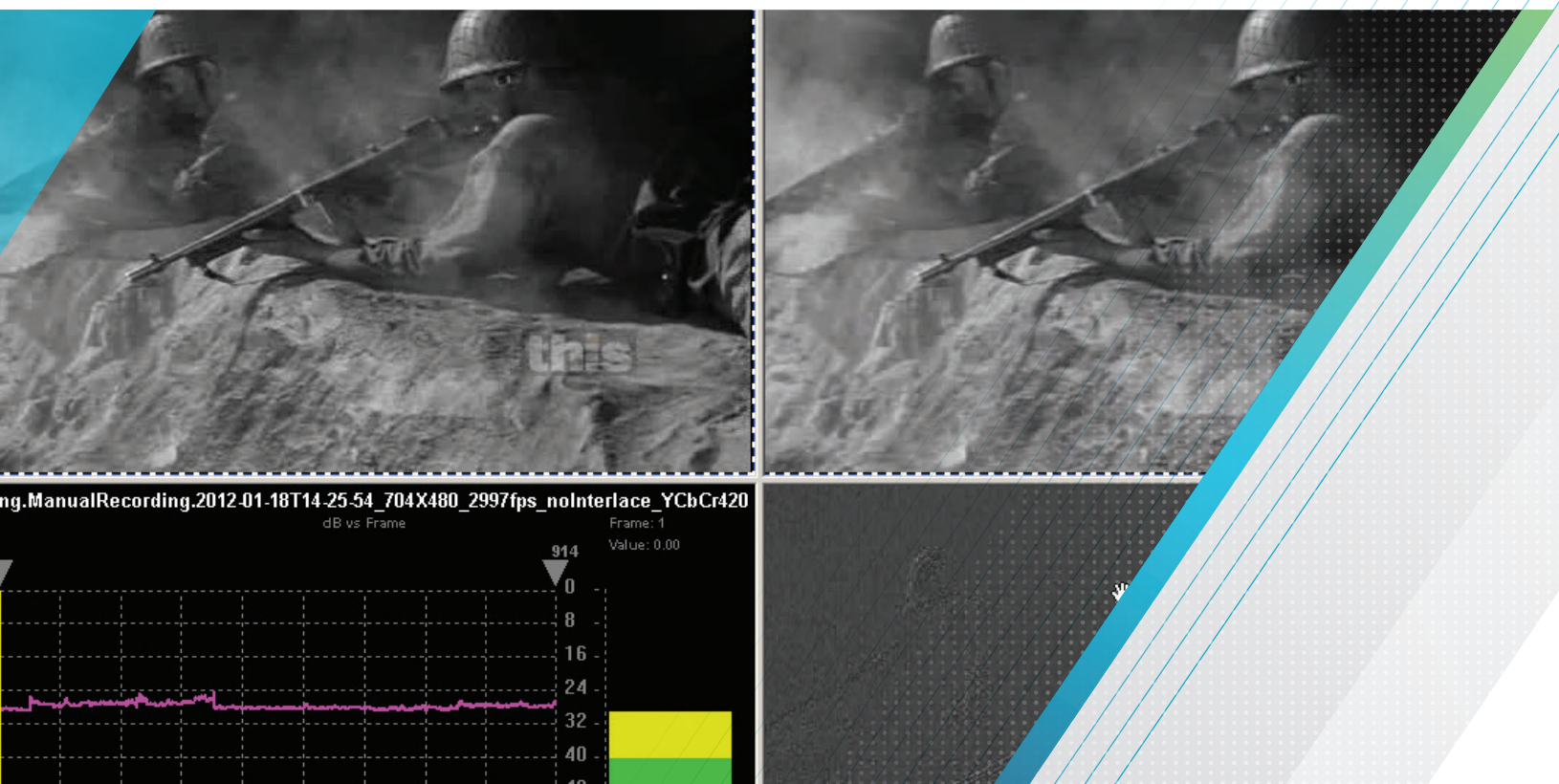


Troubleshooting Quality of Service (QoS) and Quality of Experience (QoE) issues in Video Service Provider Networks

APPLICATION NOTE



Video Service Providers deliver TV programs using a variety of different network architectures. Most of these networks include satellite for distribution (ingest), ASI or IP throughout the facility, and often RF to the home or customer premise (egress). The quality of today's digital video and audio is usually quite good, but when audio or video issues appear at random, it is usually quite difficult to pinpoint the root cause of the problem. The issue might be as simple as an encoder over-compressing a few pictures during a scene with high motion. Or, the problem might be from a random weather event (e.g., heavy wind, rain, snow, etc.). In some cases, it is as simple as adding too many 3 dB RF splitters in the home.

No matter where the problem comes from, it is important to be able to quickly identify and log the audio or video problem that occurred, then identify or pinpoint the equipment (or network link) that needs attention. To identify and isolate problems, it

is critical to have access or test points throughout the facility. The minimum set of test points in any network should be at the point of ingest where the signal comes into the facility, the ASI or IP switch, and finally egress where the signal leaves as IP or RF. With a minimum of these three access points, it is now possible to isolate the issue to have originated at either: ingest, facility, or egress.

To begin testing a signal that may contain the suspected issue, two related methods are often used, Quality of Service (QoS), and Quality of Experience (QoE). Both methods are useful in troubleshooting and analysis, but each of the two methods quantify issues using completely different metrics. This application note will focus on both types of testing methods using the Tektronix MTS4000 MPEG Analyzer, and will explain the benefits and faults with each type.

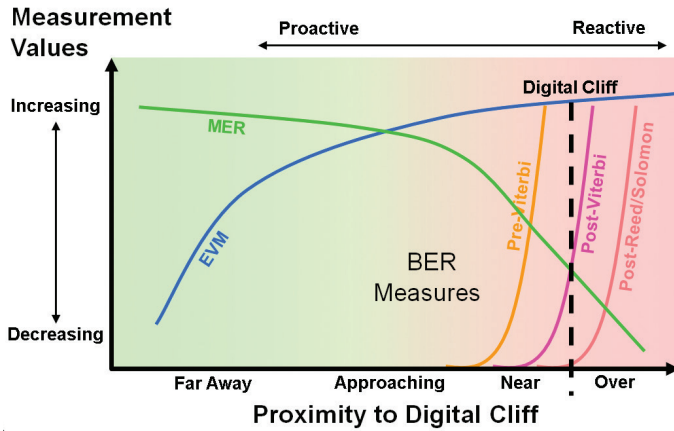


Figure 1. As the amount of noise per symbol increases, the MER drops, bringing it closer to the digital cliff.

What is Quality of Service (QoS)?

One simple explanation of QoS is simply the ratio of good bits to total bits. This is also related to the bit error ratio (BER). The optimum method for testing a transmission link is to take it out of service in order to make use of special test patterns. These patterns are repetitive, making it easy to quantify when a bit or byte is in error. The problem with testing live TV programs is that there is very little repetitive data to test. The use of BER is critical in monitoring the satellite signals because it does carry a small amount of redundancy. To make any additional tests beyond BER, we need to look to the MPEG^{1,2} and RF^{3,4,5,6}/IP standards for QoS as it relates to digital TV. Each RF standard has its own modulation requirements and characteristics. The ETSI TR 101 290⁷ document focuses on a variety of modulation types and defines measurement algorithms for modulation error ratio (MER) as well as many other RF and transport measurements.

Before we begin with the more complex measurements, let's agree to start with one basic, but extremely important RF test - signal power. TV receivers are designed to work with a reasonable amount of signal power. Too much, or too little causes the receiver to fail. Next, the receivers are designed to work with reasonable amounts of noise. With these two measurements, we get the signal to noise ratio (SNR). The higher the SNR, the easier it is for the receiver to recover the transmitted bits or symbols. The lower the SNR, the more probable the bit will be received in error. For digital TV transmission measurements, we refer to MER rather than SNR. As long as the receiver has a high MER, then the QoS is assumed to be very good. The problem begins when the receiver is near the fringe area of reception, or when low power and high noise corrupt the signal. This is often referred to as the "digital cliff" area. When approaching this point,

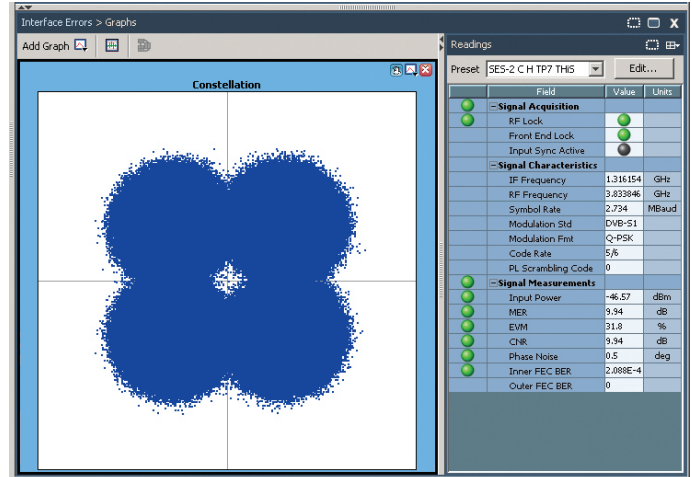


Figure 2. Even though this DVB-S QPSK signal includes high amounts of noise (poor MER) and some symbols landing in the wrong quadrants, FEC and R/S have effectively corrected all errors (QEF).

some of the symbols are incorrectly received. Figure 1 shows MER gradually decreasing as it approaches the cliff. After the MER decreases enough to cross the cliff, the TV picture and sound go from great to terrible. The first two important RF QoS measurements are signal level and MER.

Transmitting digital TV over RF is assumed to occur in relatively hostile domains, so redundancy is always added to overcome the loss or corruption of good symbols. Some digital RF modulation methods provide varying degrees of redundancy to account for different weather or interference conditions. The term used to describe this redundancy is called Forward Error Correct (FEC, or inner FEC). One additional layer for digital TV is called Reed/Solomon (R/S, or outer FEC). This inner code is the Viterbi convolutional coding and the use of both together is often called "concatenated" FEC coding. R/S can correct up to eight bad bytes in each transport packet of 188 bytes for DVB broadcasts (satellite, cable, and terrestrial). ATSC⁶ Terrestrial and SCTE⁸ Cable use slightly different amounts of R/S protection, but are also in less harsh environments and travel much shorter distances than satellite signals (ten bytes per 187 for ATSC 8VSB, three bytes per 122 for SCTE cable). For most RF transmissions, 100% of the digital transport can be received "effectively error-free" or "quasi-error-free" (QEF) as long as the RF bit error ratio (inner FEC) is better than $5 \times 10E-3$ (i.e., 995 good bits for every 1000). R/S will take care of the rest resulting in zero received errors. In this case, the QoS of the demodulated signal could be considered perfect because 100% of all bits sent have been received error-free. Figure 2 shows a signal with a BER better than $5 \times 10E-3$, and therefore, the received transport is considered to be quasi-error-free.

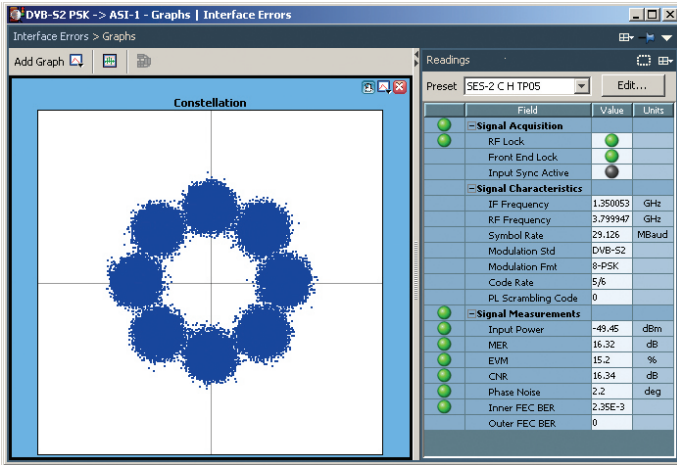


Figure 3. DVB-S2 8PSK signal with high amounts of noise, but enough Forward Error Correction to provide a quasi-error-free transport stream.

Much of today’s video satellite transmission uses QPSK (e.g., DVB-S, DigiCipher II, DSNG, etc.) to send a transport stream from point to point, or point to multipoint. Recent technology improvements such as 8PSK from DVB-S2⁴ allow more bandwidth while using the same frequency spectrum as a QPSK signal. Figure 3 shows an example of a DVB-S2 signal carrying a 70 Mbps transport stream with 31 H.264² SD programs. Similar to the QPSK example, there is a significant amount of noise in each of the eight unique symbols, but enough FEC and R/S has been provided to effectively correct the misplaced symbols (QEF).

Similar digital modulation schemes can be used on the egress side for sending a transport stream over cable. Figure 4 shows a 256 QAM signal with well-behaved symbols in its constellation display.

Continuously measuring BER is a good thing, but when the error ratio exceeds $5 \times 10E-3$ and one or more errors make it into a transport packet (ISO/IEC 13818-1 – MPEG-2), it is almost impossible to tell what will happen to the video and audio. It might make no difference at all. If the error ratio is high enough ($>5 \times 10E-3$), then the errors will be landing in many of the audio and video frames. With this level of errors, it can be assumed that the problems will be noticed by the all viewers. The more difficult issue in monitoring a digital network is determining the extent of the problem on less frequent errors.

When monitoring ASI⁹ or video over Ethernet¹⁰, FEC is usually not performed as the delivery does not take place in a hostile environment. Therefore, it is assumed that the BER is always



Figure 4. 256 QAM cable TV signal.

0.0. It would be nice if this were always the case, but given the complexity of IP switches, occasional failures or overloads do occur and cause IP packets to get dropped. And to make matters worse, the low-latency requirement mandates UDP (send and forget), which is unlike TCP/IP which uses a handshake for every IP packet. Video over IP commonly puts seven 13818-1 transport packets into each IP packet, so the occasional loss of a single IP packet translates into seven transport packets being lost. If video over IP used RTP, then the additional counters in the IP header make it trivial to know when an IP packet was lost. But, since most video over IP uses UDP, then we need another method to know when an IP packet has been lost. With MPEG-2 transport packets, each TV program includes separate 4-bit counters for each audio and video element, so it is possible to determine when a packet might have been dropped. Calculating the ratio of missing packets to total packets is possible, but not too helpful as each missing packet will usually generate audio or video degradation. The indication of a single or occasional missing transport packet is a very bad thing, so once the event occurs, troubleshooting usually begins immediately. The magnitude of the degradation will depend upon where the error occurred within each video frame.

Beyond RF and BER testing, the TR 101 290 document includes a recommended set of transport stream measurements for broadcast operators (section 5 of the standard).

- Priority 1 errors mean that the receiver will not be able to lock the signal.
- Priority 2 errors imply that quality of video and audio may be impaired.
- Priority 3 errors imply a problem in the electronic program guide.

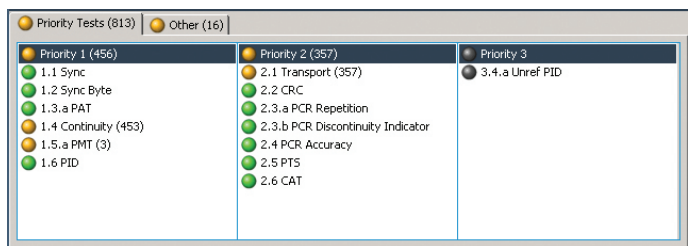


Figure 5. TR 101 290 table tabulating several corrupt TS packets as well as lost or missing packets.

This parameter set was chosen in an attempt to quickly determine the QoS of a live TV signal. From a troubleshooting and diagnostics point of view, the most critical of these tests are Sync Byte Error, Continuity Counter Error, and Transport Error Indicator Flag. Any errors found in these three categories usually means that something very bad is happening in the transmission of the stream, or possibly in the building or multiplexing of the stream. The TR 101 290 test parameters are a great way to quickly get an idea of the health of the transport stream and its audio and video elements, but some of the other tests are often misleading. Most important are the timing tests of the MPEG-2 Program Association and Program Map Tables (PAT and PMT) tests. Without these two tables, a normal digital TV or set top box would fail to decode a program. Figure 5 shows an example table from the MTS4000 MPEG Analyzer with the TR 101 290 Priority 1 and 2 tests. The green LEDs show a perfect error-free history, and the amber LEDs and their counters show a series of impairments in the past. The 2.1 Transport error means the RF demodulator failed to recover a transport packet (357 times), and therefore, introduced one or more bit-errors in each of the 357 transport packets. The 1.4 Continuity Counter errors (453 times) means that 453 transport packets were lost from the stream, or possibly the counter was corrupt during a 2.1 Transport error. Errors of this type and magnitude would normally cause video and audio artifacts or impairments.

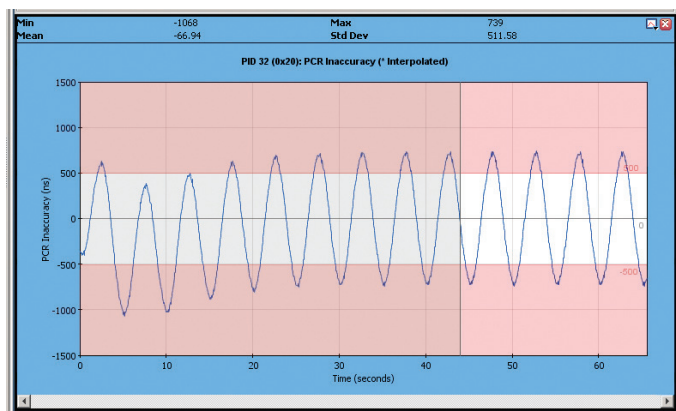


Figure 6. TR 101 290 PCR Inaccuracy test.

It is true that PAT and PMT are needed (Priority 1), and a minimal arrival time interval requirement is good, but if the tables arrive just one millisecond late, then the TR 101 290 display goes from green to red. This is considered to be a critical error, even though the extra millisecond of latency in the table is never noticed by the TV, set top box, or the TV viewer. For this reason, some interpretation of the TR 101 290 results are needed. Relying solely on TR 101 290 can get you into trouble. It is common practice for many network operators to increase the threshold of the arrival time intervals to allow for some deviation, while still testing that the tables are arriving at some rate or interval. Keeping the interval times tight will ensure end users can change channels quickly, but TR 101 290 might falsely flag a problem when a stream or program has a slight delay in parts of its electronic program guide. The same goes for many of the program clock reference (PCR) measurements in priority 2. A deviation of a few percent in the interval will not make a difference to virtually every TV or set top box, but this deviation will cause TR 101 290 to change from green to red. Figure 6 shows a graph where the PCR Accuracy is out of specification by a few nanoseconds (deviations outside of the white band).

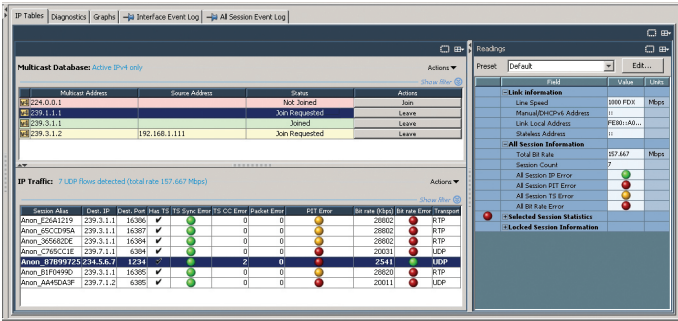


Figure 7. MTS4000 tracking IP packet gap, packet loss, packet rate, etc.

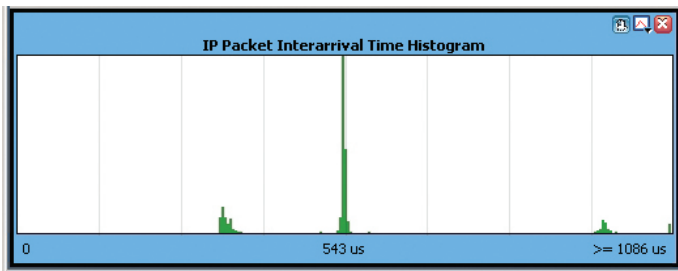


Figure 8. High resolution histogram of selected IP stream.

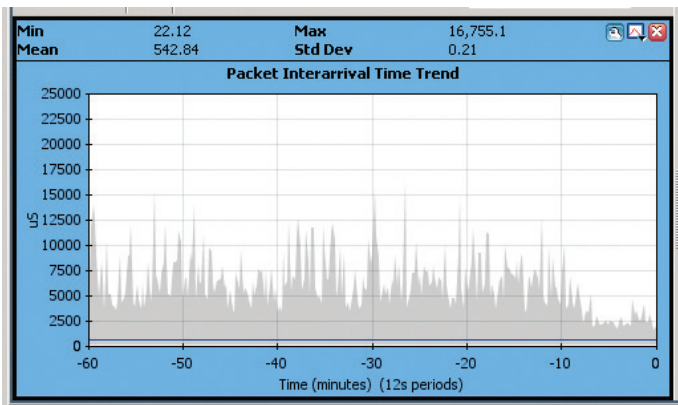


Figure 9. Plotting jitter trend up to 7 days.

Given that most facilities today route MPEG over IP, it may become necessary to diagnose transports in an IP network as well. All of the same TR 101 290 tests can be applied to transports over IP, but it is critical to pay attention to massive latency issues between IP packets. RF modulators that use IP as an input will require consistent arrival times in order to keep the buffers from overflowing or underflowing. Therefore, measuring the gap between any two IP packets can be critical. Figure 7 shows the MTS4000 measuring the Packet Interval Gap on several MPEG transports, all in parallel on a GigE link.

A high resolution histogram plot is also available for detailed jitter measurements on IP signals. Figure 8 shows a mean gap of about 543 µs, along with a small grouping around 300 µs and 900 µs.

Tracking jitter or packet gaps over time can be very powerful in proving the maximum gap found over a 7-day period. Figure 9 shows the same 543 µs average gap (dark blue line), but there are rare occasions where the gap between two IP packets is up to 16.7 ms (gray envelope).

To measure the QoS of the MPEG-2 transports, TR 101 290 will suffice. Determining the quality of the audio and video (QoE) will require much more processing than simply looking at MPEG transport packet headers and packet gaps.

Given that compressed TV is not linear like uncompressed audio, a bad bit landing in the middle of an MPEG video I-frame is much worse than a bad bit landing in the corner of a B-frame. The reason that this issue is not considered linear is because an error in an I-frame may reside on the TV screen for an entire Group of Pictures (about 500 ms). With a bad B-frame picture, it only stays on the TV screen for one frame (e.g., 1/30th or 1/25th of a second). Also, a bad bit could cause an entire slice or rows of pixels to turn green. Or maybe the bad bit only causes a signal pixel to change color. Therefore, unlike in telecom testing where BER was an adequate measurement, in digital TV, we need to measure BER, and also keep track of impairments within each audio and video element.



Figure 10. Low blockiness from satellite distribution feed.

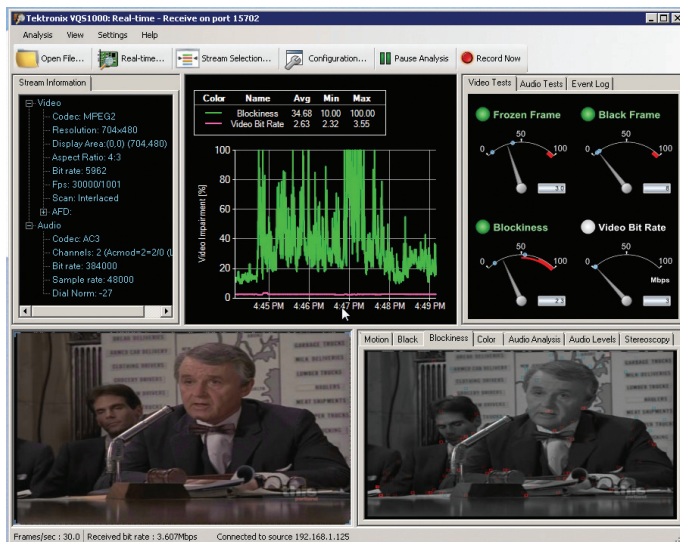


Figure 11. Transcoded content with significantly higher blockiness (green plot) over the same timeframe.

What is Quality of Experience (QoE) ?

QoE is a metric that relies much more upon our ability to notice a problem in the midst of a TV program rather than a single bit error. For instance, if a single bit error affected the far corner of a B-frame, then that bit error would have a much lower impact to QoE than if the error had landed in the middle of an I-frame. Subsequently, if the single error in the I-frame ended up corrupting an entire slice of 16x16 pixel blocks, then the impact to QoE would be huge. Therefore, to be able to measure the impact of an error upon a TV program, it is critical to know exactly where the error occurred. Another issue is that the transport may be perfectly error free (no syntax or semantic errors), but the video is objectionable because there is not enough bandwidth to clearly portray the high-motion video scene. In this case, there are no protocol errors, but the picture frame or video sequence is made up

of large 16x16 solid squares rather than a clean picture. Viewers often refer to this as a blocky video problem. Figure 10 shows a low blockiness measurement on a TV program from satellite distribution at 4.1 Mbps. Figure 11 shows the same TV program over cable (256QAM), but only after it has been transcoded (or rate-shaped, clamped, etc.) to a lower rate of 3.6 Mbps. The resulting changes cause significantly more blockiness artifacts over the same period of time. In this case, the QoE has deteriorated enough that viewers would recognize the blockiness issues in the video.

Another video QoE issue occurs when the same video frame occurs repeatedly for a long period of time (frozen or black frames). Obviously, video frames repeat when there is no activity, but at some point (e.g., 2 minutes), an alarm is needed to alert the operator if the frozen frames are on purpose (for example an all black scene for artistic effect), or from a broken link or piece of equipment.

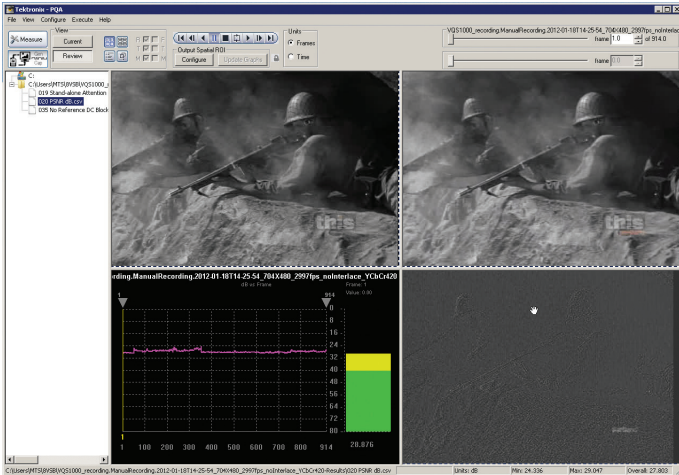


Figure 12. The Picture Quality Analyzer performs direct comparisons of video frames before and after a live transcode.

When a reference signal is available for direct comparison as in the case shown above from satellite and cable, an objective picture quality rating can be made using the Tektronix PQA600 Picture Quality Analyzer, or from the Picture Quality Analysis software option in the MTS4000. In this case, a stream is recorded from both satellite and cable, and then the Picture Quality Analyzer will convert to YUV, perform spatial and temporal alignments, and then measure the difference between the two video clips. The basic Peak Signal to Noise Ratio (PSNR) measurement is shown, but the Picture Quality Analyzer also performs many more important tests such as Picture Quality Rating, Attention model, etc. Figure 12 shows the Picture Quality Analyzer making a frame by frame PSNR comparison between every frame of the satellite and cable video clips.

The Picture Quality Analyzer has been pitted directly against human view trials in a variety of studies. The IRT study¹¹ (Institut fuer Rundfunktechnik in Munich, Germany) showed extremely high correlation between the PQR results and human viewer results using ITU-R_BT.500-11¹² testing.

Audio is another element of QoE as it occasionally increases to an unbearable level, or maybe drops out altogether over a long period of time. To accurately track these problems, loudness measurements have been defined for instantaneous peaks (ITU-R BS.1770/1771¹³), as well as short term and long term filters for ATSC¹⁴, ITU, and EBU¹⁵. These measurement algorithms allow test equipment to track audio over long periods of time and trigger on deviations outside allowable levels. One method of tracking this is to use a DialNorm level as a target for normal dialog levels. Averaged or filtered levels are allowed to rise or fall within a few dB of this set level. Many

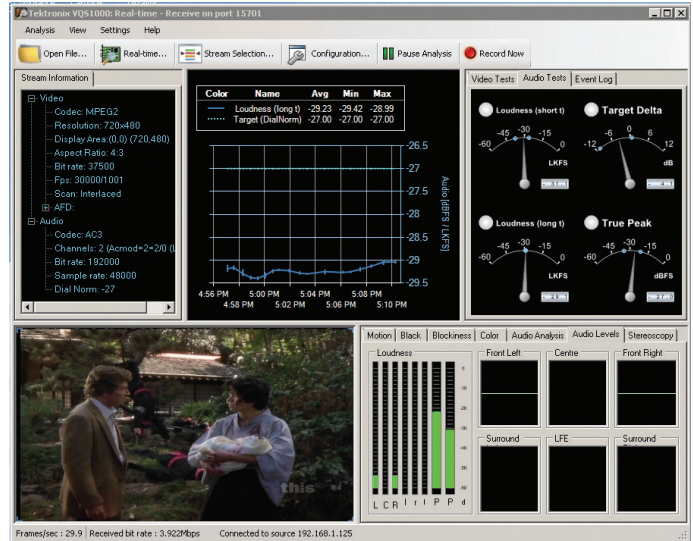


Figure 13. Satellite ingest with audio loudness at 2 dB quieter than DialNorm.

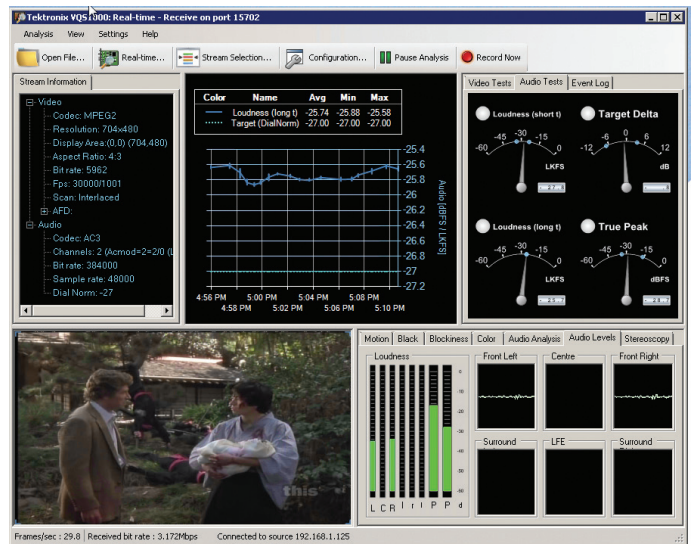


Figure 14. Cable broadcast with audio at 2 dB louder than DialNorm. This is a 4 dB increase over the satellite feed.

governments have adopted this requirement and now impose fines on broadcast operators who deviate beyond the agreed limits. With this agreement in place, TV viewers should now be able to switch from channel to channel without having to adjust the audio levels. The same goes for the commercials or ad-inserts between programming. The loudness of commercials must also stay within the agreed upon levels. As an example of audio levels being altered, figure 13 shows the satellite distribution signal at one audio level (-29 LKFS¹³) while its cable-broadcast counterpart (figure 14) is at a different audio level (-25 LKFS). Both signals use a DialNorm reference level of -27 LKFS.

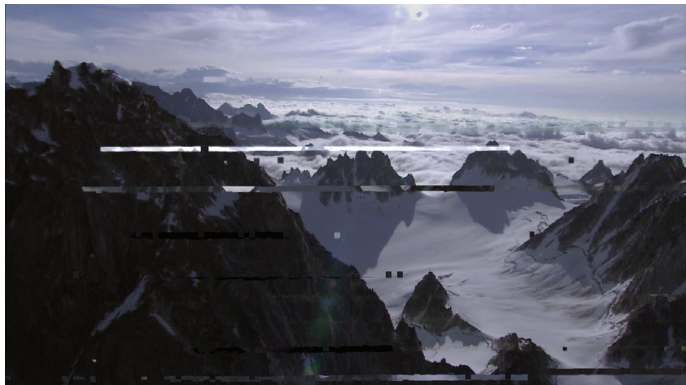


Figure 15. Significant loss of RF or IP packets cause blockiness and slice errors.

How are QoS and QoE interrelated?

Both QoS and QoE are interrelated, but their relationships are not equal. Case in point:

- 1) A good QoE usually means a good QoS
- 2) But a good QoS does not always mean you will have a good QoE. For example, Encoder bandwidth starvation or over-compression.
- 3) A bad QoS, or at least frequently occurring QoS issues usually lend toward a poor QoE. With enough bad bits randomly distributed into all video and audio elements, poor quality of video and audio is bound to occur (see figure 15).
- 4) A bad QoE is not always related to a bad QoS, but it is always worth validating the QoS performance. Poor QoE can come from misbehaving ad-inserted and any other equipment (not related to TR 101 290). Poor QoE can come from over compression (this is perfectly legal). Poor QoE can come from dropped IP packet, corrupt RF packets, or other transmission problems.

Therefore, it is common to measure both QoE and QoS in order to quickly identify the root cause of a problem and in turn correct the issue.

One problem with QoS issues is that they can be easily hidden or masked when the program is handed off from one network to another (e.g., dropped video packets). Once the program is decoded back to baseband for manipulation and re-encoding, all previous QoS errors are lost forever. After encoding a second time, the transport and video protocol is now error free, but visually, there may be large slice errors seen in the middle of the video frames. At this point, then only thing available to catch these artifacts is to decode the video and audio content and look for anomalies. A common problem

in sporting events is when the remote transmission link is lost for a fraction of a second. The video breaks apart, but the receiver is able to catch this QoS issue due to Transport Error Indicator flags (TR 101 290 Priority 2.1 – Also bit #9 in each transport packet). Although, if the decoded and blocky or broken video is passed on to the network as a re-encoded program, then the program will often have a perfect QoS, as well as its compressed video syntax being perfect due to its decode and then second encode process. The video may still look like garbage (as seen in figure 15), but according to QoS and video syntax, the results are error free. To find an artifact, we must actually look at the decoded picture content and determine if the frames contain objectionable artifacts (blockiness, slice errors, frozen frames, etc.).

If we have a high BER or poor QoS, then it is inevitable that the QoE will deteriorate also (as in figure 15). In this case, we do not care which frames the errors are landing in because the rate is high enough to land in every frame. Therefore, the QoE rating becomes highly objectionable. In this case, the QoS measurement is the key to isolating the problem.

When the QoS is quite good (no transmission errors), then there is a very low correlation between QoS and QoE. In this case, it is more important to focus on the QoE.

There is an important case where the QoS can be perfect, but the QoE is highly objectionable. This is due to the common case with networks running constant bit rate (CBR) video. Normal video with low motion may look decent, or even very good. But the scene occasionally changes from slow moving objects (e.g., talking head) to something with a high rate of change (e.g., sporting event, quick successive scene cuts, etc.). If the encoder does not have enough bandwidth to accommodate the transition to the high-motion events, then the only recourse for the encoder is to start decimating the 8x8 blocks of video. At its worst, the 8x8 blocks may only contain a single value representing the video luminance and color. This scenario is important for QoS and QoE testing because the end-user notices a very objectionable TV program segment, although the difference between over compression and near total loss of signal are almost indistinguishable. Figures 10 and 11 show a program at both 4.1 and 3.6 Mbps with the lower rate having much more blockiness due to high motion in a bandwidth starved stream. Figure 15 shows blockiness too, but due to a completely different reason (missing packets). Therefore, QoE testing is extremely important, but testing QoS helps to determine the difference between malfunctioning equipment causing a problem verses a poor transmission link.

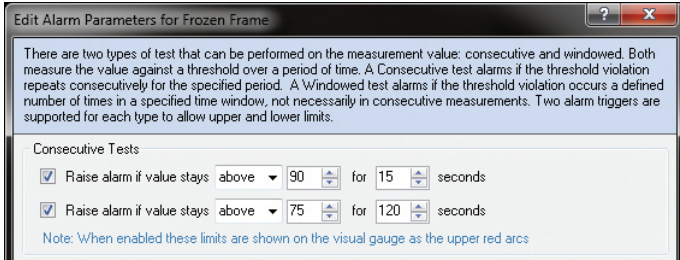


Figure 16. Dual trigger levels for Frozen and Black video.

Using QoS and QoE to identify “real” problems

Most facilities use an ingest, manipulate, and egress architecture. Cable headends use a wide variety of video equipment to capture, manipulate, and ultimately, broadcast. Most all test points would be in the form of QPSK/8PSK, 8VSB, QAM, IP (GigE or 10G), or ASI.

Monitoring QoE aspects on live programming falls into several different audio and video categories.

Video QoE

Video QoE includes monitoring video for frozen frames, black, and blockiness. In the case of frozen frames, a long series of nearly identical frames should trigger a problem since live TV programming is usually made up from moving video. We know that there are many occasions where it is acceptable for a frame to be repeated many times, but at some point, the operator should be alerted to excessive repetitive frames. It could be the sign of a broken encoder or multiplexer, or any of a number of pieces of hardware. Figure 16 shows the VQS1000 QoE software setup for Frozen Frames that will be triggered if 90% of the frame is frozen for more than 15

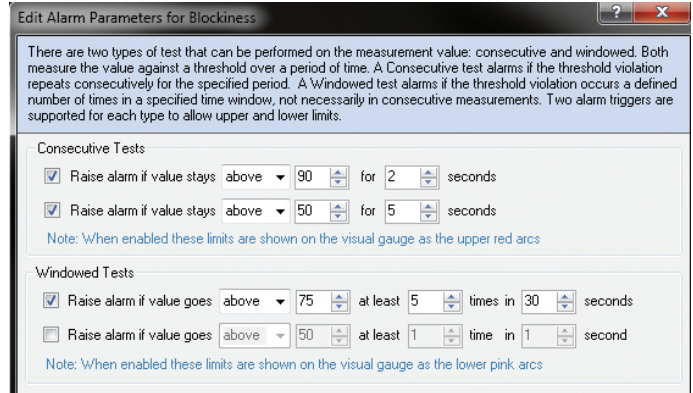


Figure 17. Dual trigger levels and gating for Blockiness.

seconds, or when 50% of the frame is frozen for more than 120 seconds. The requirements are similar for excessively long periods of black. Blockiness is a little different in that it can occur in a small portion of the picture, or over the entire picture. Blockiness will often occur during short scenes where the video content is moving too quickly for the encoder to faithfully compress all of the details. Therefore, the encoder tends to throw away high frequency details rather than failing completely. In this case, the monitor must be set to trigger once the blockiness level has crossed a threshold, and then maintain that level for a significant amount of time. Otherwise, the monitor would trigger an alert every time the scene was overly blocky, even for one single frame. Figure 17 shows the VQS1000 QoE software setup for Blockiness triggering when

- level of 90% is reached for over 2 seconds, or
- level of 50% is reached for over 5 seconds, or
- level of 75% is reached at least 5 times within 30 seconds.

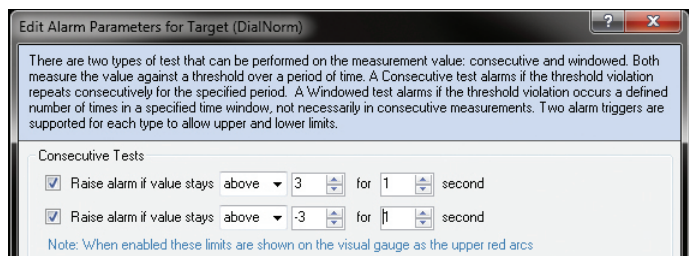


Figure 18. VQS Target DialNorm settings.

Audio QoE

Audio QoE includes monitoring audio for loudness levels. With such a high emphasis on audio levels today, it has become critical to measure overall loudness levels according to the new guidelines. Just as with video QoE monitoring, triggers for levels, deviations, and durations are important to minimize the many false positives than can occur. Figure 18 shows audio loudness limits must be within 3 dB of DialNorm.

Troubleshooting Over-compression

The VQS1000 QoE software allows for real-time monitoring of multiple RF and IP signals in both the transport layer as well as the video and audio layer. When blockiness is found to be excessive on in a TV program, the MTS4000 MPEG Analyzer can measure the broadcast program as well as the ingest distribution feed.

When the QoE of a TV program is called into question, the MTS4000 MPEG Analyzer with its VQS1000 QoE software can quantify the magnitude of the audio and video issues. Once the levels are proved to be unacceptable (as seen in figure 11), the MTS4000 MPEG Analyzer can quantify the ingest program (as seen in figure 10), and then a comparison can be made between the two programs. In the example here, the cable broadcast program has a stable QoE or blockiness as long as the content does not contain too much action. Once the scene changes quickly, the blockiness rating spikes dramatically. In comparison, the ingest content is much less blocky and only varies slightly when the scenes change quickly. It can noted that the ingest content is coming into the facility at about 4.1 Mbps whereas the broadcast program is leaving at only 3.6 Mbps. Therefore, the blockiness issue can be identified as a result of decoding and re-encoding without providing sufficient bandwidth to maintain the quality of the original picture.

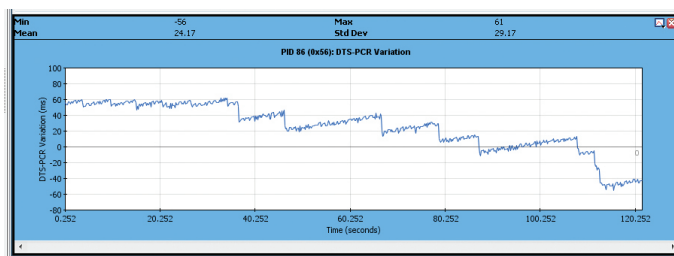


Figure 19. Terrestrial encoder generating negative DTS-PCR delta values (should be impossible).

Troubleshooting Interoperability

The MTS4000 can also be used to identify non-compliance issues when televisions and set top boxes begin to react abnormally to specific programming. In a local broadcast case, several viewers complained about inaudible audio on the SAP channel (audio sounding like motorboats). Other viewers had no problem at all on the SAP channel. It sounded like an interoperability issue, so the MTS4000 MPEG Analyzer was tuned to the local off-air channel to look at its TR 101 290 results. The TR 101 290 tests, and virtually every other test came up clean except for one. The MTS4000 MPEG Analyzer can graph the audio or video Decode Time Stamp (DTS) value as it arrives, against its current PCR value. This delay is always a positive number (representing buffer delay time) and by definition it is required to be between zero and one second. The newly added SAP audio had its DTS-PCR delta ranging above and below zero, which is by definition, never allowed to happen. Figure 19 shows the local broadcaster SAP audio delay issue.

This same terrestrial broadcast feed was also sent to both local cable companies for redistribution.

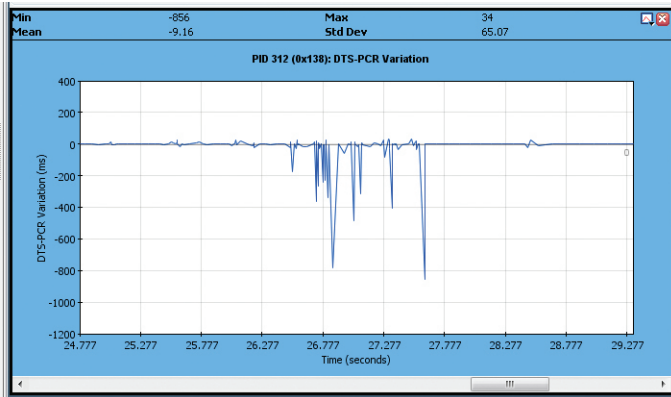


Figure 20. Cable headend encoder generating negative DTS-PCR delta values (should be impossible).

Another similar audio example was found at the cable headend where the audio buffer delay was a negative value, which is impossible for a set top box to support. The MTS4000 MPEG Analyzer once again showed the values going below zero. Figure 20 shows the same audio problem, but at a different facility.

In both cases, reconfiguring the encoder fixed the problem.

The MTS4000 MPEG Analyzer with its unique DTS-PCR measurement was able to quickly pinpoint the problem to the recent configuration change to the encoder, and thus take the blame off the small set of failing decoders throughout the city.

Troubleshooting Transport Streams with CaptureVu

A key part of delivering a quality experience is finding the root cause of problems in the transport stream. It's no longer good enough just to spot a problem, reboot the set-top box and hope that it goes away. The MTS4000 can uniquely go as deep as is needed into the transport and elementary streams to track down sources of picture anomalies, like discovering that a closed caption stream contained too much information, causing set-top box buffers to overflow and precipitate automatic reboots.

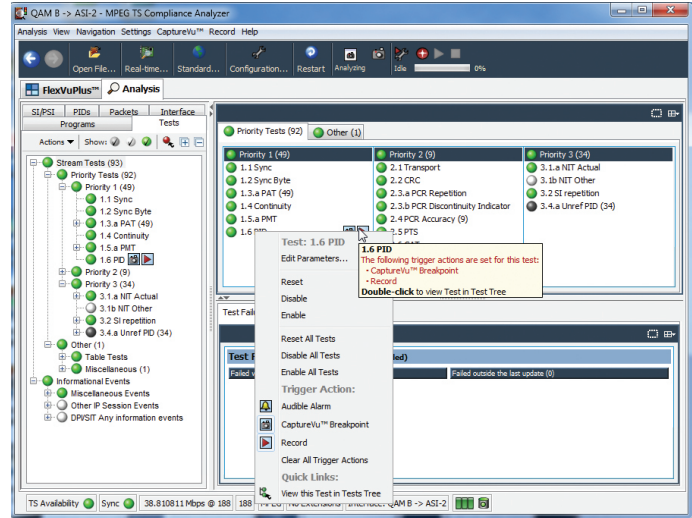


Figure 21. MTS4000 Transport Stream Compliance Analyzer: Trigger and capture enabled for missing encryption key packet.

While the MTS4000 provides real-time analysis, the instrument's ability to capture events for deep analysis is critical to identifying the root cause of problems. Especially useful are the pre-triggers in CaptureVu that not only capture an error, but also provide information leading up to the error.

This last troubleshooting example occurred with a DVR from a major consumer electronics manufacturer that was misbehaving whenever it was tuned to a Switched Digital Video (SDV) channel. The MTS4000 discovered the device was dropping the encryption key whenever a service was added to or removed from the multiplex. Figure 21 shows the MTS4000 TSCA setup screen for enabling a trigger on a specific test or missing PID. By documenting the scenario with screen shots and test reports, the DVR manufacturer was contacted, and they provided a software update that solved the problem and eliminated many expensive truck rolls for the Cable Operator.

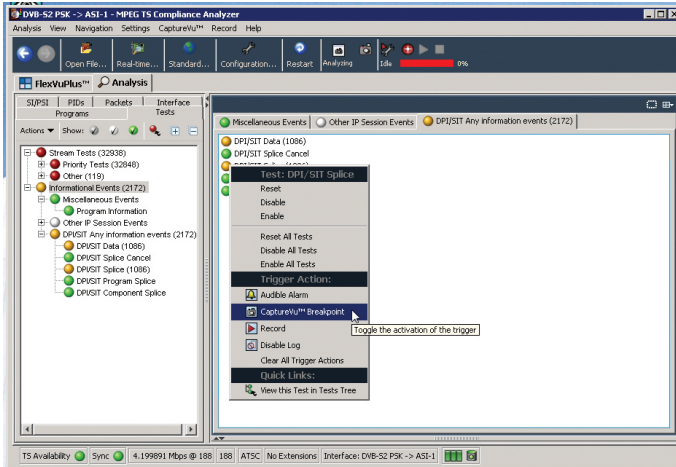


Figure 22. MTS4000 Transport Stream Compliance Analyzer enabling the CaptureVu and Recording for SCTE35 DPI packets.

Troubleshooting Ad Insertion

All broadcast companies fund their businesses based upon selling advertising time between programs. The ads need to be inserted at very specific time, and often timed by SCTE35¹⁶ Digital Program Insertion (DPI) cueing. If these cue tones do not make it through, then the national ads go through and local revenue is lost. The MTS4000 MPEG Analyzer can trigger on the STCE35 packets as well as record programming to disk before and after the event (pre-trigger). Figure 22 shows the MTS4000 Transport Stream Compliance Analyzer enabling the CaptureVu and Recording feature based upon SCTE35 packets. The trigger/record function allows the recorded file to be as small as possible.

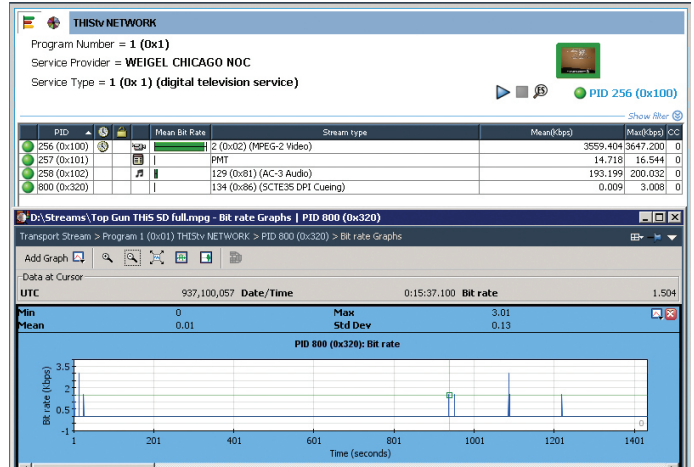


Figure 23. PID 800 DPI packets arrive about every 15 minutes.

In the case of much larger recordings, the MTS4000 can record for many hours. In the example shown in figure 23 running at 4 Mbps, the MTS4000 could hold over 11 days of continuous programming. The figure also shows the PID 800 DPI packets arriving around 15 minutes apart, as you would expect for local ad-inserts.

With these SCTE35 DPI tools, troubleshooting and debugging missing DPI packets becomes relatively simple.

Summary

Troubleshooting video signals in a broadcast or cable facility requires the use of an MPEG Analyzer that provides multiple input signals all running in parallel. The minimum set of test points should be at ingress, ASI/IP switch, and egress. It is critical to measure QoE at both egress and ingest, but in the case of audio and video problems, QoS testing at multiple points in the facility may be needed to pinpoint the source of the impairments. Critical capabilities in an MPEG Analyzer should include:

- Transport stream generation, modification and analysis
 - Real and deferred time
 - Including MPEG-2, H.264 & MPEG-4 AAC
 - Multiplexer
 - Automatic error triggered recording and capture
- A range of physical interfaces
 - Multi-port ASI
 - RF (DVB-S2, 8-VSB & QAM-B)
 - Video over IP analysis and generation (1Gbps and 10Gbps)
- Comprehensive suite of software tools for analyzing all layers of video
 - High accuracy RF layer analysis
 - Transport, Program & Elementary Streams
 - Video & Audio Quality of Experience (QoE)
 - Picture Quality

A facility equipped with an analyzer having the feature set listed above should be able to resolve problems in minutes rather than hours or days using alternative methods.

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