Troubleshooting IP Video Quality of Service

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Introduction

The cost-effective delivery of triple-play services (voice, data, and video) is made possible in large part by next-generation, standards-based, distributed network architectures that use packet transport mechanisms. These new networks must deliver specific class of service (CoS) support for all three service types. In addition, each of these service types requires its own application-specific quality of service (QoS) needs. IP-based delivery of video signals also places unique demands on these networks.

Built to deliver data only, today's data networks meet specific performance levels that are generally based on packet loss parameters. Typical data applications compensate for lost packet events by retransmitting lost data. This retransmission enables error-free performance at the application level. However, voice over IP (VoIP) and IP video services place more stringent requirements on data networks.

New CoS mechanisms enable these networks to achieve acceptable levels of performance to support VoIP and IP video. For example, packet loss rates of up to 4% of the traffic flow and packet jitter approaching 40 ms can be tolerated without negatively impacting the delivery of toll-quality voice. Broadcast video services, on the other hand, cannot tolerate a packet loss rate of more than 0.1% or more than 5-10 ms of packet jitter. In actuality, voice CODECs can hide lost data more effectively than lost video can be masked. In addition, network performance demands of today are much greater than in the past.

When video flows are mixed with data and voice flows, network planning and engineering become even more challenging in terms of meeting the demands for the increase in bandwidth required to transport triple-play services. Complexity increases as new signaling protocol flows, such as Internet group management protocol (IGMP) for broadcast video and real-time streaming protocol (RTSP) for video on demand (VOD) services, are introduced. The dynamic nature of video flows is affected by viewing habits, channel changing loads, and dynamic VOD media requests. All of these factors add to the demands and complexity of delivering the required CoS.

To ensure proper service delivery on these new networks, measurement of the accumulated effects of the IPTV network on critical, application-specific QoS parameters must be made at the customer premises during service installation. Gathering and recording this QoS data is the key to rapid, efficient trouble resolution. The quality of the source material or video content and the quality of the video decoder in the set top box (STB) determine the "potential" quality of the video. The network is a variable, and it can only detract from the design quality. Thus, the focus of the recommended test concept is to measure network-specific variables.

Network-Specific Variables

Network-specific variables affecting the quality of service of a network include content quality, PCR jitter, error indicator, network quality, packet loss, packet jitter, and IGMP latency. It is important to note that this discussion of network-specific variables applies to both MPEG-2 and MPEG-4 compression technologies used in the video payload as long as the ISO/IEC 13818 packet format (described below) is used.

Content quality

The quality of the content is the starting point. Decisions made in the video headend, where the content is acquired, determine the variations in quality. The initial quality of the video stream is established by which content sources are used, which compression algorithms are implemented, which encoders are employed, and what source quality monitoring system is present. The data output of the encoders starts the video packet flow. There are two critical source quality parameters that can be measured in MPEG-2 transport stream video flows at the customer premises and/or in the last-mile access network: program clock reference (PCR) jitter and the video transport packet error indicator count.

Figure 1 shows a schematic diagram of the transport stream packet header. The components of the transport stream packet header are defined in ISO/IEC 13818.

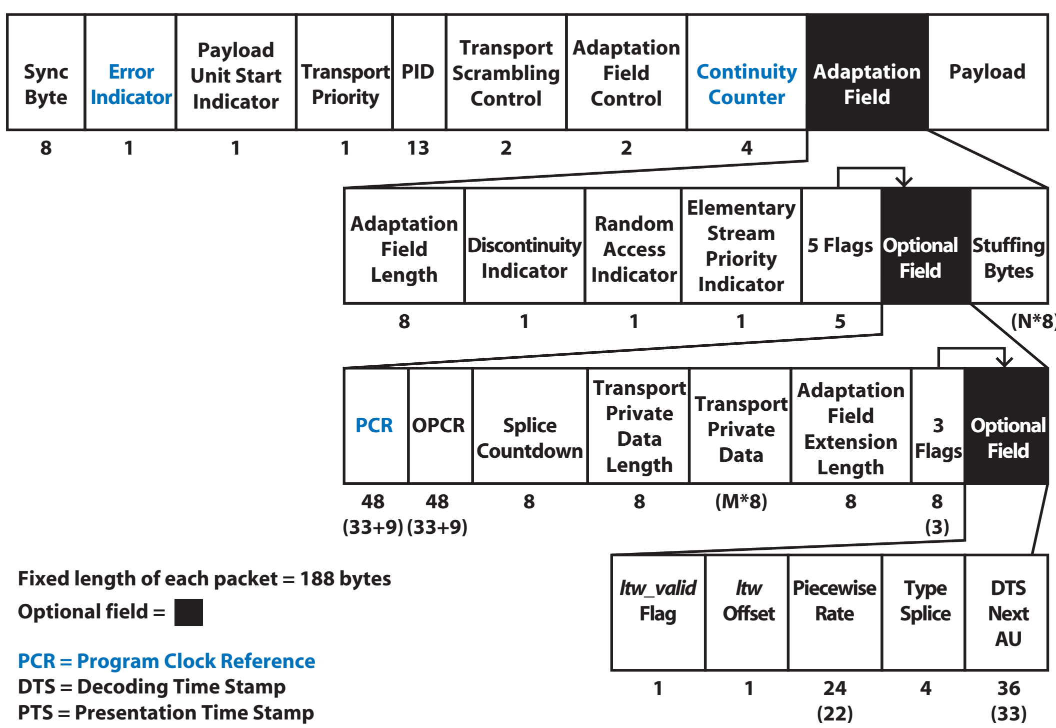


Figure 1. A schematic diagram of the transport stream packet header

PCR jitter

Timing in the transport stream is based on the 27 MHz system time clock (STC) of the encoder. To ensure proper synchronization during the decoding process, the decoder's clock must be locked to the encoder's STC. In order to achieve this lock, the encoder inserts a 27 MHz time stamp into the transport stream for each program. This time stamp is referred to as the program clock reference (PCR). Video decoders use the timing signal to synchronize to the encoded data stream so that they can derive two timing parameters embedded within each audio and video program. These timing parameters, the decode time stamp (DTS) and the presentation time stamp (PTS), are used in the decoding process to properly present the decoded video to the display unit, the television, for example.

If excessive PCR jitter is present, the decoder cannot synchronize itself correctly to the data stream. The end result is visual impairments, such as pixelization, frame freezes, and loss of color. The amount of PCR jitter that is considered excessive is not a constant. It is determined by various parameters, including the input buffer sizes of the decoder and the design of the STB software. However, in today's typical packet video networks, PCR jitter should be less than 10 ms. Preferably, it should be less than 5 ms, depending on the specific decoder/STB design.

Several factors can cause PCR jitter. The mostly likely causes include:

– Overall network packet jitter

– Transcoding problems in the encoder

– Local ad insertion issues

If packet jitter is not excessive when PCR jitter is present, then the cause is specific to the particular program flow. In this example, an encoder may not be performing up to specifications. If this is the case, PCR jitter will be constantly excessive. If PCR jitter is not constant, then a momentary problem from inserting local programming may be the cause.

Error indicator

The error indicator is a bit that is set by the encoders in any transmitted video packet where the encoders detect corrupted source content. The presence of packets with this indication is strictly an issue related to content quality. It is not related to the performance of the distribution network. Monitoring of video encoder output streams in the headend can detect this condition and provide an early opportunity for problem resolution. Error indicator counts seen at the customer premises reveal a source quality problem.

Network quality

Network performance is another determinant of video quality. It can be divided into a few specific parameters:

– Packet loss

– Packet jitter

– IGMP latency

Each of these parameters can be analyzed at the customer premises or in the last-mile access network.

Packet loss

Packet loss is measured by analyzing video packet flows and determining the presence of a continuity error event. Standards define the process. Because each video packet carries a sequence number, continuity errors can be determined with certainty. For example, because an MPEG-2 transport packet is 188 bytes in length, an IP frame carries seven MPEG-2 transport packets within it. Thus, losing one IP frame results in the loss of seven MPEG-2 transport packets. Conversely, if the MPEG-2 continuity counter jumps by seven MPEG-2 transport packets between two consecutive IP frames, one can be fairly certain that an IP frame has been lost.

Missing packets, out of sequence packets, and duplicate packets are all counted as errors. Each of these events can cause decoding errors. Depending on the temporal or spatial components contained in the frames within an MPEG-2 transport packet, a single packet error event may or may not be seen on the TV screen. However, actual network performance is measured by the packet loss parameters regardless of whether or not the decoder can hide the problem.

Packet jitter

If the overall packet flow experiences excess jitter due to congestion problems and resulting CoS mechanism performance issues, packet jitter can be the cause of PCR jitter. If it is excessive enough, packet jitter can cause decode buffers to deplete, which, in turn, causes gaps in the decoder output. Gaps may appear as freeze frame or pixelization events seen on the TV screen.

IGMP latency

IGMP is the signaling protocol used to access broadcast video services that use a multicast network design to efficiently manage network bandwidth. IGMP enables each STB to obtain only the programming that the viewer is interested in watching, conserving bandwidth in the access network.

In this implementation, a join message is sent from the STB to the network. The join message asks the network to send the requested program/channel to the STB by joining a multicast group carrying the desired broadcast channel. IGMP latency is the period between the time the join message is sent and the time the first video packet is received by the STB. Thus, IGMP latency is a measure of both service provisioning and the network's response performance.

These messages travel upstream into the network to the first device that can add (join) the requestor to an existing broadcast channel flow. This parameter measures network performance, but not the end user’s experience, with regard to channel changing time. The IGMP latency plus the time it takes to fill the decode buffer and to decode and display the content is the total user experience time. However, the buffer fill time and the decode time are functions of the network architecture and are not variables. Thus, the measurement of the variable network performance aspect of IGMP latency is the critical parameter for measuring actual network performance.

Figure 2 shows an example of an IGMP message flow when a broadcast program, Channel 2, is requested and then a channel change to another program, Channel 3, is requested.

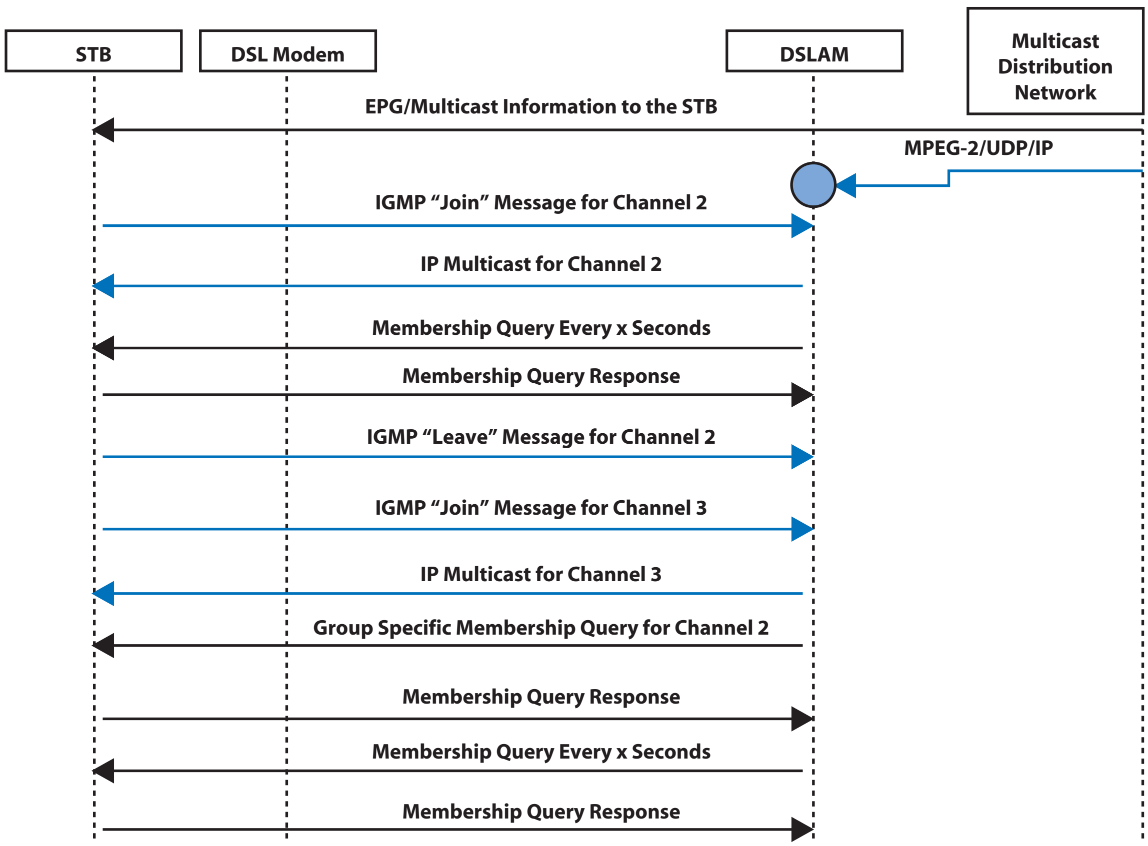


Figure 2. An example of an IGMP message flow

The DSLAM, in this example, has access to the requested programs. It performs a snooping function, looking at the requested program material. If the requested program is directly available at the DSLAM, the join is performed. If the requested program is not available at the DSLAM, the IGMP messages continue upstream to a location, such as a video hub office, where the material is available. The join is performed at this location, and the program is routed to the DSLAM and on to the STB.

The following parameters measure the critical variables in the delivery of broadcast video service.

Network quality parameters:

– Packet loss (continuity error, as a percentage of packet loss)

– Packet jitter (ms)

– IGMP latency (ms)

Content quality parameters:

– PCR jitter (ms)

– Error indicator (count)

In the case of narrowcast services, that is, services tailored for a specific viewer such as VOD, IGMP latency does not apply because RTSP is used. More importantly, the request for access to a stored program does not carry the same latency expectations. If the actual request involves the completion of a payment transaction, then latency becomes even less critical. The network can then easily meet expectations for initial access to the desired program.

Therefore, a simple definition of video QoS used to target network performance includes the following parameters for broadcast video service performance:

– PCR jitter (level)

– IGMP latency (level)

– Continuity error (rate)

– Error indicator (count)

For VOD service, the following parameters define video QoS:

– PCR jitter (level)

– Continuity error (rate)

– Error indicator (count)

Analyzing a single program flow may not reveal the whole picture as it relates to actual service quality. Because some impairments are source specific and some are network performance specific, it is important to take these measurements on more than one stream at a time. Thorough testing requires simultaneous analysis of at least two streams. If the access network bandwidth allows for it, simultaneous analysis of three streams is preferable.

Service Testing and Troubleshooting

As previously discussed, individual program flows should be analyzed to ensure that content quality and network performance parameters are within the desired network specifications. The suggested thresholds for these parameters are listed in Table 1.

Parameter	Threshold
Packet loss	≤ 0.1%
Packet jitter	≤ 5 ms
PCR jitter	≤ 5 ms
Error indicator	zero count
IGMP latency	≤ 250 ms

Table 1. The suggested thresholds for content quality and network performance parameters

Specific STB/decoder designs may be able to tolerate higher levels of jitter based on larger buffer designs. In some cases, jitter levels of 10 ms or higher may be acceptable.

When these measurements are made simultaneously on more than one channel, it is easy to separate content problems from distribution network problems. This is a critical determination for problem resolution.

All Channels vs. One Channel

Packet loss

Packet loss (continuity error) problems are typically seen on all channels/programs coming to the customer premises because they are not source- or content-related problems. If packet loss is present, analysis of the physical layer at the xDSL interface or Ethernet interface will aid in the sectionalization of the problem. If no physical layer errors are present, then packet loss is most likely being caused by the distribution network and not by the access network. In this case, congestion is, more than likely, the issue.

Analyzing further into the temporal component is important. Are packets being lost during known peak traffic times during the day? Are the packet losses coming in bursts with intervals with no loss? Or are they random, single, or small packet loss events? Bursts of loss are symptomatic of buffer overflows related to heavy traffic. Random, single, or small events are more likely caused by noise hits on the access network that are impacting packet flows.

Packet loss may be due to DSL loop performance in the access network that is pushing the bandwidth limits of particular areas where signal-to-noise margins are low and the addition of a second or third channel flow reaches 100% capacity of the loop. In addition, the copper may be poorly balanced, thus allowing high impulse noise to impact data flows. Or, in-home wiring may be introducing noise, damaging data flows.

PCR jitter

PCR jitter problems may be due to content quality problems or overall network packet jitter. The source of the problem can be differentiated by evaluating more than one channel/program at a time. If excessive PCR jitter is present at more than one channel, network jitter is most likely at fault. If excessive PCR jitter is present on only one channel, then a source problem is typically the cause.

Error indicator

Error indicator analysis further reveals content problems. Since the indicator can only be set by the encoder, it specifically reveals content-only problems. Typically, this affects only one program or channel. However, if a multiple program feed in the headend is experiencing problems, more than one program/channel may be affected. In this case, analyzing a channel from another source, or different feed, is suggested.

IGMP latency

IGMP latency measures the network's performance. Typically, IGMP latency is similar for multiple channels. However, if network topology and network management place access to certain program materials deeper in the network, then differences in IGMP latency may be experienced. If such a hierarchical approach is used, then differences may be detected based on which programs are accessible deeper in the network. Testing multiple channels/programs to exercise this network design is useful.

A practical example

Using a JDSU HST-3000, three parameters are measured: PCR Jitter, Latency, and Continuity Error (Cont. Err). Latency is the IGMP latency, which defines the channel changing time not including the decoder buffer file and the decode times. Continuity error is the lost packet rate. These three parameters summarize the video quality of the selected program flow (Figure 3).

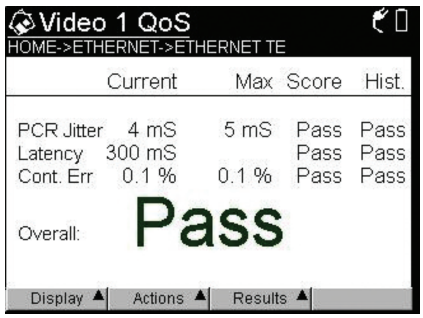
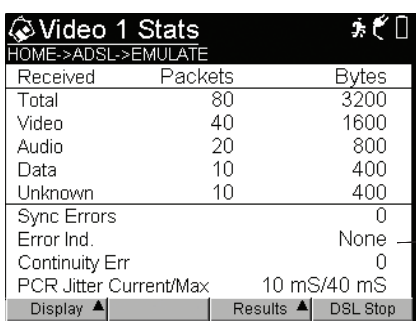


Figure 3. A measure of video quality of service

Additional data can be obtained by analyzing the stream in more detail. Because each stream includes video, audio, and data (program table data) portions, packet statistics can be obtained for each stream (Figure 4). The error indicator count will reveal any content problems.



Video may display poor quality if the **error indicator count** is the only parameter showing a problem. Therefore, any value in the error indicator count other than “none” is bad.

Figure 4. Packet statistics for video, audio, and data portions of the stream

Summary

In summary, analysis of whether or not poor video QoS performance is seen on more than one channel when troubles are reported will effectively initiate the resolution process. Simultaneous analysis of the key QoS parameters on multiple video flows will further refine this analysis and lead to efficient and effective trouble resolution.

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