Application Note

- **Title** **TVQM Video Quality Metrics**
- **Series** Understanding IP Video Performance
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Overview

IPTV, Internet TV, and Video on Demand provide exciting new revenue opportunities for service providers. This Application Note discusses some of the typical issues and problems affecting IP Video service quality and describes Telchemy's Video Quality Metrics (TVQM), which provide key insights into video service performance.

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IPTV Services

IPTV offers exciting new opportunities for service providers to introduce integrated voice, video, and data services over broadband. A number of video service types can be delivered over IP:

- "IPTV" is generally used to refer to a closed Video over IP service with a broad range of content delivered by a service provider. Some definitions of IPTV would suggest that the broadband connection to the home would also be part of this service; however, within the context of this Application Note we regard the two as independent.
- "Internet TV" is generally a more open, Web-like service in which "stations" can broadcast on the Internet. This potentially allows anyone to generate video content and make it publicly available.
- Video on Demand (VOD) is a service that provides access to movies or other video content on demand. This could be part of an IPTV service or could be a service offered independently over the Internet.

In some cases, IPTV services are not seen as directly competing with existing cable or satellite service but may provide some specialization; for example, a VOD service or a service with content in some particular language.

Many different models can be used for delivery of content over IP. Real-time streaming services play video content over IP using either the UDP or TCP protocol and the video stream is decoded and played in near-real time. Other approaches include the download of entire movies or large "chunks" of video to a disk drive or memory.

This Application Note discusses some of the factors affecting the performance of IPTV and Internet TV services that use a streaming model, and describes Telchemy's TVQM video quality metrics, which provide key insight into video service performance.

Factors Affecting the Performance of IPTV

Codec, Bit Rate, and Video Content

Video content is typically encoded and compressed using MPEG-2, MPEG-4 Part 10 /H.264, Microsoft WMV9/VC1, or other codecs. Video codecs typically support a wide range of compression rates, allowing a trade-off between quality and bandwidth.

Much of this compression comes from the use of inter-frame difference encoding—rather than sending every video frame, only the difference between one frame and the previous frame is sent. This works well if there is relatively little change in the image; however, if there is considerable movement across much of the image, then either the bandwidth will increase or quality will reduce. Many video codecs allow either a constant bit rate (in which case quality may vary) or variable bit rate (in which case quality will vary less).

Common video codecs use a combination of intra- and inter-frame coding. For intra-frame encoding, the image frame (I frame) is divided into blocks, a Discrete Cosine Transform is used to convert each block to a set of coefficients, and then variable length coding applied. A group of blocks is combined into a single entity (slice), sometimes carried within a single packet. If a transmission error occurs, then the whole group may be lost, creating a "stripe" within the decoded image. This may occur because, for example, the DC coefficients within each block are predictively encoded from the first block in the slice; an error makes this information unusable for the remainder of the slice. Some errors may damage the frame structure and render the whole frame unusable.

For inter-frame or motion-based coding, motion vectors are determined for each block and encoded. As for intra-frame coding, errors can render a whole slice or frame unusable. In simple inter-frame coding systems, the loss of one frame can make all subsequent frames unusable until the next I frame is received, resulting in a significant period of degraded, frozen, or blank video.

In most cases, the standards for video coding provide considerable flexibility to both encoder and decoder, allowing a range of cost/performance tradeoffs to be made. This can make it difficult to precisely assess the impact of network impairments without knowledge of the exact implementation.

Limited Bandwidth

Bandwidth limitations often occur in the access link, which is typically a DSL or cable connection. If there is insufficient bandwidth for the video stream, then some packets may be discarded in router buffers, leading to quality degradation.

A more subtle problem can occur due to the highly variable rate of packet transmission due to dissimilar sized I, B, and P frames. The peaks in packet transmission rate that occur during I frames can lead to packet loss and hence quality degration.

Packet Loss and Loss Concealment

Packet loss may occur for a variety of reasons, including network congestion, link failure, insufficient link bandwidth, and transmission errors. Packet loss is often bursty, with periods of high loss occurring when network congestion levels are high.

The type of quality degradation that occurs due to packet loss will depend on the protocol being used to carry video:

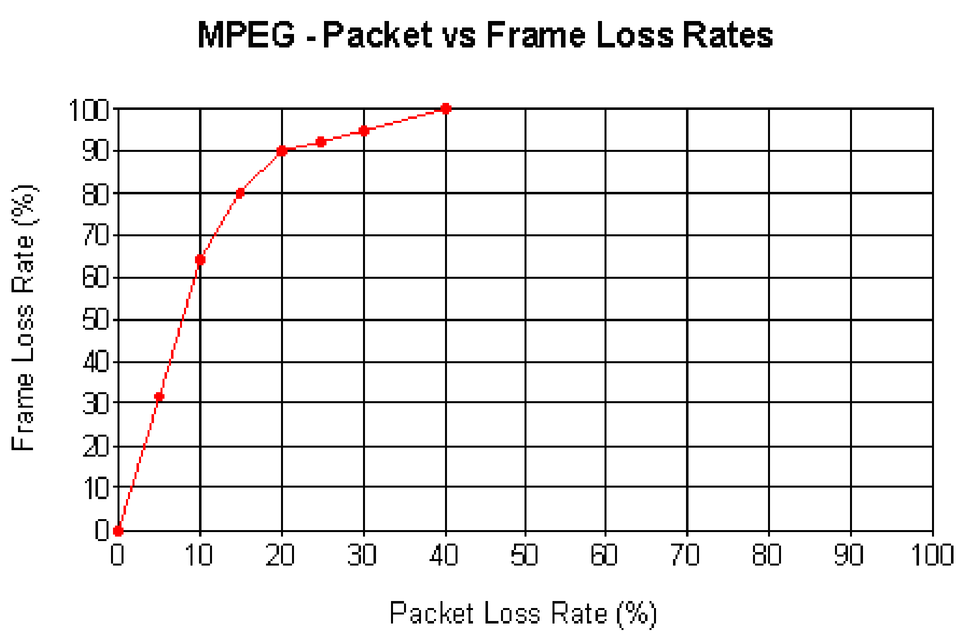
- If UDP is being used, packet loss will directly impact image quality, as some parts of the video stream will be missing.
- If Reliable UDP is used, then lost packets can be retransmitted; however, if retransmitted packets are lost, they will not be "re-retransmitted" and hence image quality will suffer.
- FEC may be used with UDP to replace lost packets; however, if the packet loss rate is too high (e.g., during bursts of loss), then FEC will not be as effective.
- If TCP is being used, then packet loss will lead to retransmission, which can in turn lead to the playout buffer in the set-top box being starved and to pauses in video playback.

For UDP-based video streams, packet loss can cause sections of frames, or complete frames, to be corrupted. As a frame often spans multiple packets, and typical video streams include interpolated frames, a given packet loss rate can result in a frame loss rate six times higher (Figure 1 below).

Server Congestion

Not all problems are due to IP impairments. If servers are not provisioned to support the maximum number of expected users, then server congestion can occur. This would generally lead to pauses in video playback as the playout buffer level is too low.

The use of protocols such as UDP Multicast can help with reducing server loading; however, these rely on a large proportion of the subscribers watching the same content at approximately the same time.



■ *Figure 1. Impact of Packet Loss Rate on MPEG Frame Loss/ Error Rate*

Jitter and Timing Drift

Network jitter is short term variation in packet arrival time, typically due to network congestion. IPTV set-top boxes or playback software typically buffer received video for 5-20 seconds before playout, which in essence means that typical network jitter levels have little effect. Larger delay variations, for example due to server congestion, can cause problems due to playout buffer starvation.

Timing drift occurs when the sending and receiving end clocks are running at slightly different rates. This may require the receiving end to adjust its clock rate in order to avoid occasional problems due to buffer underflow or overflow.

Core Network, Access Network, and Home Network

The IP transmission path typically starts at a video server and ends at a set-top box. This means that packets traverse multiple networks, often owned by different service providers. Core IP networks are often high capacity optical networks that operate well below congestion levels, and hence when problems occur, they are often located within the access network or home network.

Video Impairments

Encoding-related Impairments

Video encoding can introduce a number of different impairments, often related to the bit rate used for encoding and to the characteristics of the video sequence. Common problems include:

- *Blockiness* - typical video codecs process images in small blocks, and quantization of parameter or coefficient values can lead to discontinuities between blocks. This can be exacerbated by the Mach Band effect, in which the human eye tends to sharpen edges.

- *Blur* is a loss in detail that occurs around edges, typically due to the quantization of transform coefficients representing higher frequencies.

Modern video codecs are very effective, but can still suffer from degraded quality when operating at low bit rate where there is considerable movement in large areas of the video sequence.

Transmission-related Impairments

For IP video transmitted over the UDP protocol, packet loss can lead to significant quality degradation. A simple non-robust video stream can be severely degraded with even low levels of packet loss, due to the error propagation effects described above.

Video quality is often represented in terms of PSNR (Peak Signal-to-Noise Ratio), which is a measure of the root mean square (RMS) error between the original and reconstructed video sequences. Generally a PSNR of under 20dB is regarded as unwatchable, and this level is reached for MPEG-2 with a loss rate of under 1 percent.

Error mitigation algorithms are being increasingly applied to help to compensate for packet loss. Methods include:

- *Forward Error Correction* - redundancy is applied to the data stream to allow some proportion of lost or errored packets to be replaced.

- *Interleaving* - in which the video stream is split into alternate frames and each frame encoded separately.

- *Macroblock error concealment* - spatially corresponding macroblocks are copied from the previous frame.

These approaches are used in some modern video codecs and can help considerably with tolerance to packet loss.

IPTV Performance Management

IPTV services can benefit from real-time, non-intrusive performance monitoring. Lightweight agents, such as Telchemy's VQmon®, can be directly integrated into set-top boxes and residential gateways to provide service providers with real-time feedback on service quality, and can potentially provide feedback to servers to enable optimization of the video streaming algorithm.

Impact of Content Scrambling/Encryption

Scrambling or encryption is often used to protect video content (often called Digital Rights Management or DRM). This means that IPTV performance measurement systems cannot decode the picture stream. Telchemy's VQmon has been specifically designed to work with encrypted content, and is able to extract information related to both video content and I/B/P frame loss.

Packet Loss vs. Perceptual Quality Scores

Some service providers attempt to measure video quality using packet loss metrics (such as MDI). If packet loss affects I frames, the resulting errors will extend through the entire Group of Pictures and be very obvious to the viewer. If the same packet loss rate affects B or P frames, the resulting error typically lasts for just one frame (e.g., 1/60th second) and the viewer may not even notice the event. Video performance monitoring algorithms such as VQmon are able to measure the impact of each packet loss event on specific frame types, and hence more accurately measure performance.

TVQM Video Performance Metrics

Telchemy Video Quality Metrics (TVQM) are an extensive set of quality and performance metrics reported by VQmon.

TVQM metrics are grouped into three layers:

- **Perceptual Quality Metrics,** which are high-level Quality of Experience (QoE) metrics that measure the user's experience of the service.

- **Video Stream Metrics,** which relate to the encoded video stream, including image size, frame rate, GoP structure, codec type, etc.

- **Transmission Metrics,** which relate to the performance of the IP network and the UDP/TCP and RTP/MPEG Transport protocols.

TVQM Perceptual Quality Metrics

These metrics provide high-level QoE scores for video, audio, and overall audiovisual quality, allowing immediate visibility into the impact of a wide range of impairments. Telchemy's VQmon technology is unique in its ability to model the impact of transient IP problems on user-perceived quality.

- **MOS-V** - Video MOS, a 1-5 score (5 being best) that considers the effects of the video codec, frame rate, packet loss distribution, and GoP structure on viewing quality.

- **MOS-A** - Audio MOS, a 1-5 score that considers the effects of the audio codec, bit rate, sample rate, and packet loss on viewing quality.

- **MOS-AV** - Audio-Video MOS, a 1-5 score that considers the effects of picture and audio quality and audio/video synchronization on the overall user experience.

- **Burst MOS-V** - picture quality during burst periods where significant degradation is occurring.

- **Gap MOS-V** - picture quality during gap periods where little to no degradation is occurring.

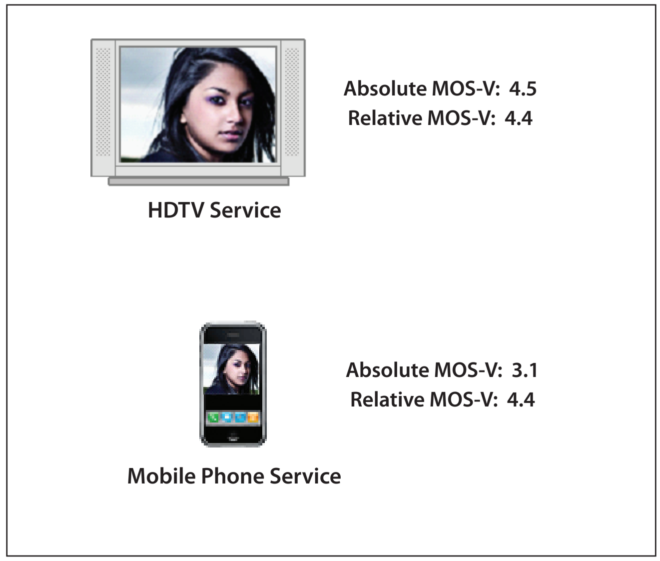
Relying solely on MOS scores may result in confusion when dissimilar types of video service are compared, as viewers tend to form expectations of quality based in part on the perceived capabilities of the medium. For example, a video viewed on a cellular handset might receive a MOS value of 3.1 when little or no quality degradation is evident, while for an HDTV video sequence, a MOS of 3.1 would suggest that there were significant impairments present.

To simplify comparing video quality for different service types, VQmon includes both Absolute and Relative MOS-V in its set of TVQM metrics.

- **Absolute MOS-V** – considers the impact of frame resolution, frame rate, codec and compression level, transmission impairments and frame loss concealment on video quality.

- **Relative MOS-V** – considers all of the factors used to determine Absolute MOS-V *except* the frame resolution, providing a measurement of video quality relative to the ideal for the current video format.

For example, the mobile handset video described above might receive an Absolute MOS-V of 3.1 but a Relative MOS-V of 4.4, indicating that the quality was close to the best possible for that display resolution.



■ ***Figure 2. Absolute and Relative MOS Scores***

In addition to MOS, TVQM perceptual quality metrics include the following:

- **VSTQ** - Video Service Transmission Quality, a 0-50 codec-independent score (50 being best) measuring the ability of the IP network to carry video reliably.

- **Estimated PSNR** - the estimated Peak Signal-to-Noise Ratio expressed in dB. This is an estimate of the distortion that has occurred between the source video stream and the output video stream.

- **Degradation Factors** - percentages of quality degradation due to packet loss, jitter, codec type, audio-video sync, delay, and recency.

TVQM Video Stream Metrics

Video stream metrics provide information on the codec, GoP structure and length, image size, and other factors.

- **Codec Type -** the type of codec used (e.g., MPEG-4).
- **GoP Type -** the Group of Pictures type (e.g., IBBP...)
- **GoP Length -** the number of frames in a Group of Pictures.
- **Image Size -** the image size in pixels (X x Y).
- **Codec Robustness -** parameter that defines the robustness of the decoder frame loss concealment algorithm .
- **Content Sensitivity -** parameter that defines the sensitivity of the video content to visual impairments.
- **Mean Bandwidth** - average video bandwidth excluding IP overhead, FEC, and retransmissions
- **Peak Bandwidth** - peak video bandwidth excluding IP overhead, FEC, and retransmissions.

As mentioned previously, the use of inter-frame video encoding means that packet loss causing errors in I frames will cause longer-lasting (and therefore more noticeable) degradation than the same level of packet loss affecting B or P frames. TVQM's individual I, B, and P frame statistics provide useful data that can be used to troubleshoot problems and help determine the optimum codec, GoP type and length for the video service.

- **I Frame Packets Received, Lost, Discarded -** count of the numbers of received, lost, and discarded I frames.
- **P Frame Packets Received, Lost, Discarded - -** count of the numbers of received, lost, and discarded P frames.
- **B Frame Packets Received, Lost, Discarded -** count of the numbers of received, lost, and discarded B frames.
- **I Frames Impaired -** proportion of I frames impaired by packet loss/discard.
- **P Frames Impaired -**proportion of P frames impaired by packet loss/discard.
- **B Frames Impaired -**proportion of B frames impaired by packet loss/discard.

TVQM Transmission Performance Metrics

The following metrics provide essential data on IP packet loss and discard, including burst and gap statistics that indicate the time distribution of lost and discarded packets. Where appropriate, separate metrics indicate performance both before and after error correction (such as FEC or Reliable UDP) is applied.

- **Packet Loss Rate** - the percentage of IP packets lost in the network.
- **Packet Discard Rate** - the percentage of packets discarded by the jitter buffer due to late arrival.
- **Out of Sequence Packet Rate** - the percentage of packets arriving out of sequence.
- **Duplicate Packet Rate** - the percentage of

duplicate packets.

- **Gap Loss Rate -** the percentage of packets lost during gap periods when little or no quality degradation is occurring.
- **Gap Length -** the average length of intervals between bursts.
- **Burst Loss Rate -** the percentage of packets lost during burst periods when significant degradation is occurring.
- **Burst Length -** the average length of burst periods.
- **Number of Bursts -** the total number of burst periods.

Forward Error Correction (FEC) can replace lost packets, but carries some overhead. FEC metrics provide a measure of the effectiveness of FEC, if it is used, and provide information on the optimum FEC configuration whether or not it is currently in use—allowing service providers to assess whether FEC would be useful.

- **FEC Effectiveness -** the percentage of improvement in the packet loss rate due to the use of Forward Error Correction.
- **Optimum FEC Block Size -** the optimum FEC block size (packets).
- **Optimum FEC Correctable Packets -** the optimum number of correctable packets within a block.

Jitter and delay metrics provide a view of the impact of deliberate packet smoothing/rate shaping and network congestion on overall delay and delay variation.

- **MAPDV -** Mean Absolute Packet Delay Variation (ITU-T G.1020).
- **PPDV -** Packet to Packet Delay Variation (RFC3550).
- **PDV Due to Smoothing -** Packet Delay Variation due to smoothing of CBR streams or due to network moderation of bandwidth peaks.
- **Positive Jitter Threshold -** defined threshold for positive jitter (packets arriving early).
- **Positive Jitter Percentile -** percentage of packets arriving with the positive jitter threshold.
- **Negative Jitter Threshold -** defined threshold for negative jitter (packets arriving late).
- **Negative Jitter Percentile -** percentage of packets arriving within the negative jitter threshold.
- **Media Path Round Trip Delay -** average round-trip packet delay (latency).

Reliable UDP metrics provide insight into the performance of retransmission-based protocols such as Reliable UDP. These protocols improve packet loss rate but increase the variability of bandwidth.

- **Proportion of Packets Retransmitted -** the percentage of retransmitted packets.
- **Ratio of Peak to Mean Bandwidth** - the ratio of bandwidth peak due to retransmission to average bandwidth.

The following TR 101 290 metrics provide information on key error types that occur with MPEG transport protocols, and are useful for identifying and resolving these error conditions.

- **TS_sync_loss** - loss of synchronization at the MPEG transport layer.
- **Sync_byte_error** - invalid MPEG transport sync byte.
- **Continuity_count_error** - incorrect packet order, duplicate packet, or lost packet.
- **Transport_error** - transport error indicator in MPEG transport header set.
- **PCR_error** - discontinuity in program clock reference (PCR).
- **PCR_repetition_error** - time interval between two successive PCR values more than 40ms.
- **PCR_discontinuity_indicator_error** - difference between two consecutive PCR values is over 100ms without discontinuity bit set.
- **PTS_error** - interval between presentation timestamps more than 700ms.

Summary

IPTV allows service providers to increase revenue by offering innovative services and specialized content to their subscribers; however, IP video is highly susceptible to quality degradation from various types of impairments.

Telchemy's TVQM video quality metrics provide a rich set of performance and diagnostic information, allowing service providers to instantly assess subscriber Quality of Experience levels and quickly diagnose problems affecting the delivery of IP video services.

Acronyms

ADSL	Asymmetric Digital Subscriber Line
DC	Discrete Cosine
DCT	Discrete Cosine Transform
DRM	Digital Rights Management
DSL	Digital Subscriber Line
FR	Full Reference
GoP	Group of Pictures
IETF	Internet Engineering Task Force
IP	Internet Protocol
LAN	Local Area Network
MDI	Media Delivery Index
MOS	Mean Opinion Score
MAPDV	Mean Absolute Packet Delay Variation
NTIA	National Telecommunications Information Administration
PCR	Program Clock Reference
PDV	Packet Delay Variation
PPDV	Packet to Packet Delay Variation
PLC	Packet Loss Concealment
PSNR	Peak Signal-to-Noise Ratio
QoS	Quality of Service
RFC	Request for Comments
RMS	Root Mean Square
RTCP XR	RTP Control Protocol Extended Reports
RTP	Real Time Protocol
SLA	Service Level Agreement
TCP	Transmission Control Protocol
TVQM	Telchemy Video Quality Metrics
UDP	User Datagram Protocol
VDSL	Very high speed/bit rate DSL
VOD	Video On Demand
VoIP	Voice Over Internet Protocol
VQM	Video Quality Metric
VSTQ	Video Service Transmission Quality
WLAN	Wireless Local Area Network

References

[1] ATIS-0800008 - QoS Metrics for Linear Broadcast IPTV (August 2007)

[2] ETSI TR 101 290, Digital Video Broadcasting (DVB); Measurement Guidelines for DVB Systems

About Telchemy, Incorporated

Telchemy, Incorporated is the global leader in Voice and Video over IP Performance Management with its **VQmon**® and **SQmon**™ families of call quality monitoring and analysis software. Telchemy is the world's first company to provide voice and video quality management technology that considers the effects of time-varying network impairments and the perceptual effects of time-varying call quality. Founded in 1999, the company has products deployed and in use worldwide and markets its technology through leading networking, test and management product companies. For more information, please visit **www.telchemy.com**.

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