Application Note to SLA Management Handbook

Video over IP / Wireline & Wireless Application Note

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Notice

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Executive Summary

The IP video market has extremely high expectations regarding video service quality and reliability. Telco video service quality will inevitably be compared to with those offered by satellite and cable/MSO providers. If customers expectations are not met, there will be many other providers to choose from, and when customers switch providers, they will most likely switch all their services. Given these considerations, QoS is perhaps the most important issue to address. By providing best practices for monitoring of quality, this Application Note will enable Service Providers to offer high quality IP video services.

This best practice presents the case for real time monitoring of live IP video sessions, and for the use of well known video quality metrics based on International Standards, rather than the simplistic but potentially misleading packet based measures of jitter and packet loss. The advantages and disadvantages of active monitoring techniques (using test sessions) are also considered.

Four types of IP video services or 'scenario' are considered, including various levels of content, wholesale and managed services. Managed services are those where the service provider provides and possibly manages equipments at the customer premises, that may include the residential gateway, the set top box and the home network. Wholesale services are that of network access providers offering connectivity service to content / application service providers. Content services are those offered by content service providers and broadcasters to distribute their content through a service provider. In this case the service provider owns the video head end, where as in the wholesale service, the network access provider merely provides IP connectivity.

Key Performance Indicators (KPIs) are defined as measurements of audio & video quality rather than measures of network performance. Examples of Key Quality Indicators (KQIs) based on these KPIs are contained in this version of the document. Methods of measuring and reporting metrics are also discussed. Measurements of system responsiveness (e.g. channel change) are also considered

The best practice presented draws on approved documents and some work in progress in the ITU, ETSI, ANSI and VQEG, specifically ITU-T Rec. J.144 [8], ITU-T Rec. J.241 [11], ITU-R BS.1387 [7], ETSI TR 102 479 [26], ANSI T1.801.03 [19] and VQEG phase II [16].

The principal recommendations of this document are that:

- Metrics of video quality that are derived from analysis of video signals or estimated from analysis of packet behaviour in real time should be used. KQIs and KPIs should therefore be based on video quality metrics expressed as a MOS rather than e.g. simple measurements of jitter and packet loss.
- Measurements of perceived quality should preferably be made in the STB to be close to the user's real experience. This will require the industry to standardize methods for the non-intrusive measurement of perceptual video quality and support protocols in order to report this information back to the service provider (It is expected that TR-69 [21] / WT-135 [22] and IETF RTCP-XR [32] will eventually evolve to support this requirement).
- The assessment of customer quality of experience should not be limited to video quality but should also include audio quality and system responsiveness.

1. Background / Introduction

1.1. Intended Audience

This Application Note is aimed primarily at Service Providers, with secondary audiences being vendors, content providers, enterprises and end-users.

SLA Management is considered both from the end-user or Enterprise and the Content Provider / Broadcaster perspective, with the objective of identifying what Service Providers should do to ensure Quality as perceived by their Customers and as required by Content Providers. As a result, this document also provides guidance to Vendors, in the capabilities required by Service Providers.

1.2. Application Note Objectives and Scope

This document defines the current best practices for the design, delivery and monitoring of SLAs and OLAs for video services over IP networks.

This document addresses the following topics:

- Defines a standard way of implementing SLAs for Video over IP services, by:
  - Defining the most important KPIs/KQIs
  - Describing a common ways of measurement (active and passive)
  - Defining aggregation methods of test data
  - Describing common SLAs
  - Describing “standard” Video over IP technology models
- Defines classic end-to-end SLA scenarios, where video is either broadcast such as IPTV services or more interactive such as IP Telephony.
- Presents best practice reporting schemas.
- Identifies differences between different Video over IP measurement strategies.
- Provides a framework for the development of future releases or versions of this application note

With the growing number and scale of wireline IPTV deployments, the first release of this document focuses on wireline Video over IP services including Broadcast TV and Video on Demand (VoD). Other services such as Internet TV, Mobile TV, Video Telephony, Video Conferencing, Video Surveillance and Video Download will be considered in subsequent releases of this document.

The main focus of the document is on video picture and audio quality:

- The synchronization between audio and video will be dealt with in a subsequent release of this document when the assessment of the user impact becomes more mature.
- The ability to switch channels in a timely manner is considered in this release. Other service interactivity and responsiveness issues such as the absolute audio & video delay that apply to interactive services like video conferencing are not considered in this release.
- A further aspect that may be considered specific to Video over IP services is security. Although this version does not address security issues, future version(s) of this application note could address security issues
- Other KQIs and KPIs are considered to be normal requirements for video services (examples would be service accessibility, service retainability, service provisioning and billing accuracy, etc.) and definitions exist elsewhere.
- Although service usage metrics are vitally important to the Service Provider, Content Provider & Advertisers alike (e.g. audience, share, customer profiling), these would probably not be used in a SLA.

The following table summarizes the scope of this document and subsequent releases thereof.

	Scope of Release 1	**Scope of Release 2**	**Out of Scope**
Services			
Broadcast TV	X		
Video On Demand (VoD)	X		
Internet TV			X
Mobile TV		X	
Video Telephony		X	
Video Conferencing		X	
Video Surveillance		X	
Video Download			X
KQIs			
Video Picture Quality	X		
Audio Quality	X		
Channel Change Delay	X		
Speech Quality	N.A.	X	
Video Delay	N.A.	X	
Audio Delay	N.A.	X	
Synchronization of Audio & Video		X	
Combined AV Quality		X	
Service Availability			X
Service Retainability			X
Service Usage			X
Security		X	

Note: in this document we are not considering the delivery of analog video on cable or passive optical networks (i.e. non IP-based delivery).

The remainder of this document consists of the following sections:

- Business Challenges
    - Service providers will need to build strong relationships with key content providers in order to deliver attractive & high quality content.
    - Technology overview (section 3 plus annex)
    - Service delivery architecture & components where measurements can be taken
        - Common terminology
        - Network architecture
        - Key assumptions
- Usage -and SLA Scenarios (section 3.2 to 3.4)
    - Broadcast TV (using IP multicast)
    - Video on Demand (IP unicast)
- KPIs/KQIs (section 4)
    - Perceptual metrics – what is used and where
    - What is MOS
    - Scenario - what is the effect of delay, jitter & packet loss.
    - Baseline IP metrics? Where do they fit and the relationship with MOS scoring
- Mapping of KPIs & KQIs to scenarios (section 5)
    - Reports (part of section 5)
- Summary and Conclusions (section 6)
- Annexes
    a. Video and IP metrics technology reference
    b. Terms & definitions
    c. References

2. Description - Business Challenge

2.1. Overview

Video over IP services offer service providers the potential for significant additional revenues from new service types, with most providers currently offering only data and voice based services. In addition current cable service providers have the potential of greatly reducing their operational costs of their network operations by moving to IP based infrastructures. The near inevitable move to IP for all network services and the need for wireline service providers to offer so called "triple-play" services to counter the reduction in data and voice service revenues, means that video over IP is of increasing interest to Service Providers.

For end-users the advent of IP video services also represents a major step forward towards the concept of « whole home entertainment » with more interactivity, more content to choose from and more personalization.

Video services are highly sensitive to network impairments and problems, potentially even more than VoIP services. This means that traditional methods of tracking jitter or packet loss often do not provide enough information to judge the quality of the end-user experience. Moreover even with identical packet loss and jitter profiles two video samples will appear very different based on the content type of each. Content that displays significant movement, such as sport, will appear much lower quality than those with lower levels of movement for example. This means that monitoring just IP or even frame based metrics may not be sufficient to ensure video quality but tests on the waveform/content may be necessary to ensure desired quality.

The added complexity of ensuring both video and audio elements are synchronized can be significant in many cases, but perceptual assessment of this issue is again highly dependent on the content of the service. The scope of this document at present however does not cover this aspect of Video services.

2.2. Customer/ End User Perspective

The customer / end user has well-established expectations of terrestrial Broadcast TV quality as the standard for TV services. Broadcast TV services have a very high availability, few problems and can be depended upon. This is similar for satellite and cable based TV services. Other video based services, such as videoconferences, video telephony, mobile video and newer services, currently have a lower perceived quality expectation from end users.

If IPTV is to attract pay-TV users over from non-IP TV technology there is the need to ensure that quality is as good if not better. This means that SLAs based on quality will be key to build confidence in end users to move or take-up new service offerings.

Other IP video services still represent fledgling markets with the early promises of Video Telephony with cellular video streaming and the wider use of video conferencing over IP still largely left unfulfilled due to the lack of video quality and current relatively high pricing for Video services. If these services are to become a significant part of the Service Providers portfolio then quality must be monitored and maintained and prices reduced.

2.3. Service Provider Perspective

Service Providers need a methodology by which they can both define the level of service that their customers need, and against which they can easily manage the quality of their network, and the level of service required by Content Providers for the distribution of their content. Service Providers may own all, part or none of the service delivery network and as a result may need to establish SLAs with Network Access Providers.
This document gives recommendations on:

- The service level objectives to be met for the satisfaction of their customers
- The service level agreement required with Network Access Providers
- The service level agreement required with Content Providers

The introduction of IP video services requires significant changes to traditional network design, implementation and operation; operational processes have to be adapted or redesigned; new expertise has to be acquired; new business partners have to be managed and in some cases new business models have to be invented/tested.
In particular Service Providers need to manage their relationships with content providers and broadcasters in terms of the quality of the video source and the video delivered to end-users. Service Providers also need to monitor the quality experienced by individual users rather than measuring the overall network grade of service.
Finally Service Providers need guidance for the definition of IP Video SLA management requirements on network vendors, measurement and test vendors and system integrators.

2.4. Vendor Perspective

Equipment vendors need to understand what measurement capabilities are needed in network infrastructure in order to support SLA management. Measurement and test vendors and operational support system vendors need to understand what metrics (KQIs) are required for IP Video. Vendors also need to work with Standardization Bodies in order to establish standardized video quality assessment methods and reporting protocols appropriate for monitoring of SLAs.

2.5. Content Providers

Content Providers need to ensure with Service Providers that their video content is delivered to end-users with adequate quality. The brand image of Content Providers will depend not only on the quality of their content but also on the quality of the service delivered to end-users. On the other hand, Content Providers need to make sure they provide Broadcasters and Service Providers with high quality video content.

3. Descriptions of Scenarios

Service Providers are offering a number of typical Video over IP services. This section captures the essential features of Broadcast TV and Video on Demand services.

3.1. Overview

Wireline IP Video services include two different service types:

- Broadcast TV (typically IP multicast)
- Video On Demand (IP unicast)

The scope of service provided by the Service Provider may be for service to the desktop or Set Top Box (STB) or to the Customer Premises. The principal extent of scope is illustrated below.

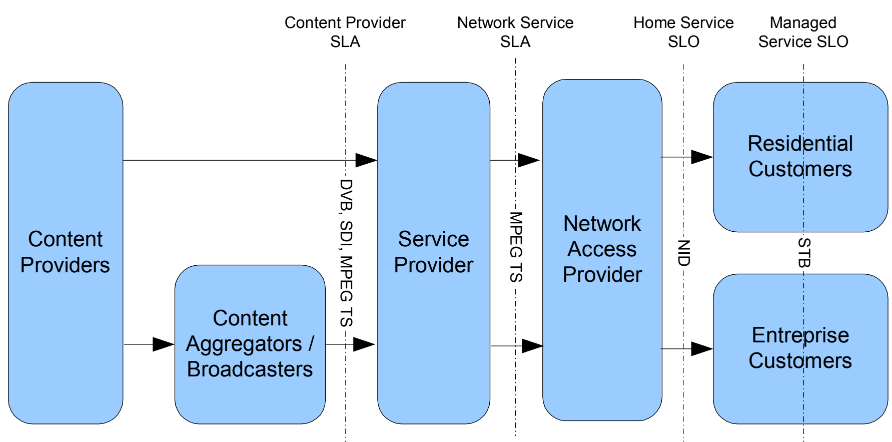


Figure 3-1 IPTV Service Delivery Chain

Figure 3-1 outlines the major relevant scenarios in terms of the business actors involved in the service delivery chain. In essence, IPTV services consist of audio/video media produced by content providers, aggregators or broadcasters and distributed to the Service Provider (typically via satellite & terrestrial/RF communications or analogue/digital interface). The Service Provider in turn delivers this content to residential and enterprise customers over an IP based wireline network that may belong to the Service Provider or a Network Access Provider.

The IPTV service delivery chain illustrates the key service delivery points:

- The service (content) provided by content providers/aggregators/broadcasters to Service Providers. Because the quality of the service delivered to the end-customer depends on the quality of the source content, Service Providers need to establish a SLA with the Content Provider to ensure that the source content has the required level of quality, availability, etc.
- The service delivered by the Service Provider to the end-customer may range from delivering the service up to the network interface device at the customer premises (e.g. residential gateway, integrated access device, DSL modem), provisioning of the Customer Premises Equipments (e.g. RG & STB) without managing the home network, or managing the home network up to the STB or desktop. While end-customers may expect IPTV services to be managed up to the STB/desktop, managing the home network may prove to be a lot more challenging than managing voice and Internet access services, even when the STB uses no wireless connection to the RG.
- Both the Service Provider and the Content Provider will need to establish Service Level Objectives for how their service respectively content is to be delivered to the end-customer, at the customer premises network interface device or up to the set top box.
- When the Service Provider does not fully own the service delivery network, the Service Provider will need to establish a SLA with a Network Access Provider in order to guarantee the service level provided to the end-customer.

3.1.1. IPTV Service Delivery Components

The delivery of IPTV services is based on the following technology components:

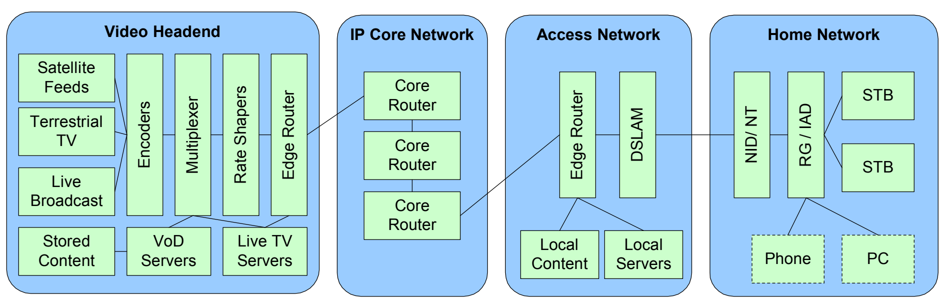


Figure 3-2 IPTV Technology Components

Typical IPTV service delivery components:

1. Video Headend / Video Head Office (Centralized or Distributed VHO)
    a. Media Reception & Digital Turn Around
        i. Satellite (e.g. DVB-S -> SDI)
        ii. Terrestrial TV (e.g. DVB-T UHF/VHF -> SDI)
        iii. Live Broadcast (SDI or analogue)
        iv. Stored Content (ES, MPEG TS)
    b. Encoding, Packetization, Rate Shaping
        i. Real-Time & Offline Encoders (SDI -> MPEG TS)

        ii. Pre-filtering, Transcoding
        iii. Rate Shaping (VBR -> CBR or capped VBR)
    c. Media Acquisition (IPTV Middleware)
        i. VoD acquisition, Broadcast TV acquisition (PIP, EPG, SI, DRM...)
    d. Services (IPTV Middleware)
        i. Electronic Program Guide, Digital Video Recorder, Service Management, Digital Rights Management, etc
    e. Media Delivery (Edge Router)
        i. Streaming, Multicasting / Unicast
2. Core IP network
    a. Provider Edge and Core Routers; handle and route packet streams
    b. Supports both multicast and unicast delivery and QoS
3. Access Network
    a. Broadband Access Server, Aggregation
    b. Access Node (DSLAM or Access Router)
    c. Access Circuit, shared with other services such as Voice & Internet Access
4. Home Network
    a. Network Interface Device, Network Termination, Residential Gateway, Integrated Access Device, CPE Router
    b. Set Top Box, Desktop
    c. Home Networking (Cat5/6 Ethernet, Coax, Phone Line, Power Line, Wireless)

3.2. IP Video Services

3.2.1. Broadcast TV (BTV)

Broadcast TV typically uses IP Multicast and is controlled through IGMP.

Note: Broadcast TV can also be delivered using a mix of multicast and unicast.

IGMP enables content designed for broadcast to be streamed efficiently over the IP infrastructure by duplicating the signal within the network, opposed to producing multiple streams from the video head-end. IGMP is a protocol used between hosts and multicast routers on a single physical network to establish hosts membership in particular multicast groups. Multicast routers use this information, in conjunction with a multicast routing protocol, to support IP multicast forwarding across the Internet. A router should implement the multicast router part of IGMP.

The encoder output is a continuous bit stream called an Elementary Stream (ES). Video programs typically contain multiple ES e.g. one or more video streams, audio streams and service information. Video programs can be packaged as a Single Program Transport Stream (SPTS) or a Multiple Program Transport Stream (MPTS). SPTS is typically used in fixed IPTV services where a single channel per TV is sent to the home due to bandwidth

restrictions whereas MPTS is used in digital cable and satellite where all content is broadcast to each home simultaneously.
Each elementary stream in a program stream is packetized into 18,800 bytes Packet Element Stream (PES) packets. PES packets are then further sliced into Transport Stream (TS) packets of typically 188 or 204 bytes. Typical IPTV deployments use 7 MPEG TS packets per IP packet (in order to avoid fragmentation) and IP over ATM (AAL5) or Ethernet in the access network. Typically MPEG SPTS and MPTS are transported using UDP or RTP/UDP. Other options include RFC2250/Sec.3 or over ATM. Because the delivery of UDP datagrams is not reliable, various techniques can be employed to either retransmit lost packets or use Forward Error Correction schemes (e.g. using RFC2733 RTP payload format).

The key impairments factors on Broadcast TV quality are:

- Coding Impairments
  - Low bit rate coding artifacts
- Network Impairments
  - Network packet loss
  - Packets discarded by the dejittering buffer on the STB due to network jitter
- Decoding Impairments
  - Clock synchronization
  - ....

3.2.2. Video on Demand (VoD)

Video on Demand is typically distributed using a tiered model where the most popular titles are distributed from local head ends or server and less popular content is distributed from higher-level / centralized head ends or servers.

The RTSP protocol allows initiating and controlling the delivery of video streams to IP clients. It does not deliver the streaming content itself but simply supports the interaction between the client and the VoD server such as play, pause and fast-forward.
Video streaming content is typically packaged as an Audio/Video file comprising RTP packets which may then be delivered via TCP. VoD streaming typically involves a certain amount of buffering on the STB that allows for TCP retransmissions in case of packet loss and can accommodate larger jitter than live TV. For VoD, the ability to sustain the delivery throughput at the required video bit rate is therefore critical to avoid the depletion / overflow of this buffer.

The key impairments factors on VOD quality are:

- Coding Impairments
  - Low bit rate coding artifacts
- Network Impairments
  - Ability to sustain required bit rate in order to avoid buffer underflow/overflow.
- Decoding Impairments
  - Clock synchronization
  - ...

3.3. Video Compression and Standards

3.3.1. Video Compression & Transmission

The transmission of audiovisual information requires a number of processing layers that may include all or part of the following:

- Conversion of analog RGB to YUV component video (where Y represents the luminance and U & V for the two-color chrominance)
- Conversion to composite NTSC (525-lines) or PAL (625-lines)
- Digital component video sampling according to ITU-R BT.601
- Digital audio & video compression e.g. MPEG & others
- Digital formatting into MPEG Transport Stream
- Digital transmission formatting into e.g. UDP/IP or Serial Digital Interconnect (ITU-T BT.656)
- Physical transmission formatting e.g. DSL, Fibber, Coax, etc.

Each of the above stage may introduce impairments on the video and audio quality. These are discussed in chapter 4.

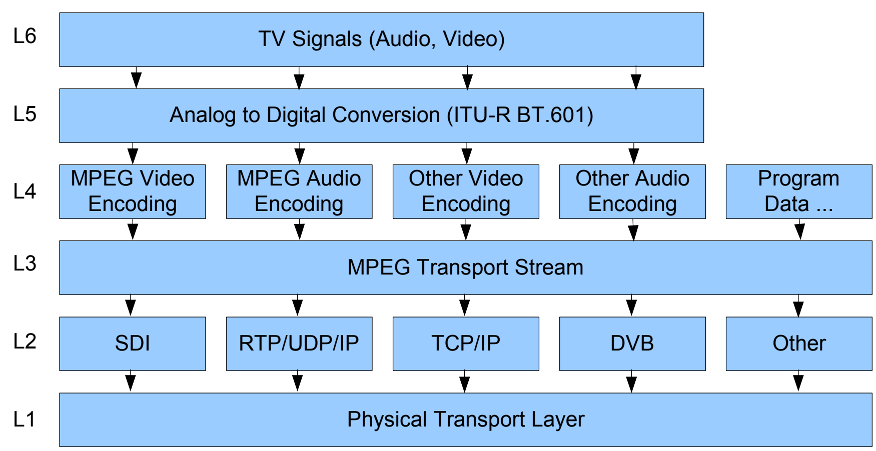


Figure 3-3 Video Processing Layers

3.3.2. IPTV Standards

Video Codecs

- MPEG2 – Most common today in IPTV applications
- MPEG4- Part 2
- MPEG4 AVC also known as MPEG4 Part 10 or H.264
- SMPTE VC1 (previously known as VC-9, the standardized version of WM9)
- Windows Media 9 (WM9) – Windows proprietary
- RealVideo – Real proprietary

Audio Codecs

- MPEG Layer II also known as MPEG-1 Audio Layer 2
- MP3 (MPEG-1 Layer 3)
- Dolby Digital (formerly AC-3)
- AAC – Advanced Audio Coding (MPEG2 AAC or MPEG4 AAC)

- HE-AAC – High Efficiency Advanced Audio Coding (new MPEG4 AAC)

Audio / Video Container File Formats

- AVI – Audio Video Interleave created by Microsoft
- QuickTime – Apple's audio/video container
- MXF – Material Exchange Format developed by ProMPEG for professional video / broadcast applications (includes program data & metadata)
- MPG – Container for MPEG-1 or MPEG-2 video.
- MP4 – ISO MPEG-4 file format.

Packetization

- RTP – Real-time Transport Protocol (RFC 3550) – payload types H.26x and MPEGx (See Appendix for RFC Standards for payload types)
- RDT (Real Data Transport) – Real proprietary
- UDP (User Data Protocol) – offers non-guaranteed datagram delivery, typically used to transport RTP encapsulated media streams.

Signaling

- IGMP (Internet Group Management Protocol) – allows clients to join/leave multicast channels
- RTSP (Real-time Streaming Protocol) – allows remote client to control streaming video with simple commands like play, stop etc.
- SRTP – Secure Real-time Transport Protocol (RFC 3711) encrypts and authenticates RTP & RTCP payloads.
- SCTP – Stream Control Transmission Protocol (RFC 2960) is designed to transport PSTN signaling messages over IP but can also be used for reliable transmission of video over IP.
- DCCP – Datagram Congestion Control Protocol

Network QoS

- RSVP – Resource Reservation Protocol
- MPLS – Multi Protocol Label Switching
- DiffServ / DSCP marking
- IP TOS marking
- Traffic Shaping, Policing...

Monitoring and Reporting

- There is no standard monitoring and reporting mechanism for BTV or VoD services. Several options are under discussion including the DSL Forum TR69 and the STB data model described in WT-135, as well as extending the IETF RTCP-XR to support video quality metrics.

3.4. SLA Interface Points

Service level agreements have to be managed at the following interface points:

- Interface Point 1: at the Video Headend (content reception) Service Providers need to monitor the quality of the content received from Content Providers.
- Interface Point 2: at the network interface between a Service Provider and Network Access Provider (could be anywhere between the VHO and the access circuit) Service Providers need to ensure the quality provided by the Network Access Provider meets the end-user service level objectives

The service level objectives of the Service Provider and possibly the Content Provider can be managed at either:

- Interface Point 3: when the Service Provider manages the quality up to the network interface of the Home Network i.e. when the customer is responsible for the Home Network.
- Interface Point 4: when the Service Provider manages the quality up to the STB or desktop i.e. truly end-to-end.

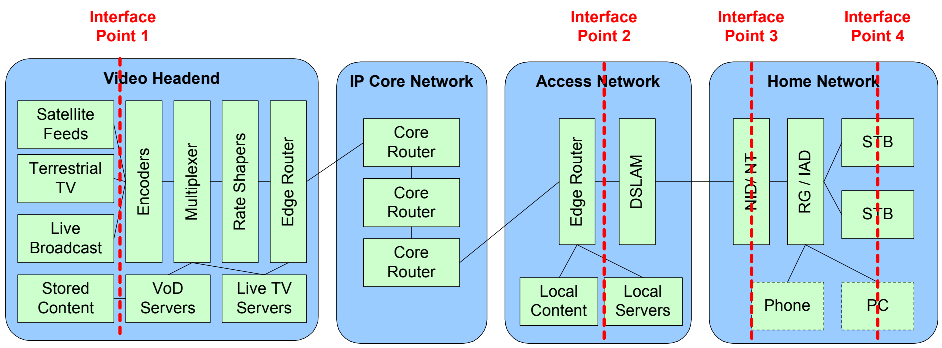


Figure 3-4 SLA Interface Points for IPTV Services

4. Discussion of Quality of Service and KQIs/KPIs

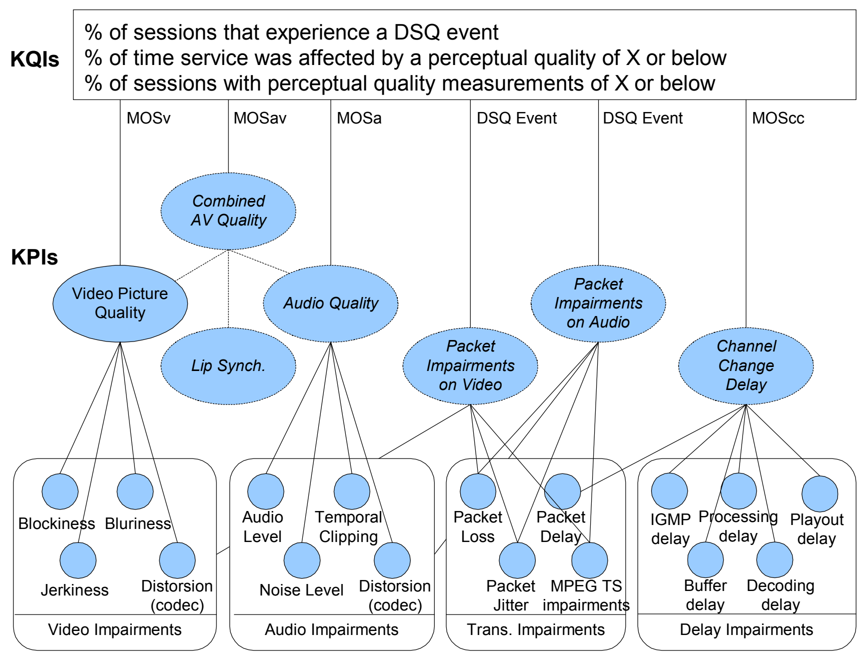


Figure 4-1 KPIs and Measurements

Figure 4-1 shows the types of impairments that can affect IPTV services, how these can be converted to KPIs, and suggests some examples of KQIs for use in SLAs. These topics are further discussed in the remainder of this section.

4.1. Discussion of Measurements and Metrics

This section lists the typical range of problems affecting IPTV performance, measurement techniques and criteria for the selection of appropriate SLA measurement strategies. A full description of these can be found in Appendix C.

The problems affecting audiovisual performance fall into four categories (refer to Figure 3-3 and Figure 4-1):

- Video impairments mainly due digital compression (Layer 4 to 6 on Figure 3-3)
- Audio impairments mainly due signal impairments & digital compression (Layer 4 to 6 on Figure 3-3)
- Transmission impairments mainly due to TS formatting and IP transmission (Layer 2 and 3 on Figure 3-3)
- Delay impairments mainly due to signaling, system processing & transmission (possibly on all layers)

4.1.1. Video Impairments

Analog to digital conversion, digital sampling, digital noise, and digital compression all potentially may create visible degradations to the video content quality. Typical video degradation artifacts include (definitions are taken from ANSI T1.801.02-1995):

a. Blurring – a global distortion over the entire image, characterized by reduced sharpness of edges and spatial details.

b. Block Distortion (Tiling) – Distortion of the image characterized by the appearance of an underlying block encoding structure.

c. Error Block – A form of block distortion where one or more blocks in the image bear no resemblance to the current or previous scene and often contrast greatly with adjacent blocks.

d. Noise – An uncontrolled or unpredicted pattern of intensity fluctuation that is unwanted and does not contribute to the desired quality of a video image.

e. Jerkiness - Motion that was originally smooth and continuous is perceived as a series of distinct snapshots.

f. Temporal Edge Noise - A form of edge busyness characterized by time-varying sharpness (shimmering) to edges of objects.

Measurements of these impairments could be made non-intrusively (or 'passively') i.e. by analyzing the content of live video streams, or intrusively (or 'actively') i.e. by analyzing the content of test video sequences.

4.1.2. Audio Impairments

The same factors as above also may create audible degradations. Typical audio artifacts include (definitions taken from ITU-R BS.1284-1):

a. Quantisation defect - Defects associated with insufficient digital resolution, e.g. granular distortions, non-stationary changes in noise level.

b. Distortion of frequency characteristic - Lack of high or low frequencies, excess of high frequencies as sibilants or hissing, formant effects, comb-filter effects.

c. Distortion of gain characteristics - Change in level (gain) or dynamic range of source signals, level jumps (steps).

d. Periodic modulation effect - Periodic variations of signal amplitude such as warbling, pumping or twitter.

e. Non-periodic modulation effect - Effects associated with transients, e.g. splats or bursts, deformation of transient processes.

f. Non-linear distortion - Harmonic or non-harmonic non-linear distortion, aliasing distortions.

g. Temporal distortion - Pre- and post-echoes, smearing (loss of time-transparency of the source signal), asynchronism of signals or channels.

h. Extra sound (noise) - Spurious sounds not related to the source material, such as clicks, noise, tonal components.

i. Missing sound - Loss of sound components of the source material, e.g. caused by masking failure.

j. Correlations effect (crosstalk) - Linear or non-linear crosstalk between channels, leakage or inter-channel correlation.

k. Distortion of spatial image quality - All aspects including spreading, movement, localization stability, balance, localization accuracy, changes of spaciousness.

In addition to basic audio quality, the impression of surround quality and correlation between sound and picture images are important attributes to be measured.

Measurements of these impairments could be made non-intrusively (or 'passively') i.e. by analyzing live audio signals, or intrusively (or 'actively') i.e. by analyzing the content of audio test signals.

Note: unlike voice over IP, audio signals used for Broadcast TV and Video on Demand are usually well controlled in terms of signal level and background noise level. Additionally echo, which is due to 4-wire to 2-wire conversion (hybrid), is not applicable to IPTV services. Other differences between audio and speech include: higher sampling rate, better amplitude resolution, higher dynamic range, larger variations in power density spectra, differences in human perception, higher listener expectation of quality, and stereo and multi-channel audio signal presentations.

4.1.3. IP Transport Impairments

IP transport impairments include:

a. Packet Loss
b. Delay (or Latency)
c. Jitter (or Packet Delay Variation)
d. Packet Out-Of-Sequence or Duplicated

The impact of packet based impairments needs to be measured in terms of the real time distribution of the impairments within live or test video sessions and the contents of the packet payload. Simply measuring jitter or loss does not accurately reflect the user experience of a service. Unlike voice over IP services, some packet loss is more significant than others, for example if an 'I' Frame or a Program Clock Reference is lost within a packet this has a greater impact on the service than others.

4.1.4. Measurement Algorithms

There are two types of measurement algorithm – packet based or signal based; examples of both are available for use in both active and passive monitoring. Algorithms exist that monitor the effect of the above impairments in different scenarios and circumstances. A full description can be found in Appendix C.

4.2. Measurement of Service Level

4.2.1. Goals of Service Level Measurement

The objective of *service level measurement* should be to verify that service quality meets customer commitments, and to identify the circumstances under which it does not. A set of *service level metrics* are measured and compared to the profile in a *service level agreement* to determine if commitments are being met.

It obviously makes sense for monitoring systems to also capture detailed information about degraded quality events or conditions in order to support corrective action. Within the context of this application note we will define this as *service level management*, i.e., part of the process of delivering service rather than part of the contractual SLA.

4.2.2. Criteria for Determining Suitable Service Level Metrics

4.2.2.1. Network Performance vs. Video Quality Metrics

Basic network performance statistics such as average packet loss and jitter are a very weak indicator of the ability to transport video with adequate quality. Obviously if there are no lost packets and a very low level of jitter then this would imply that there would be no quality degradation due to packet loss or jitter, however this relationship becomes weak as soon as there is even a low level of packet loss or network congestion.

The effects of packet loss and discard on audio/video stream quality will depend on the distribution of loss and discard (i.e. what is the density of loss/ discard during degraded periods), the packet payload contents (frame type, service information), the codec used, the video bit rate, the frame rate, the packet loss concealment algorithm, etc. For example a program stream may have 2% packet loss / discard and may seem acceptable whereas another may have 1% packet loss / discard and appear degraded.

Stream quality metrics combine these various factors and are hence able to achieve a much higher correlation with subjective opinion than simple packet metrics.

4.2.2.2. Aggregate Quality Metrics

Aggregation of metrics is essential, in order to distil a large quantity of data to a sensible level. However, video is very sensitive to impairments, and unlike email or web applications, subscribers to IP Video services are very aware when problems occur, hence averaging statistics can mask problems that affect user opinions.

Aggregate metrics such as the average MOS score per month are essentially meaningless. Problems on IP video call should not be masked by duration. For example the duration of an IPTV session is likely to be in hours, whereas IP video telephony is in minutes. The users' perception of each service will be affected by the number of quality issues over time. It is therefore preferable to define aggregate metrics that relate to:

- *The probability of a video session being degraded*
- *The severity and duration of degradations relative to the total session*

4.2.2.3. Active versus Passive Measurements

The performance of IP Video services may be measured using active or passive testing. These are to some extent complementary techniques.

Active testing systems measure the quality of test video sessions.

Characteristics

Can be used for pre-deployment or off-peak testing where no live customer traffic exists
Creates additional network traffic/loading
Correlates highly with end users' perception of quality but not based on live session.

Pseudo active testing on broadcast services where known content can be used as a reference.

Passive testing systems measure the quality of live sessions. In Video over IP scenarios there are two possibilities for passive testing, those based on IP bearer / MPEG TS characteristics and those that address the signal payload.

Characteristics of both passive techniques
Does not generate any additional network traffic
Can't be used when there is no traffic

Passive IP Bearer / MPEG TS characteristics
Provides an accurate repeatable estimate of end-user experience
Can't reflect signal-based impairments (assumes high quality of encoded video)
Lower computational complexity than active or payload signal testing

Passive Payload Assessment (Signals Based)
Reflects IP impairments and signal based impairments
Computationally more complex than IP bearer measurements
Requires real-time decoder and fast processor for real-time computation of quality prediction.

It should be noted that a hybrid active testing scenario exists where (active) test sessions are used to generate traffic and then passive test systems can be used to make the assessment.

4.2.2.4. Service Level Metrics related to Transient Quality Problems

Some IP quality impairments are transient, notably packet loss and jitter. Studies for VoIP, but also applicable for Video over IP, have shown that packet loss, and packet discards due to jitter, tend to occur in sparse bursts several seconds in length with a 20-30% loss/discard rate. This results in brief periods of degraded quality, or transient quality problems.

Given that transient quality problems occur then it is reasonable to construct metrics related to these:

- Probability of a session (or percentage of sessions) experiencing one or more transient problems
- Probability of a session experiencing N transient problems
- Severity of transient problems

These types of transient quality problems are sometimes termed "*service quality affecting events*" or "*degraded service quality events*".

4.2.2.5. Midstream versus Endpoint Measurement

The impact of IP impairments on the video quality experienced by the user can be determined by making measurements directly in the endpoint (e.g. STB, PC) or at some (midstream) demarcation point along the packet flow.

In the endpoint, access to the IP stream can be gained after the jitter buffer and packet re-sequencing process, thus removing the requirement to account for the effects of the jitter buffer. However, packet loss concealment algorithms will still be specific to the endpoint and should be accounted for when making assessments of delivered audiovisual quality to the end user.

A midstream measurement system generally determines a quality measurement by estimating what would happen if an endpoint were connected at that midstream location.

Endpoint measurement

- *Provides a better guide to the actual user experience*
- *Allows the effect of the home network, RG and STB behavior to be accounted for*

Midstream measurement with endpoint emulation

- *Provides a more general measurement of network performance*
- *Does not take into account degradations occurring in the access circuit & the home network (assuming the midstream measurement will most likely be placed on the aggregation network)*
- *Measurements may differ from actual user experience*

The choice of which technique to use will be determined by the availability of appropriate RG/STB measurements and associated management protocols versus the feasibility and accuracy of emulating the remaining access path between the midstream point and the endpoint.

4.3. Key Quality Indicators

Figure 4.1 shows the mapping of impairments to metrics and then suggests suitable KPIs and KQIs based on these metrics:

- KPIs to be used for KQIs
    - Video Picture Quality - MOSv (ITU-T J.144, ITU-R BT.1683)
    - Audio Quality - MOSa (ITU-R BS.1387)
    - Lip Synchronization (ITU-R BT.1359)
    - Combined Audio & Video Quality – MOSav
    - Channel Change Delay – MOScc, Tcc
    - Packet Impairments on Video – Rate/duration/severity of DSQ events
    - Packet Impairments on Audio – Rate/duration/severity of DSQ events
- KQIs
    - % of sessions experiencing at least one DSQ event
    - % of sessions experiencing a rate of DSQ events > N / hour
    - % of sessions experiencing a DSQ event of > T seconds / hour
    - % of session time with audio DSQ < X (on MOS scale)
    - % of session time with video DSQ < X (on MOS scale)
    - % of session time with channel change DSQ < X (on MOS scale)
    - % of session time with combined audio/video DSQ < X (on MOS scale)
    - % of sessions with non-synched audio/video of > T seconds / hour

Note: DSQ means Degraded Service Quality. A DSQ event is defined as "a noticeable impairment of the audio quality, video quality or service response time" (see Appendix C for examples of impairments).

5. Description of Best Practices

5.1. Overview

This section explains how the metrics and measurement methodologies outlined in Section 4 can be applied to the Scenarios described in Section 3.

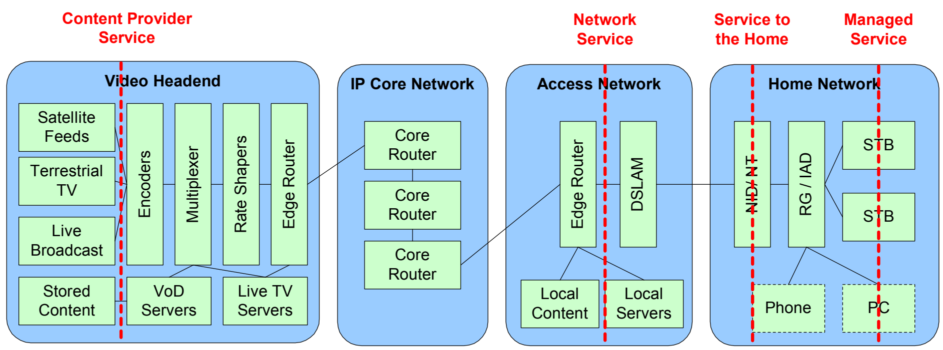


Figure 5-1 IPTV Applications - Scope of SLA

5.1.1. Key recommendations of best practice

The key recommendation of best practice in all cases is that the key quality indicators should be based on actual user experience, and therefore should measure the quality as perceived by end-users.

The techniques already discussed provide MOS and DSQ metrics based on test or live session measurements, and either assess the actual audio/video signal, or provide an estimate of audio/video quality based on the real time behavior of packet or frame based impairments introduced by latency, jitter, packet loss, etc.

The second important recommendation is that key quality indicators should be based on the objective end-to-end measurement of the values of a small number of parameters on the delivered IP stream at the customer premises equipment and relayed back to the head end. Measurements and monitoring taken at the STB are the one closest to the user's real experience of the service. The STB should have the capability to give both IP layer measurements and video stream measurements

5.1.2. Discussion

5.1.2.1. Location of monitoring function

The monitoring function should be located as close to the edge of the service demarcation between service provider and customer, to give the best possible estimate of customer opinion. However this may not always be practical or even available.

The location of the monitoring function is therefore very dependent on the scope of each scenario.

While some service providers want to control the gateway, the home network and all of the attached devices, others see potential dangers becoming involved in this complex set of activities. But in all cases, the customer expectation may likely be that the service provider will manage the service till the STB or the PC, and service providers may eventually be in a situation where they have to handle the home networking issues.

The ITU-T Rec. J.241 provides guidance on measurements methods, metrics and location for digital video services over broadband IP networks. The relevant recommendations for SLA management can be summarized as follows:

- The determination of service level should be based on end-to-end measurement which should provide information on the quality offer the user and the influence of the IP network on the video signal
- The STB should be used as measurement end-point to estimate the video quality offered to the end-user of the service. This assessment can be based on: the video frame rate, buffer underflows, buffer overflows, and coding specific parameters. A measure of the frame rate at the output of the decoder gives a rough estimate of the continuity of the service.
- Measurements and quality parameters at the IP layer make it possible to define reference values for network requirements that are agnostic of the underlying transmission technology and are suitable for use in end-to-end quality assessment. These include: packet loss ratio, latency, jitter and their time distribution.

Note: Using STB as a measurement end-point raises concerns when the STB is not under the control of the service provider and may be affected by home network issues.

5.1.2.2. Type of measurement

It is recommended that passive/non-intrusive monitoring be used to observe the behavior of all live sessions. In practice this means that measurements of IP and MPEG TS impairments are the principal type of measurement used. Where possible, endpoint measurements should be used, to ensure estimates of audio & video quality are closely aligned with customer's quality of experience. This however is dependent on the scenario being managed.

In many scenarios, it is also useful to use active signal based measurements, to provide a more accurate and controlled reference for all quality scores; these will also provide a good means of supporting troubleshooting.

5.1.2.3. Suitable metrics and aggregate metrics (KPIs)

It is recommended that the Common IPTV Metrics (Appendix C) be used as a minimum data set for per-session data collection.

In all scenarios, aggregate metrics for passive IP & MPEG TS impairment measurements should include one or more of the following:

(i) Degraded Service Quality Events (DSQ Event)
Defined as buffer holes/discards (due to packet loss, late arrival or overflow) leading to noticeable visual or audio disturbances
Applies to active/passive packet based measurements

(ii) Session Quality Metrics
MOS audio
MOS video
Channel change delay
Play delay
DVR delay

Other metrics in the Common IPTV metrics table are principally used for service level management rather than for SLA definition.

5.1.2.4. KQIs

The detailed definition of KQIs is a matter for service provider marketing policy; however the following is a good guide to best practice.

For each scenario, the KQIs can be one or more of the following:

(i) Degraded Service Quality Events (DSQ Event)
Percentage of sessions experiencing one or more DSQ Event
Percentage of sessions experiencing multiple DSQ Events
Mean rate of DSQ Events
Percentage duration of DSQ Events

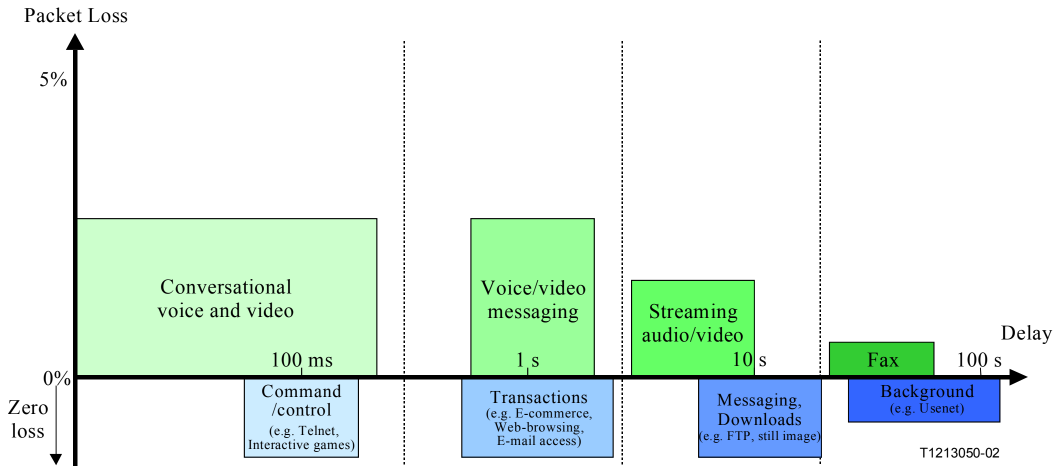
(ii) Session Quality Metrics
Percentage of sessions with video quality less than X
Percentage of sessions with audio quality less than X
Percentage of sessions with channel change delay more than X
Percentage duration with video quality less than X
Percentage duration with audio quality less than X

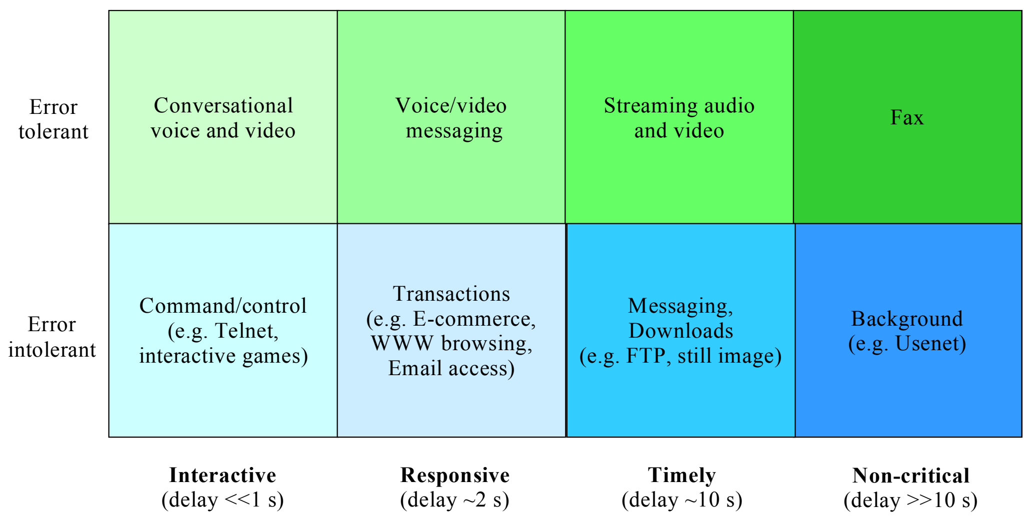
5.1.2.5. OSS considerations

Metrics can be fed back from monitoring points using standard protocols such as the DSL Forum TR-069 for WAN-side CPE management. In particular the DSL Forum is defining object models for specific services such as the TR-104 for VOIP and the WT-135 for managing STBs, which is currently in the form of a working text. TR-069 is becoming a key standard for service providers and residential gateway and set-top box vendors.
In addition, most vendors support proprietary protocols or vendor specific extensions to offer additional management capabilities.

5.1.2.6. Performance Guidelines

The ITU-T Rec. G.1010 provides guidelines of suitable performance targets for a range of multimedia applications classified in eight categories based on tolerance to information loss and delay. The relevant targets for IP Video are reproduced below:





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Further information can be found in ETSI ETR 102 479, DSL Forum WT-126 and ITU-T Rec. J.241.

Note however that SLAs should be based on measures of the perceived audio quality, video quality and responsiveness rather than underlying IP performance. These targets should therefore be used rather as engineering guidelines than service level objectives.

5.2. Content Provider Service

This scenario addresses the delivery of video content from Content Providers, Aggregators, Advertisers or Broadcasters (collectively referred to as Content Providers in the following) to Service Providers.
In this scenario Service Providers need to ensure the quality of the content received from Content Providers (SLA from the Content Provider to the Service Provider) and conversely Content Providers need to ensure that their content will be delivered to the end-user with satisfactory quality. The delivery of content to end-users will be dealt in the subsequent scenarios and therefore this section addresses the SLA offered by the Content Provider to Service Provider regarding the quality of the content offered for subsequent distribution.

In this scenario, measurements are performed in the video head-end either on the base-band signal directly (e.g. RF, SDI, ASI or analogue) typically using a combination of visual monitoring through video wall displays and video signal analyzers, or on the MPEG stream using a combination of visual inspection through decoder and mosaic displays or MPEG analyzers/monitors.
In addition, because all measurements are centralized in the head-end, objective video quality assessment techniques in presence of full reference or partial reference can be used in this scenario.

5.3. Network Service

This scenario addresses the case where a service provider is delivering content to end-users through a third-party network access provider.

In this scenario the service provider needs to ensure the quality of the content handed-over to the network access provider according to the transport rules appropriate to the access network in terms of: maximum packet rate per stream, maximum number of sustainable streams, transport protocol to be used, frame size, packet size, allowed inter-packet gap profile, maximum burst size, etc. Typically the content will be delivered to the access network as MPEG transport stream that can be measured at the POP location using IP layer and MPEG stream probes/analyzers.

The service provider also needs to ensure the quality of the content delivered by the network access provider to the end-user. To this end the network access provider would have to measure the quality delivered to end-users and handover this information to the service provider. This implies that the network access provider has to offers an IPTV transport service (and associated SLA) rather than a mere broadband IP transport service.
Consequently, the service level agreement from the network access provider to the service provider would then be either an IPTV service SLA or a backbone IP transport service SLA. The first case is described in the following sections and the later would be based on IP packet measurements between the POP and the CPE.

5.4. Service to the Home

This scenario consists in delivering IP video services to the CPE (i.e. between the NID and RG). In this case the service provider makes no commitment on the in-home delivery but has to ensure the quality on the access circuit. In order to reflect the impact of the access circuit, the NID/RG should be capable of providing IP packet and video stream measurements allowing to estimate the video quality that would be offered to the end user in the absence of impairments due to the home network and associated CPE. Although this estimation may not necessarily reflect the true user opinion, it can be used as a reference quality for comparison between different customers, independent of the home network and CPE behavior.

In this scenario the types of measurements to be used essentially include IP packet and possibly MPEG TS measurements for in-service/passive monitoring, measurements of video test signals and measurements of response times using test channels.

5.5. Managed Service

In the managed service scenario, service providers assume control of the complete path between the head-end and the STB, including the access circuit, customer premises equipment and home network. This means that the service provider assumes full responsibility for the end-user experience of the service, which ultimately is what the end-user has been accustomed to expect from traditional video broadcasting services.

In this scenario, measurements have to be performed on the STB in order to accurately reflect the end-user’s opinion and take into account the last mile and in-home issues, which cause the majority of the video quality impairments.

The key measurements to be provided by the STB are identified in ITU-T G.1020 & J.241 and DSL Forum WT-126 & WT-135. This may include:

- Video frame rate measurement at the output of the decoder.
- Buffer underflows, buffer overflows, and coding specific parameters.
- Command and control measurements: channel change delay, DVR command delay, STB boot delay
- Failure measurements: channel changes, DVR
- IP layer measurements: packet loss ratio, latency, jitter and their time distribution.

It is recommended that STB measurement should be reported back to the head-end via standardized protocols such as DSL Forum TR-69.

5.6. Summary of Recommendations

The tables below summarize the key recommendations of best practice for Service Providers and Vendors.

Key Recommendations to Service Providers
1. SLA should be based on indicators of the actual user experience
2. SLA should include not only video quality but audio and responsiveness as well
3. Measurements should be made at the service demarcation point, preferably the STB
4. KQIs should be based on perceptual quality rather than network performance metrics
5. KQIs should take into account the time-varying nature of quality degradations rather than averages (in particular when considering the duration of video sessions).
6. KQIs should be based on standards (definition, measurement method, quality assessment model).

Key Recommendations to Vendors
1. STB should support non-intrusive packet or frame based measurements for the assessment of audio and video quality.
2. STB & RG should support standard reporting protocols and data models such as TR-69 / WT-135 and/or RTCP-XR.
3. Measurements and reporting techniques should be scalable to allow the real time monitoring of all and every single customer.
4. Vendors should develop and standardize objective perceptual audio and video quality assessment techniques in the absence of reference in order to offer standardized and comparable KQI measurements.

6. Conclusion, Summary and Next Steps

The Best Practice for IP video SLAs makes use of KQIs that truly represent the User's experience of the service. For video quality, this means that either perceptual measurements of live or test video signals are used, or that metrics that estimate video quality based on observation in real-time of factors affecting video quality are used. In both cases these measurements should be made as close as possible to the STB in order to reflect the user perception.

Best practice KQIs therefore provide MOS metrics based on international standards, including ITU-T Rec. J.144 [8], ITU-T Rec. J.241 [11], ITU-R BS.1387 [7], ETSI TR 102 479 [26], ANSI T1.801.03 [19] and VQEG phase II [16].

These standards however are essentially based on the analysis of test video signals using sophisticated signal processing techniques that require real-time decoders and fast processors for real-time computation of quality predictions.

Although there is no standardized solution today for mass deployment, the current industry practice is to use packet based monitoring systems.

Future releases of this Application Note will include results from industry initiatives to address the standardization issues, in particular:

- Non-intrusive packet based or signal-based models for video quality measurement currently being developed/evaluated by ATIS, VQEG and the ITU-T IPTV focus group, etc.
- Video quality reporting mechanisms based on future versions of the DSL Forum TR-69 / WT-135, the IETF RTCP-XR [32], etc.

In addition, the topics identified in section 1.2 will be addressed in the next release of this Application Note.

Appendix A: Terms and Abbreviations

Terminology

Term	Definition
Burst	A series of packet loss and/or discard events occurring in close proximity.
DSQ Event	Any service quality affecting event occurring during a session i.e. a noticeable audio or video impairment or service response delay.
Edge Device	A device on the edge of a customer or service provider network
Enterprise	Any corporate body – in this case the customer of a service provider
Full Reference	A perceptual quality assessment method using a full copy of the original audio or video signal.
Gap	The period between bursts
Jitter	Packet delay variation
No Reference	A perceptual quality assessment method making no reference to the original audio or video signal.
Non Intrusive	A perceptual quality assessment method applicable to live audio and video signal.
Packet Discard Rate	The rate at which a jitter buffer discards late packets
Reduced Reference	A perceptual quality assessment method using only a reduced reference (e.g. features) of the original signal.
Service Provider	The provider of IP Video services
Vendor	Supplier of IP Video equipment

Abbreviations and Acronyms

ASI	Asynchronous Serial Interface
CBR	Constant Bit Rate
CPE	Customer Premise Equipment
DSL	Digital Subscriber Line
DSQ	Degraded Service Quality
DVR	Digital Video Recorder
ES	Elementary Stream
ETSI	European Telecommunications Standards Institute
FEC	Forward Error Correction
IETF	Internet Engineering Task Force
IGMP	Internet Group Management Protocol
IP	Internet Protocol
IPTV	IP Television
ITS	Institute for Telecommunication Sciences
ITU	International Telecommunications Union
KPI	Key Performance Indicator
KQI	Key Quality Indicator
LAN	Local Area Network
MOS	Mean Opinion Score
MPEG	Motion Picture Expert Group
MPTS	Multi Program Transport Stream
NAT	Network Address Translation
NTIA	National Telecommunications and Information Administration
OLA	Operational Level Agreement
OOS	Out of Sequence
OSS	Operations Support System
PCR	Program Clock Reference
PEAQ	Perceptual Evaluation of Audio Quality
PES	Packet Elementary Stream
PLC	Packet Loss Control
PPDV	Packet-to-Packet Delay Variation
QoS	Quality of Service
RFC	Request For Comment (IETF)
RG	Residential Gateway (also called Home Gateway)
RTCP	RTP Control Protocol
RTP	Real Time Protocol
SDI	Serial Digital Interface
SLA	Service Level Agreement
SLO	Service Level Object
SP	Service Provider
SPTS	Single Program Transport Stream
STB	Set Top Box
TR	Technical Report (ETSI, DSL Forum recommendation)
TS	Transport Stream
UDP	User Datagram Protocol
VBR	Variable Bit Rate
VOD	Video On Demand
VoIP	Voice over IP
WAN	Wide Area Network
WT	Working Text (DSL Forum draft)

Appendix B: References

References

Reference	Description
[1]	GB917 SLA Management Handbook, TMF, 2005
[2]	ITU-R BT.500-11 Methodology for the subjective assessment of the quality of television pictures
[3]	ITU-R BT.1683 Objective perceptual video quality measurement techniques for standard definition digital broadcast television in the presence of a full reference
[4]	ITU-R BT.1359-1 Relative timing of sound and vision for broadcasting
[5]	ITU-R BS.1284-1 General methods for the subjective assessment of sound quality
[6]	ITU-R BS.1286 Methods for the subjective assessment of audio systems with accompanying picture
[7]	ITU-R BS.1387-1 Method for objective measurements of perceived audio quality
[8]	ITU-T J.144 (03/2004) Objective perceptual video quality measurement techniques for digital cable television in the presence of a full reference
[9]	ITU-T J.147 (07/2002) Objective picture quality measurement method by use of in-service test signals
[10]	ITU-T J.148 (05/2003) Requirements for an objective perceptual multimedia quality model
[11]	ITU-T J.241 (04/2005) Quality of Service ranking and measurement methods for digital video services delivered over broadband IP Networks
[12]	ITU-T Y.1540 (12/2002) IP packet transfer and availability performance parameters
[13]	ITU-T G.107 (see also ITU-T G.108, G.113) The E-model, a computational model for use in transmission planning.
[14]	ITU-T G.114 One-way transmission time.
[15]	ITU-T G.1010 (11/2001) (see also G.1020) End-user multimedia QoS categories
[16]	VQEG August 25, 2003 (see also ITU-R BT.1683 & ITU-T J.144) Final report from the VQEG on the validation of objective models of video quality assessment phase II.
[17]	ANSI T1.801.01-1995 Digital Transport of Video Teleconferencing/Video Telephony Signals - Video Test Scenes for Subjective and Objective Performance Assessment.
[18]	ANSI T1.801.02-1996 Digital Transport of Video Teleconferencing/Video Telephony Signals - Performance Terms,

	Definitions, and Examples.
[19]	ANSI T1.801.03 Digital Transport of One-Way Video Telephony Signals - Parameters for Objective Performance Assessment
[20]	ANSI T1.801.04-1997 (see also ITU-T P-931) Multimedia Communications Delay, Synchronization, and Frame Rate Measurement.
[21]	DSL Forum TR-069 May 2004 CPE WAN Management Protocol
[22]	DSL Forum WT-135 Data model for a TR-069 enabled STB
[23]	DSL Forum WT-126 Triple Play Services Quality of Experience (QoE) Requirements and Mechanisms
[24]	ETSI TS 101 329-5 (V1.1.1) (TIPHON) Technology Compliance Specification; Part 5: Quality of Service (QoS) measurement methodologies.
[25]	ETSI TR 102 356 (V1.1.1) (STQ) Application and enhancements of the E-Model (ETR 250); Overview of available documentation and ongoing work.
[26]	ETSI TR 102 479 V1.1.1 (TISPAN); Review of available material on QoS requirements of Multimedia Services
[27]	ETSI TR 250 (TM) Speech communication quality from mouth to ear for 3,1 kHz handset telephony across networks
[28]	ETSI TR 290 (DVB) Measurement guidelines for DVB systems
[29]	ISO/IEC 11172 (MPEG 1, 5 parts) Information technology - Coding of moving pictures and associated audio for digital storage media at up to about 1,5 Mbit/s.
[30]	ISO/IEC 13818 (MPEG 2, 11 parts) Information technology - Generic coding of moving pictures and associated audio information.
[31]	ISO/IEC 14496 (MPEG 4; currently in 16 parts) Information technology - Coding of audiovisual objects.
[32]	IETF RFC3611 RTP Control Protocol Extended Reports (RTCP XR)

Source or Use

Sources of technical information have been provided where relevant within the body of this application note.

IPR Releases and Patent Disclosures

This Application Note makes reference to international standards and recommendations, including:

- ITU-T Rec. J.144 [8]
- ITU-T Rec. J.241 [11]
- ITU-R BS.1387 [7]
- ETSI TR 102 479 [26]
- ANSI T1.801.03 [19]
- VQEG phase II [16]

In order to implement this Application Note, it is not necessary to implement any or all of these recommendations in full. However, a best practice solution would use one or more of these standards and recommendations.

Some of these standards and recommendations include IPR claimed by one or more organizations, which, to the best of the knowledge of the SLAM team, has been made available under the usual conditions of the ITU-T, ETSI and IETF. In addition, it is believed that there is no new material introduced in this Application Note (i.e. material that has not already been defined in other standards and recommendations) that is the subject of any IPR claim.

Appendix C – Common IP Video Impairments and Measurements

This Appendix describes the typical range of problems affecting IP Video performance, measurement techniques and criteria for the selection of appropriate SLA measurement strategies.
Problems affecting IPTV performance fall into two types: IP & MPEG TS Transport Impairments (Transport Layer) and Base-band Video & Audio Signal impairments (Application Layer).

IP Transport Impairments

IP Video streaming applications are much more sensitive to network quality of service than data applications such as email, or even voice applications such as VoIP. Quality of Service (QoS) refers to intelligence in the network to grant appropriate network performance to satisfy each service types requirements. For multi-media over IP networks, the goal is to preserve both the mission-critical data in the presence of multi-media voice and video and to preserve the quality of these services with the bursty nature of data traffic.

There are four traditional parameters generally used to describe quality of service: Latency or delay, the amount of time it takes a packet to transverse the network: jitter, the variation in delay from packet to packet; bandwidth, the data rate that can be supported on the network; and packet loss, the percentage of packets that do not make it to their destination.

Many of the problems affecting IPTV are IP related, generally categorized as “loss, delay and jitter”.

Packet Loss

Packets may be lost due to line noise, home networking problems, transmission errors, route changes, buffer overflows in routers, intentional loss introduced to trigger flow control and other causes. The STB decodes the received packet stream and attempts to reconstruct the audio and video signals, if packets are missing then gaps will occur in the reconstructed audio and/or video signal. Typical IPTV systems attempt to conceal the effects of packet loss by using forward error correction or packet retransmission, however some degradation may still occur (for example if too many consecutive packets are lost or if the retransmitted packet arrives too late).
Packet loss is often caused by transient network congestion. This implies that packet loss will occur more frequently during congested periods and less frequently between. Observations of packet loss distribution on IP networks do in fact show that packets are typically lost in sparse bursts - periods of several seconds during which the packet loss rate can reach 20-40 percent separated by longer periods during which lost packets are less frequent. It is therefore quite possible to have a program stream in which the average packet loss rate is only 1 or 2 percent but which experienced multiple periods of 30 percent loss, which would cause very noticeable audio degradation.

Jitter (Packet Delay Variation)

The transit time of packets through a network can vary significantly due to network congestion, route changes, load sharing and other causes. Short-term packet-to-packet delay variation is known as jitter or packet delay variation. A STB contains a de-jittering buffer designed to remove these delay variations however if packets arrive excessively late then they will be discarded.
As jitter is usually due to network congestion then it is strongly time varying. During periods of higher congestion the jitter level is increased, leading to a higher packet discard rate.

Delay (Latency)

In non-interactive IP Video services (such as video broadcast and video on demand) the user is essentially unaware of the round trip delay. However the round trip delay influences the channel change delay.

Duplicate and Out of Sequence

Duplicate or out-of-sequence packets should generally not cause problems with IPTV services although may indicate some network problems. An out-of-sequence packet is by definition associated with a jitter event (e.g. the packet is significantly delayed and hence arrives out of order) or with a route change that reduces delay.

Video Impairments

Blockiness (or Blocking)

Block distortion is a spatial degradation and is caused by coding impairments. It is characterized by the appearance of an underlying block structure in the image. This block structure is a common feature to all DCT-based video compression techniques. Technically, it is caused by coarse quantization of the spatial frequency components during the encoding process. In practice, Blockiness appears as high data compression ratios are used in order to transmit video content (especially) those with high level of motion) on low bandwidth networks.

Bluriness

Bluriness is a special degradation and is mostly caused by coding impairments. It is characterized by reduced sharpness of edges and loss of special detail. Technically, compression algorithms often cause Bluriness by trying to trade-off bits to code resolution and motion Bluriness is a common feature of wavelet-based video compression techniques. In practice Bluriness appears as an attenuation of high spatial frequencies in the image (fuzzy image). Some coding algorithms intentionally blur content prior to encoding while during playback some video players introduce processes that can smooth the image.

Transmission Distortion

Transmission impairments, such as packet loss or bit error, will impact differently on a viewer's perception depending on different variables. The effect of a packet loss is depends on the type of frame that is corrupted. The degree of error concealment depends on the codec itself and also on the video player. Finally, the video content (still, moving..) affects the extent of the perceived degradation.

Simple network performance statistics, such as percentage of packet loss or bit error rate, cannot therefore mirror quality as perceived by end-users since the same level or error can produce a wide range of different qualities.

The same video sequence with the same codec, same frame rate and bit rate suffering from transmission errors can produce very different quality results. The burstiness of the packet loss and the packets affected can have significant differences dependent on different packets that have been loss. For example if so-called I-Frames are loss their significance is much greater than others.

Jerkiness

Jerkiness (or jerky motion) is motion perceived as a series of distinct "snapshots", rather than smooth and continuous motion. It is commonly observed in video-telephony or video conferencing application and other low-bit rate video systems. When transmitting video data over low-bandwidth networks, encoding bit rates must be lowered by reducing the amount of information to transmit. As a consequence, the frame rate of the delivered video may be reduced. Jerkiness is the result of skipping video frames to reduce the amount of video information that the system is required to transmit to process per unit of time. Lack of motion smoothness can be due to frame dropped by the encoder or decoder, and repeated frames.

Mosquito effect (Gibbs noise)

Mosquito noise is most apparent around artificial computer generated objects or scrolling credits (lettering) on a plain colored background. It appears as some haziness and/or shimmering around high-frequency content (sharp transitions between foreground entities and the background or hard edges) and can sometimes be mistaken for ringing.
Unfortunately, this peppered effect is also visible around more natural shapes like a human body. Mosquito noise is a form of edge busyness distortion sometimes associated with movement, characterized by moving artifacts and/or blotchy noise patterns superimposed over the objects (resembling mosquito flying around a person's head and shoulders).
"Mosquitoes" can also be found in other areas of an image. For instance, the presence of a very distinct texture or film grain at compression will also introduce mosquito noise. The result will be somewhat similar to random noise; the mosquitoes will seem to blend with the texture or the film grain and can look like original features of the picture.

Audio Impairments

Basic Audio Quality

The main attributes, sub-attributes and example of common descriptive terms for the absolute assessment of sound quality are (see ITU-R Rec. BS.1284, §5.1 & Annex 1):

Attribute	Sub-Attribute	Examples of common descriptive terms
1 Spatial impression The performance appears to take place in an appropriate spatial environment	Homogeneity of spatial sound Reverberance Acoustic balance Apparent room size Depth perspective Sound color of reverberation	Room reverberate/ dry Direct/indirect Large room/small room
2 Stereo impression The sound image appears to have the correct and appropriate direction distribution of sound sources	Directional balance Stability Sound image width Location accuracy	Wide/narrow Precise/imprecise
3 Transparency All details of performance can be clearly perceived	Sound source definition Time definition Intelligibility	Clear/muddy
4 Sound balance The individual sound sources appear to be properly balanced in the general sound image	Loudness balance Dynamic range	Sound source too loud/ too weak Sound compressed/ natural
5 Timbre Accurate portrayal of the different sound. Characteristics of sound source(s)	Sound color Sound attack	Boomy/sharp Dark/light Warm/cold
6 Freedom from noise and distortions Absence of various disturbing phenomena such as electrical noise, acoustic noise, public noise, bit errors, distortions, etc.	Perceptible/imperceptible disturbances	
7 Main impression A subjective weighted average of the previous six attributes, taking into account the integrity of the total sound image and the interaction between the various parameters.		

Front Image Quality

The front image quality is related to the localisation of the frontal sound sources. It includes stereophonic image quality and losses of definition (see ITU-R Rec. BS.1116, §5.3).

Impression of Sourround Quality

The impression of surround quality is related to spatial impression, ambience, or special directional surround effects (see ITU-R Rec. BS.1116, §5.3).

Correlation between sound and pictures

The correlation between sound and accompanying picture may include the following characteristics (see ITU-R Rec. BS.1286, §5):

- correlation of source positions derived from visual and audible cues (including azimuth, elevation and depth);
- correlation of spatial impressions between sound and picture;
- time relationship between audio and video.

Measurement Techniques

Packet Impairments

Packet Loss

The principle of measuring packet loss is quite simple, basically detecting when packets fail to arrive. In practice this becomes slightly more complex, for example:

(i) packets may occasionally be significantly delayed by several hundred milliseconds, potentially due to a transient routing loop. The decision on whether a packet is lost therefore has to be delayed for a sufficient time to allow for "normal" packet arrival time variations but at some point packets are declared lost, even if they eventually arrive.

(ii) packets are sometimes duplicated, and primitive packet loss counting algorithms (such as RFC3550) may count duplicates as received packets. This means that if two percent of packets were lost and two percent were duplicated then some systems would report this as zero packet loss.

It is quite possible for one IPTV session with a low rate of packet loss to appear worse than another session with a high rate of loss. This may be due to the *distribution* of lost packets. Packet loss is often caused by network congestion and hence can often be strongly time varying; a call with a low average packet loss rate may actually be losing packets at rates of 20-30 percent during brief periods of congestion.

Many of the emerging QoS reporting protocols report the distribution of lost (and discarded) packets using a two state model - a burst state in which the packet loss rate is high (typically 5 percent or higher) and a gap state in which the packet loss rate is low. Loss distribution is typically reported as the mean length and density of burst periods and gap periods (a total of four parameters).

Jitter

Packets often suffer from some variation in transmission delay, again often due to congestion. The receiving STB or PC incorporates a jitter buffer to remove these variations however packets may arrive *too* late in which case they may be discarded. It is essential to measure jitter, discard rate or both in order to assess impact on IPTV performance.

Jitter measurements usually comprise some method of estimating delay variation on a per-packet basis and then averaging this.

P*acket to Packet Delay Variation (PPDV)*, defined in RFC3550 is calculated by subtracting the delay of one packet from the delay of the previous packet, and then calculating a running average using a scaling factor of 16. This is very commonly used however can give misleading results for some delay distributions.

Mean Absolute Packet Delay Variation (MAPDV), defined in ITU G.1020, is calculated by measuring the delay of a each packet relative to a moving average of delay. This approach is more stable and accurate than PPDV, and correlates well with packet discard rates in Voice over IP systems.

Packet Discard Rate, defined in RFC3611, reports the proportion of packets discarded due to jitter - i.e. reports the effects of jitter.

Combined Loss/ Discard Measurement

The effects of lost and discarded packets on audio & video quality are very similar and packet loss is often associated with high levels of jitter, hence it makes sense to report a combined loss and discard measurement. Most QoS reporting protocols combine loss/discard measurement with a burst model, reporting the length and density of bursts of loss/discard and the length and density of the gaps between bursts.

Delay

Delay comprises several elements: the transit time of packets through the IP network and associated access links, the delay within the IP endpoints (due to jitter buffer, encoding and decoding delays) and potentially delay within the non-IP part of the network.

Audio & Video Quality Estimation Algorithms

Two types of algorithm exist – packet based and signal based. Both can be used in non-intrusive (passive) or intrusive (active) monitoring. Typically, signal analysis is more computationally complex than packet based analysis, and both should be capable of providing an accurate estimate of user opinion.

Non-Intrusive Monitoring Algorithms

Non-intrusive monitoring algorithms are able to measure attributes of a session whilst passively observing either the packet stream or audio/video stream. This type of algorithm is suitable for estimating the quality of live sessions.

Packet Based Algorithms

Packet based algorithms could be based on IP layer, RTP layer or MPEG Transport Stream layer analysis.

Various vendors have developed proprietary packet-based audio and video quality estimation algorithms on the real time distribution of packet loss, delay and jitter, coding and transmission characteristics (e.g. codec, frame rate, bit rate, resolution, error correction/concealment, interleaving, etc).

In parallel, work on packet based algorithms is advanced in organisations such as ATIS IIF and the VQEG as well as in the IETF and is being introduced into the ITU's IPTV Focus Group.

The industry acknowledges that packet based algorithms are currently the most practical method of monitoring the perceived quality of video over IP in live network deployments because full reference measurements are not possible and no-reference signal based algorithms are still very processing intensive, especially for STB or multi-stream mid-network monitoring.

Potential limitations of packet based algorithms include difficulties in assessing the visual impact of specific packet degradations.

<u>Video Signal Analysis Algorithms</u>

There are currently no standardized methods for the non-intrusive measurement of perceptual video quality. However, the Video Quality Experts Group (VQEG; http://www.vqeg.org), which operates under the umbrella of the ITU-T and ITU-R, is currently carrying out large scale benchmarking/evaluation of several proposals for signal-based, non-intrusive models for video quality measurement that will be suitable to IPTV scenarios.

<u>Audio Signal Analysis Algorithms</u>

To be developed.

Active Testing Algorithms

Active or intrusive algorithms measure the attributes of test sessions. This allows them to measure service quality when there is no live traffic, or to measure aspects of sessions that non-intrusive techniques may be unable to.

<u>Packet Based Algorithms</u>

Not Applicable.

<u>Video Signal Analysis Algorithms</u>

When known test video signals are used the following methods can be used:

- The ITU-R Rec. BT.1683 and the ITU-T Rec. J.144 define a method for objective measurement of perceived video quality in the presence of full reference. The four models adopted for standardization are based on work from the Video Quality Expert Group (VEQG) – Phase II.
- The ITU-T Rec. J.147 defines an objective picture quality measurement method by use of in-service test signals.
- Peak-Signal-to-Noise-Ratio (PSNR) is a widely used video quality indicator based on pixel-by-pixel picture differencing but may not always predict accurately the user perceived quality, particularly in case of highly structured blocking and ringing impairments.
- The ANSI T1.801.03 standard uses a gradient model to define various objective video quality metrics based on the spatial and temporal features of the video stream that measure specific impairments:
    - Blurring measured with the Lost Edge Energy Parameter

- Block Distortion (Tiling) measured with the HV to non-HV Edge Energy Difference Parameter
- Error Blocks measured with Added Motion Energy Parameter
- Noise Measured with Motion Energy Difference Parameter
- Jerkiness Measured with Lost Motion Energy and Percent Repeated Frames Parameter
- Temporal Edge Noise Measured with the Added Edge Energy Frequencies Parameter

Objective parameters	**Impairments**
Maximum added motion energy	error blocks, jerkiness, noise
Maximum lost motion energy	jerkiness
Average motion energy difference	jerkiness, noise, error blocks
Average lost motion energy with noise removed	jerkiness
Percent repeated frames	jerkiness
Maximum added edge energy	spatial edge noise, block distortion, tiling, noise
Maximum lost edge energy	blurring, smearing
Average edge energy difference	blurring, smearing, spatial edge noise, block distortion, tiling, noise
Maximum HV to non HV edge energy difference	block distortion, tiling
Added edge energy frequencies	temporal edge noise, spatial edge noise, edge busyness
Maximum added spatial frequencies	spatial edge noise, block distortion, tiling, noise
Maximum lost spatial frequencies	blurring smearing

Audio Signal Analysis Algorithms

The ITU-R Rec. BS.1387-1 defines a method for objective measurement of perceived audio quality in the presence of full reference that can be used when the test signal is known. This method is also referred as Perceptual Evaluation of Audio Quality (PEAQ). Note however that PEAQ was validated against data collected with a subjective assessment method designed to assess only very small audio impairments.

The NTIA/ITS has developed Measuring Normalizing Block algorithm (MNB) (see ANSI T1A1.7/97-003R1). The MN1 algorithm measures the quality of narrowband voice by transforming the input and output audio signals into a perceptual domain. The perceptually transformed signals are then compared using the MNBs to detect frequency and temporal distortions in the output relative to the input. The output of this algorithm is auditory distance (AD). It is a measure of how different the output audio signal is from the input audio signal. Thus, larger auditory distances indicate poorer output audio quality.

Subjective Video Quality Metrics

Mean Opinion Score (MOS)

MOS is the ITU's definitive measure of perceptual voice, audio and video quality for a number of years and is accepted across the industry as the standard subjective quality assessment metric. MOS is the mean value given by subjects in tests when asked to measure quality of Audio and Video on a scale of 1 to 5.

Perception	Score
Excellent	5
Good	4
Fair	3
Poor	2
Bad	1

In order to predict subject scores a number of companies offer software assessment modules that take various aspects of a service, measure and weight them and then deliver a score.
There are three basic methods for achieving this one of which is based on the underlying IP or packet transport layer. As we have suggested in the previous sections the delay, jitter and packet loss in a network will affect the overall quality of a service. In looking at these factors and some finer ones such as the burstiness of packet loss, the size of jitter buffers on equipment and content type added to human factors, such as where in the session did it degraded, a predicted MOS value can be generated. This is effectively a passive model where the no reference signal is required to be injected into the network.

There are other MOS predictors that do require known or reference signals to be inserted into the network and then have the degraded signal compared against the original reference. These models look at the decoded video signal and are highly accurate. They are used to access both live and re-launch video services as well as a method to access equipment.

MOS provides a simple effected test of customer quality, which requires little understanding of network measurements and thus can be useful for high-level Service Level Agreement measurements.

Objective Video Quality Metrics

Peak-Signal-to-Noise Ratio (PSNR)

Is the ratio between a processed video frame and its synchronized reference video frame. For a degraded frame the synchronized reference frame is the frame in the reference sequence that has the same number in the sequence. Range: 0 to 48dB. A PSNR value is provided for each frame.

$PSNR = 10Log_{10} 255^2/MSE$

Note: PSNR is a frequently used video quality indicator. However PSNR does not always correlate well user perceived video quality (i.e. subjective MOS measurements according to ITU-R BT.500).

Mean Squared Error (MSE)

This is the square of the pixel-to-pixel difference (error) between a degraded video frame and its synchronized reference video frame.

Objective Measurement Methods

There are thre basic methods to perform objective measurements (ITU-T J.143):

Full Reference (FR)

This method is applicable when the full reference audio or video signal is available. This is a double-ended method and is the subject of ITU-R Rec. BS.1387 (audio), BT.1683 (video) and ITU-T J.144 (video).

Reduced Reference (RR)

This method is applicable when only reduced reference information is available. This is also a double-ended method. The ANSI T1.801.03 is an example of reduced reference measurement model.

No Reference (NR)

This method is applicable when no reference signal or information is available. This is a single-ended method. Typical NR metrics include blockiness and frame rate measurements.

Common IPTV Metrics

The following metrics have been adopted by several standardization bodies such as the ITU, ANSI and ETSI, and hence provide a common set of IPTV metrics for active or passive measurement of live or test sessions.

Common IPTV Metrics			
Metric Type	**Metric**	**Description**	**Reference**
Video Quality	MOSv	Subjective measurement of video quality	BT.500
	Vq	Objective measurement of video quality	BT.1683/J.144
	Vq(Aq)	Objective measurement of video quality accounting for the influence of audio quality	TR 102 479
	PSNR	Peak Signal to Noise Ratio	(See note)
	Rate of Disturbance	Frequency of noticeable video impairments (DSQ events)	
	Actual Frame Rate	Video frame rate at the decoder output	J.241
	Fluency	Fluency of video	
Audio Quality	MOSa	Subjective measurement of audio quality	BS.1284

	Aq	Objective measurement of audio quality	BS-1387
	Aq(Vq)	Objective measurement of audio quality accounting for the influence of video quality	BS.1286 (*)
	Rate of Disturbance	Frequency of noticeable audio impairments (DSQ events)	
Responsiveness	Channel Change Delay	Delay between channel change	G.1010
	DVR Delay	Fasf forward, Pause, Rewind delay	G.1010
	Play Response Delay	Delay between play and display of the first video frame	G.1010
Multimedia Quality	MOSav	Subjective measurement of overall multimedia quality accounting for the influence of both audio & video	BT.500
	MMq	Objective measurement of overall multimedia quality accounting for the influence of both audio & video	J.148/G.OMV TR 102 479 T1A1.5
	Lip Synchronisation	Delay between audio and video (positive or negative)	BT.1359 ETR 297 ATSC 191 ANSI T1.522 T1.801.04
	Rate of Disturbance	Frequency of noticeable video or audio impairments (DSQ events)	
Packet Metrics	Packet Loss Rate	Network packet loss rate	Y.1540/J.214
	Packet Discard Rate	Rate of packet discard due to jitter	Y.1540/J.241
	Burst Density	Density of loss/discard during bursts	TS 101 329
	Burst Length	Mean length of bursts	TS 101 329
	Gap Density	Density of loss/discard during gaps	TS 101 329
	Gap Length	Mean length of gaps	TS 101 329
Delay Metrics	Round Trip Delay	Packet path round trip delay	Y.1540
	End System Delay	Delay due to buffering/ decoding/ encoding in reporting endpoint	G.1020
Video Quality Parameters	Video Codec	Video codec profile	ISO/IEC
	Audio Codec	Audio codec profile	ISO/IEC
	Resolution	Video picture resolution	ISO/IEC
	GOP structure	Group of Picture Structure	ISO/IEC
	Bit rate	Video bit rate	ISO/IEC
	Nominal frame rate	Video frame rate at the encoder output	ISO/IEC
	FEC	Forward Error Correction	
Endpoint Configuration	Jitter Buffer Type & Parameters	Fixed or Adaptive, size, adjustment…	G.1020
	PLC Type	Silence insertion/ Normal/ Enhanced	G.1020
	Player Characteristics	Fast start streaming / Adaptative bit rate streaming	

The principal metrics to be used for development of SLAs are video quality metrics, audio quality metrics, responsiveness metrics and possibly combined metrics (i.e. metrics that measure the combined effect of audio, video & responsiveness).

Note: Although PSNR has been widely used as objective quality metric, it is broadly acknlowleged that PSNR does not always correlate well with perceived picture quality and therefore it should be used with caution.

Administrative Appendix

This Appendix provides additional background material about the TeleManagement Forum and this document. In general sections may be included or omitted as desired, however a Document History must always be included.

Document Life Cycle

The Voice over IP Application Note is being issued as Member Evaluation Version <<#>>.

The purpose of an Evaluation Version is to encourage input based on experience of members and the public as they begin to use the document. Following the Evaluation Period, documents that are seen to deliver value are candidates for formal approval by the TM Forum. All documents approved by the TM Forum undergo a formal review and approval process.

This document will continue under formal change control. Further work will be reflected in revisions to this document.

Document History

Version History

Version Number	Date Modified	Modified by:	Description of changes
0.1	March 2006	L. Philippart	Based on original document from Montreal 2005 TAW. Now focusing on IPTV services; draft proposal for Maidenhead SLAMM meeting.
0.2	June 2006	L. Philippart	Draft version produced in Maidenhead and subsequently.
0.3	26 June 2006	L. Philippart	Updated for team review
0.4	10 July 2006	L. Philippart	Includes comments from NCS and Psytechnics
0.5	12 July 2006	L. Philippart	Added Exec summary & conclusion.
0.6	12 Sept 2006	T. O'Sullivan	Final updates before submission to AC.
0.7	13-Oct-2006	T. O'Sullivan	AC approved, prepped for Member

			Evaluation
0.8	22-01-2007	L. Philippart	Comments from member evaluation reviewed during TAW Lisbon.

Release History

Release Number	Date Modified	Modified by:	Description of changes
1.0	June 2006	Team	Initial release.

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Documentation and work from standards bodies and other forums have also contributed to the evolution of the Video over IP Application Note. This access was via public information or TM Forum member knowledge. This list of standards bodies and forums is not exhaustive and does not imply review and concurrence by these organizations or their representatives. It is important however to acknowledge the work and their influence on the TeleManagement Forum work:

- ITU-R & ITU-T
- ANSI
- VQEG
- DSL Forum

About TeleManagement Forum

TeleManagement Forum is an international consortium of communications service providers and their suppliers. Its mission is to help service providers and network operators automate their business processes in a cost- and time-effective way. Specifically, the work of the TM Forum includes:

- Establishing operational guidance on the shape of business processes.
- Agreeing on information that needs to flow from one process activity to another.
- Identifying a realistic systems environment to support the interconnection of operational support systems.
- Enabling the development of a market and real products for integrating and automating telecom operations processes.

The members of TM Forum include service providers, network operators and suppliers of equipment and software to the communications industry. With that combination of buyers and suppliers of operational support systems, TM Forum is able to achieve results in a pragmatic way that leads to product offerings (from member companies) as well as paper specifications.

