Primer

A Guide to IPTV

The Technologies, the Challenges and How to Test IPTV

Tektronix

Table of Contents

Introduction .1
IPTV and the Triple Play .1
The Attraction of IPTV .1
How IPTV Works .2
Challenges in Delivering IPTV Services .2
IPTV Technology Overview .5
Video Compression Technologies .5
Network Protocols .6
UDP or User Datagram Protocol .6
RTP or Real Time Protocol .7
RTSP or Real Time Streaming Protocol .7
IGMP or Internet Group Management Protocol .8
Network Evolution .8
Network Architectures .9
Access Network Technologies .10
xDSL .10
HFC .10
FTTx .10
WiMAX .11
Control and User Planes .11
IPTV Network and Transmission Errors .12
Video Problems .12
Physical Layer and Protocol Stack Problems .13
Testing IPTV Networks .14
The Technology Lifecycle .14
IPTV Test Methodology .15
Test Tools and Technologies .16
Cross Layer Measurement and Test .16
Distributed Multi-layer Monitoring Tools .17
Layer-specific Probes .17
Extended Monitoring Capability .18
Multi-layer Monitoring .18
Picture Quality and Quality Indicator Tools .20
Subjective and PQ tests .20
Quality Control for Stored File Based Content .20
QoS Measurement Using MDI .20
Tektronix in the Converged World .21
Conclusion .22
Glossary .23

Table of Figures

Figure 1. Various Unicast / Multicast Scenarios 3

Figure 2. Video Coding Trends .5

Figure 3. IP Packet Format 6

Figure 4. IP/UPD/RTP Packets .6

Figure 5. RTP Headers 7

Figure 6. RTSP Protocol 7

Figure 7. Network Architecture 9

Figure 8. xDSL 10

Figure 9. FTTx Characteristics 10

Figure 10. Effects of Dropped Packets 12

Figure 11. Physical Layer and Protocol Stack Problems 13

Figure 12. Technology Lifecycles 14

Figure 13. Simplified IPTV Network 15

Figure 14. IP–MPEG TS Cross Layer Correlation . . .16

Figure 15. Examples of Multi-Layer Test Points 19

IPTV

Enabling innovation in the new digital world

Ensuring IPTV Quality of Experience in a triple play environment

High consumer expectations have put pressure on content producers, network operators and equipment manufacturers to deliver consistently high quality audio and video to their end users. The proliferation of enabling technologies has resulted in a wide variety of formats and standards adding to the complexity of the challenges faced in establishing new services and entering new markets. With a broad portfolio of products and deep technology expertise in both modern telecommunications networks and video, Tektronix is uniquely positioned to deliver test, measurement and monitoring solutions in the IPTV environment.

Tektronix Solutions:

- Baseband Video & Audio
- Analyzers & Diagnostics
- Monitoring Probes
- Monitoring Systems
- File Based Content QC (quality control)

Is my source video good?

Source
Content can be validated in the form it was received or after being ingested.

Did it encode properly?

Video Headend
The video head end must ensure video quality after compression and encryption and provide secure storage for future access.

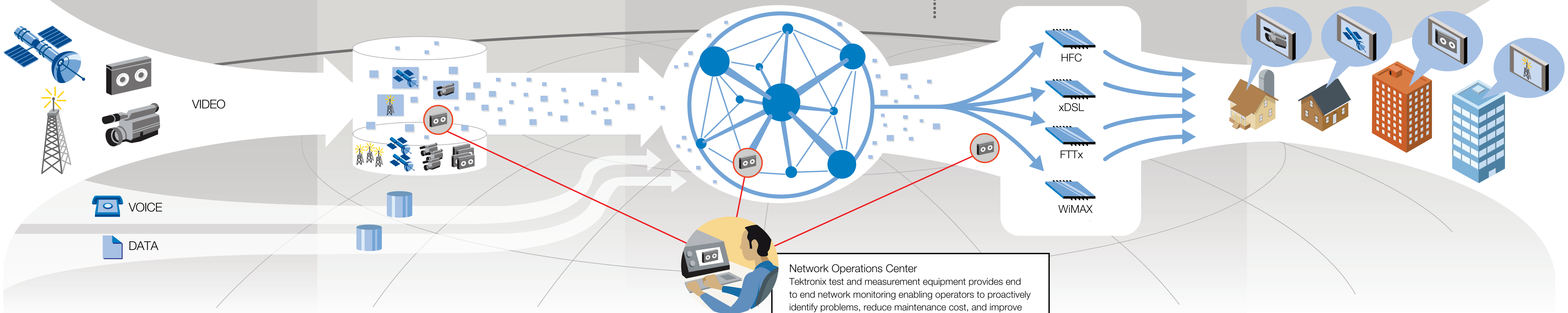
Did it get there?

Core IP Network
Video is intolerant of best effort environments. It requires adequate bandwidth, no packet loss and minimal jitter.

Access Network
Delivering content across diverse access technologies and networks requires broad protocol support and connectivity solutions.

Was it a good experience?

User Experience
Meeting and exceeding consumer expectations makes our customers successful. Verifying content quality and designing, deploying and managing high quality networks are critical to achieving desired QoE levels.



Network Operations Center
Tektronix test and measurement equipment provides end to end network monitoring enabling operators to proactively identify problems, reduce maintenance cost, and improve customer service.

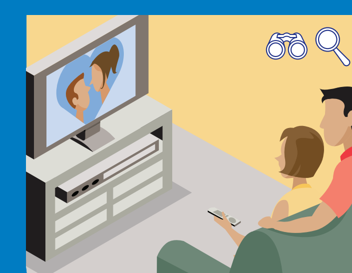
How Tektronix can help:

DESIGN

DEPLOY

MANAGE

VERIFY CONTENT



Content Creation and Post Production

Challenge: Create content that looks and sounds great while meeting broadcast standards and customer requirements.

Tektronix Solutions:
- Baseband Video & Audio
- Analyzers & Diagnostics
- File Based Content QC

Research & Development

Challenge: Ensure standards compliance and design integrity. Quickly and efficiently test and validate, reducing time to market and overall development cost.

Tektronix Solutions:
- Baseband Video & Audio
- Analyzers & Diagnostics

Equipment Manufacturing

Challenge: Eliminate errors and quality issues from reaching your customers. Ensure your products meet design parameters and comply with industry standards.

Tektronix Solutions:
- Baseband Video & Audio
- Analyzers & Diagnostics

Systems Integration

Challenge: Quickly and efficiently roll out new services; integrate new equipment and technologies; and accelerate the initial return your customers expect from their vestments.

Tektronix Solutions:
- Baseband Video & Audio
- Analyzers & Diagnostics
- Monitoring Probes
- Monitoring Systems
- File Based Content QC

Network Engineering

Challenge: Monitor appropriately to proactively address degrading network conditions and when a problem does arise, identify the source, isolate and correct the condition before it impacts the end user.

Tektronix Solutions:
- Baseband Video & Audio
- Analyzers & Diagnostics
- Monitoring Probes
- Monitoring Systems

Network Operations

Challenge: Monitor and maintain content integrity and ensure the right content gets to the end user when expected. Ensure your team has the tools necessary to identify potential disasters and avoid them.

Tektronix Solutions:
- Baseband Video & Audio
- Monitoring Probes
- Monitoring Systems
- File Based Content QC

Tektronix
Enabling Innovation

Introduction

The Digital Video Broadcasting Project was founded in 1993. The initial task was to develop a complete suite of digital satellite, cable, and terrestrial broadcasting technologies in one ‘pre-standardization’ body. The DVB Project was formed from broadcasters and equipment manufacturers and made use of ETSI standards for the physical layers, error correction, and transport for each delivery medium.

The cornerstone of the technology was the transport mechanism, the MPEG-2 Transport Stream. This ISO/IEC standards based mechanism defines the syntax and semantics of a bit stream in which packetized digital video and audio are carried, along with the timing mechanisms which allow the signals to be decoded by a set-top box (STB). This work has become the foundation of IPTV systems having provided a well documented and reliable data transport arrangement for the delivery of video over various physical layers.

Coupled with the standardization of the Transport Stream there are several other technologies that have enabled the rollout of IPTV. These include new compression technologies like H.264/AVC and VC-1 (allowing more efficient use of the limited bandwidth links to the home), improved system security and Digital Rights Management (providing confidence to the content providers in these systems), IP core networks and faster more cost effective access technologies (such as WiMax, ADSL, etc.).

Despite the maturing of the enabling technologies, the deployment of IPTV presents many technical challenges to those required to successfully provide these services. This document explores some of those challenges and how Test and Measurement equipment can be used to facilitate the design, rollout and management of these systems.

IPTV represents the convergence of the broadcast and telecommunications worlds. Successful deployment requires tools and expertise from both worlds. Tektronix provides a wide portfolio of products designed to address the converging world, those products having been derived from our long experience in both Video and Telecommunications test and measurement.

IPTV and the Triple-play

Triple play is a term used to describe the delivery of voice, video and data services to the home. There are a number of commercial offerings that deliver these services to the consumer over different access technologies to the home but true Triple Play normally provides these services through a single connection to the home (e.g. fiber to the home). IP technologies are not necessarily used to deliver these services, For example, cable companies may deliver “Digital Voice” over QAM and DOCSIS systems.

IPTV is a component of the Triple Play. IPTV is used to describe the delivery of broadcast quality video over an IP network. Note this is not the same as streaming video over the public Internet, which relies on third party decoders to be used, or downloaded on to the decoding device e.g. a PC. This primer does not consider streamed video.

The Attraction of IPTV

IP networks offer two-way, interactive capabilities which traditional TV technologies lack. This type of interactivity will enable ‘one-to-one’ distribution, allowing individual viewers control of their chosen content along with so called ‘trick mode’ facilities like live pause, fast forward and rewind. This interactivity can also be used to provide targeted advertising, one-to-one marketing that could include instantaneous end-user feedback and other services coupled to programming such as online shopping (for articles actually shown in the program), gaming, etc.

The two way nature of these networks enable Video on Demand (VoD) and network digital video recording (NDRV), which are two of the most popular differentiators provided by IPTV systems over the traditional unidirectional broadcast system where programming is pushed to the consumer rather than pulled when required.

How IPTV Works

In standard broadcast systems all of the normal broadcast channels (e.g. CNN, HBO, etc.) are delivered to the STB in the home (via Cable, Satellite or Terrestrial). There could be hundreds of channels, all of which are delivered simultaneously. The STB tunes to the desired channel in response to requests from the viewer's remote control. As a result of this local tuning the channel changes are almost instantaneous.

In order to preserve bandwidth over the final link to the house, IPTV systems are designed to deliver only the requested channel to the STB. Note there could be several programs (or channels) delivered to different IP addresses in the same home (i.e. separate STB's or other IP enabled receivers).

In order to change channels, special commands are sent into the Access network requesting a change of channel. There is a complex protocol exchange (using IGMP "Leave" and "Join" commands) associated with this technique. This exchange requires a finite time to complete and the time taken is heavily influenced by transmission delays in the network which in turn has a direct impact on the channel change timings of the system. In essence, in IPTV systems the channel change is made in the network and not on the local STB. While preserving precious last mile bandwidth this approach presents a number of challenges to the scalability and usability of the system.

Broadcast TV makes use of IP Multicasts (and IGMP as mentioned) to deliver the programming efficiently through the IP system. A Multicast is designed to allow multiple users simultaneous access to the session.

VoD employs unicast IP services using the RTSP control mechanism. At the request of the viewer, the selected programming is located from within the network (from a server) and a unique unicast is setup to deliver the program to the user. This is in effect a private network connection between the server and the viewer's STB.

The variable nature of a sample network is shown in Figure 1. Various Unicast / Multicast Scenarios.

Unicasts, Multicasts, IGMP and RTSP are discussed in detail later.

Challenges in Delivering IPTV Services

Video, voice and data are all IP data services, but each has its own Quality of Service (QoS) requirements when being transported across IP networks.

In order to be successfully decoded at the STB, the Transport Stream carrying the video needs to arrive at a known and constant bit rate, in sequence with minimal jitter or delay. The requirements for the successful delivery of voice or data are just as important but less stringent than those needed by video. The differing characteristics of these services all contribute to the complexity of designing, deploying and maintaining networks required to deliver high quality services to the consumer.

By their very nature, IP networks are "Best Effort" networks initially developed for the transport of data. As a consequence these networks are susceptible to lost or dropped packets as bandwidth becomes scarce and jitter increases. In the vast majority of cases this problem has no significant impact on data services which can cope with packet resends and packets arriving out of order as they get routed along different paths through networks. Video is completely intolerant to the vagaries of a best effort network. QoS (Quality of Service) for video services requires:

1. **High availability and sufficient guaranteed bandwidth to allow the successful delivery of the service.** Without this, video delivery will be "bursty" which will cause issues at the Set Top Box (STB) which expects its data at a constant bit rate and in the correct sequence.
2. **Low transmission delay through the network.** This impacts quality of experience as it will impact the response time to requests from the user's remote control.
3. **Low network jitter.** Jitter affects the variability of packet arrival through the network. This variability can lead to buffer under- and overflows at the receiving equipment (STB). Jitter can impact the way packets are handled at various net work elements. If the jitter is too high, packet loss will increase as queuing software tries to load balance traffic at network elements.
4. **Low Packet Loss.** Lost packets have the greatest impact on the quality of received video and will generally lead to highly visible blocking errors. If lost packets contain I-frame Video the impact will be more pronounced as the STB has to wait for the next I-frame to arrive to allow it to "reset" itself. This problem is aggravated by the use of H.264 which uses a longer GOP (Group of Pictures) structure (increasing the chances of lost frames) and because of the increased compression ratio each frame contains more information. Consequently, the loss of a single H.264 frame is likely to have a greater impact on the picture quality.

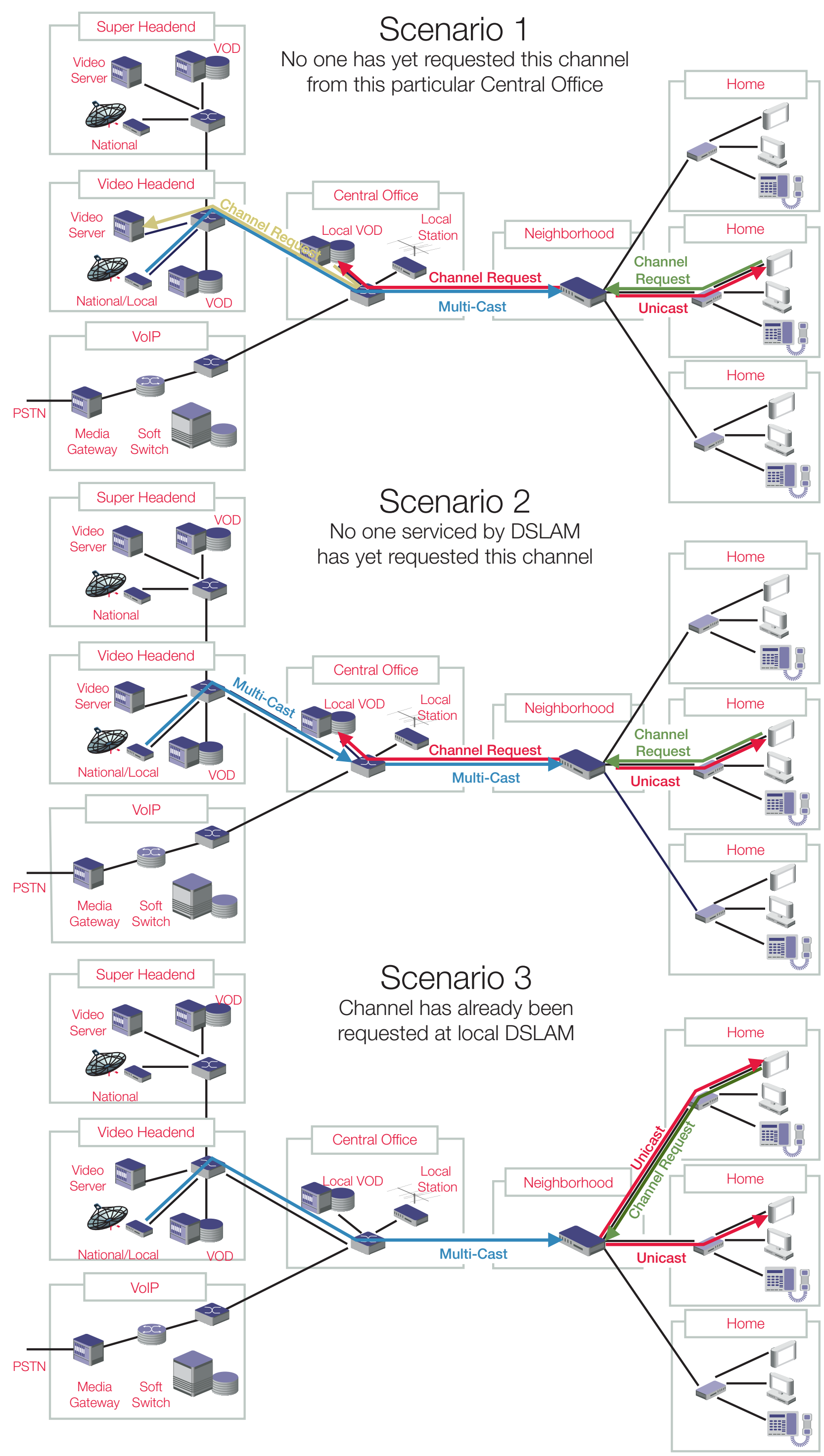


Figure 1. Various Unicast / Multicast Scenarios.

From a pure bandwidth perspective, video requires more bandwidth than voice or data. In order to ensure quality of service, effective network management policies in tandem with a well designed, robust network are essential. A standard video signal using MPEG-2 encoding uses approximately 3.75 Mbps of bandwidth over an IP network. A high definition MPEG-2 signal may require 12-15 Mbps. By adding a mix of services together such as video on demand, standard and high definition channels (along with voice and data services), it is easy to see that the bandwidth budget of most access technologies can quickly become restrictive.

Today's broadcast systems have set current user expectations about a television experience. To be successful, any IPTV service will have to be at least as good as the next best alternative, be that traditional broadcast, cable or satellite services, and probably better. Quality of Experience (i.e. is the subscriber receiving content adequately?) has become a critical element to IPTV services. QoE needs to consider items such as:

- Is the subscriber consuming more bandwidth than is provisioned?
- Are all promised capabilities being delivered (e.g. the Electronic Program Guide (EPG), the right program, etc.)?
- What is the subscriber experiencing at this moment (e.g. picture quality, channel change times, etc.)?

Quality of Service (i.e. is the network delivering content adequately?) at the packet level is as important as QoE. Good QoS provides the foundation from which high QoE expectations are more likely to be met. QoS focuses on the performance of the network and its ability to deliver content to the required standard. Examples of quantifiable QoS measures could be:

- Where in the network is congestion occurring?
- Are there adequate VoD content servers in appropriate markets?
- Are interactions between voice, data and video causing problems within the network?
- How many IP packets are being lost, and is there excessive jitter on the physical layer?

Given that QoS and QoE are vital pointers to whether subscribers will receive a quality end product, the ability to test monitor and measure the parameters that impact these is essential.

IPTV systems consist of a number of key components (often referred to as the Ecosystem) all of which can have an impact on the QoE and QoS. Some of the most important components are:

- **Middleware** – The software and hardware infrastructure that connects the IPTV components together. It normally includes subscriber-facing EPG, application control, back office/billing, etc.
- **STB (Set Top Box)** – The Consumer Premise Equipment (CPE) used to interface with the user and the IPTV services provided by the network.
- **Video Encoder/Transcoder/Stream Processor** – Responsible for the transformation of an input stream that can be of various formats into a digital compressed stream targeting the CPE.
- **Core Network Elements** – The key elements used to make up the Next Generation core network capable of prioritizing Video, Voice and Data through the network.
- **Access Network Technologies** – Access technologies capable of providing the bandwidth required to deliver TV services to the home or receiving equipment (for example: ADSL2, FTTx, WiMax, DVB-H).
- **Video Servers** – Computer based multi-stream playout devices connected to large storage systems.
- **CAS/DRM** – A Conditional Access System (CAS) allows for the secure delivery of content. Digital Rights Management (DRM) controls subscriber usage of the delivered content (for example: view once, unlimited view during calendar window, etc.).

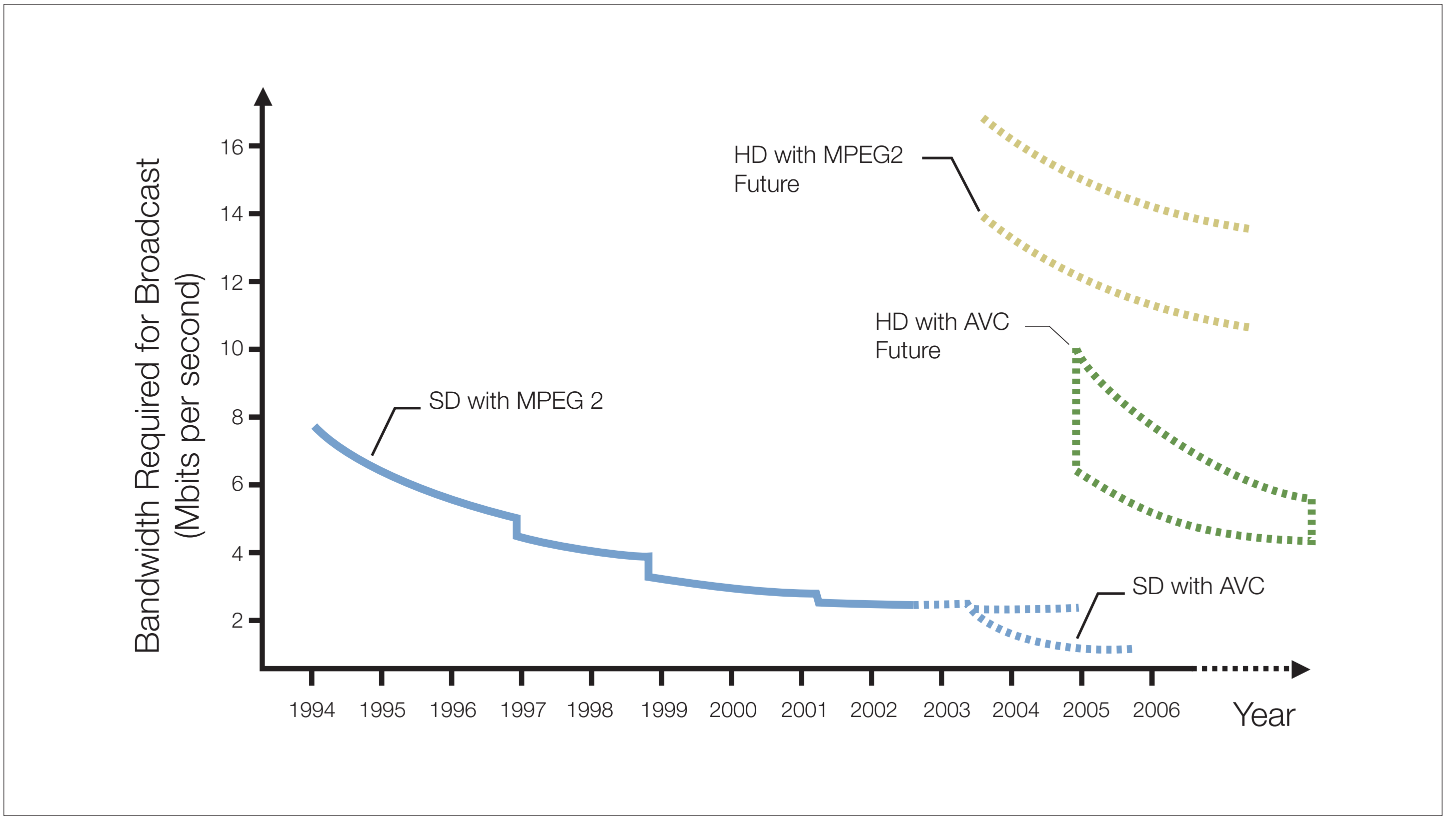


Figure 2. Video Coding Trends.

IPTV Technology Overview

Video Compression Technologies

Digital TV systems came to fruition during the '90's and are accessible worldwide across satellite, cable and terrestrial broadcast networks. They use MPEG-2 compression systems that have also been used for early deployment of IPTV by telcos and cable companies. As mentioned earlier, a standard video signal using MPEG-2 encoding uses about 3.75 Mbps of bandwidth over an IP network. A high definition signal may require 12-15 Mbps. So in order to deliver 2 channels of SD encoded TV to a home, almost 8 Mbps bandwidth is required. If xDSL is being used to access the home, it is easy to see why bandwidth is an issue. One way to alleviate bandwidth restrictions is to use new video compression technologies such as H.264/AVC or VC-1. H.264 can offer up to a 50% reduction in bandwidth utilization for the same picture quality compared to existing MPEG-2 compression. The progression in encoder technology is shown in Figure 2. Video Coding Trends.

Bandwidth is one consideration when selecting the compression technology to be used in the system. However there are a number of other factors that need to be considered. Using MPEG-2 encoding, the average Group of Pictures, or GOP length, the Group of Pictures between I-frames is approximately 12 – 18 (see the Tektronix MPEG Primer for a full description of GOP's). Using H.264 encoding, this GOP length could be as long as 300 frames; This makes the video stream even more susceptible to dropped packets, as each H.264 encoded frame effectively contains more information (because of improved compression efficiency), and so losing H.264 frames is likely to have a greater impact on the viewing experience. Beyond technical considerations there are a number of other things to be contemplated such as availability of commercially viable encoders and receivers (STB's), patent and royalty payments and interoperability with other network components.

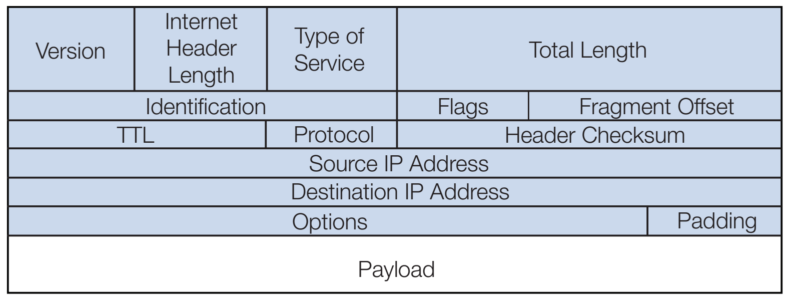


Figure 3. IP Packet Format.

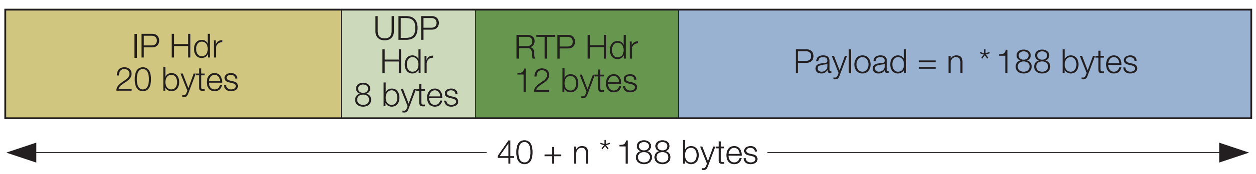


Figure 4. IP/UPD/RTP Packets.

Network Protocols

No study of IPTV can be complete without some understanding of the protocols used in these systems. These include IP transmission protocols such as UDP and RTP, and also signaling protocols such as RTSP and IGMP. Although these are the protocols in this document, it is far from an exhaustive list. There are many more protocols to be considered in modern networks - MPLS, SIP and PIM just to name a few. These are beyond the scope of this document.

UDP or User Datagram Protocol

UDP is defined in IETF RFC 768 and is one of the core protocols of the IP protocol suite. The term 'datagram' or 'packet' is used to describe a chunk of IP data. Each IP datagram contains a specific set of fields in a specific order so that any receiver knows how to decode the data stream. Many protocols can be encapsulated within the IP datagram payload.

One of its main advantages of UDP is its simplicity that reduces the amount of overhead carried, compared to the amount of data in the payload. The datagram headers contain:

- 16 bit source port address.
- 16 bit destination port address.
- 16 bit length field.
- 16 bit checksum.

The 16 bit length field therefore defines a theoretical limit of 65,527 bytes for the data carried by a single IP/UDP datagram. Figure 3. IP Packet Format shows the framing of an IP packet/datagram.

In practice, this UDP packet length means that it can carry up to 7 (188 byte) Transport Stream packets.

It is the simplicity of UDP that can cause issues. Its stateless form means there is no way to know whether a sent datagram ever arrives. There is no reliability or flow control guarantees such as are provided by TCP, which can identify lost packets and re-send them as necessary. UDP has been described as a 'fire and forget' protocol because it is difficult to discover if a packet has been lost before the subscriber does. In an IPTV environment, where it is essential that the video data is delivered reliably and in the correct sequence, the use of UDP can be precarious.

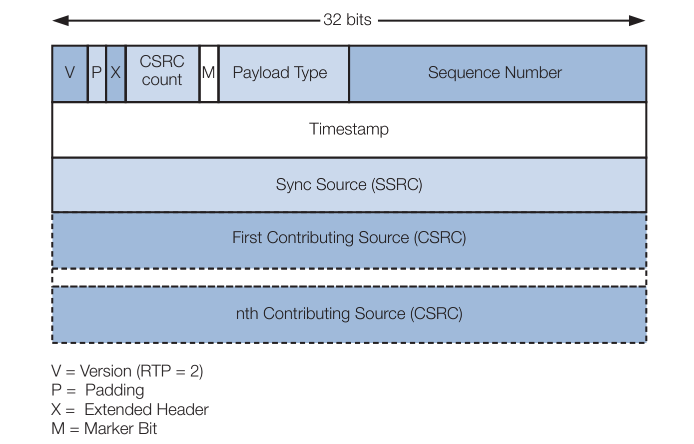


Figure 5. RTP Headers.

RTP or Real Time Protocol

RTP is defined by IETF RFC 3550 and IETF RFC 3551 and describes a packet-based format for the delivery of audio and video data. RTP actually consists of two closely linked parts:

- **Real Time Protocol** provides time stamping, sequence numbering, and other mechanisms to take care of timing issues. Through these mechanisms, RTP provides end-to-end transport for real-time data over a network. Use of sequence numbering also enables lost or out of order packets to be identified.
- **Real Time Control Protocol** is used to get end-to-end monitoring data, delivery information, and QoS.

Although RTP has been designed to be independent of the underlying network protocols, it is most widely employed over UDP. When MPEG-2 video is being carried, the RTP timestamp is derived directly from the 27 MHz sampled clock used by the Program Clock Reference (PCR) carried within the Transport Stream, thus further ensuring good timing synchronization.

It is, however, important to note that RTP does not define any mechanisms for recovering from packet loss, but lost packets can be detected as described above. Figure 5. RTP Headers shows the header format.

RTSP or Real Time Streaming Protocol

RTSP is defined by IETF RFC 2326 and describes a set of VCR like controls for streaming media. This is shown below in Figure 6. RTSP Protocol.

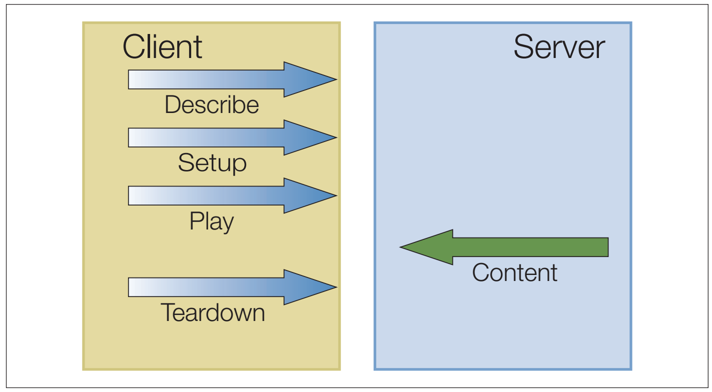


Figure 6. RTSP Protocol.

Typically, RTSP messages are sent from client to server, although some exceptions exist where the server will send to the client. In IPTV systems, RTSP is used in VoD applications for the consumer (client) to access and control content stored at the VoD servers. VoD is essentially a one-to-one communication established using unicast. Unicast is the exact opposite to broadcast, in which we send information to all users on the network. Unicast allows the VoD service to be requested by and sent to a single user.

IGMP or Internet Group Management Protocol

IGMP is defined by several IETF RFCs, the latest version being RFC 3376. IP multicasting is defined as the transmission of an IP datagram to a "host group". This host group is a set of hosts identified by a single IP destination address. In an IPTV system, the host group would be a set of subscribers who wish to receive a particular program.

In practice, what this means is that the transmission systems using IGMP do not send all the content to all the users. Multicasting, using IGMP, allows control of which content goes to which users and therefore controls the amount of data being sent across the network at any one time.

IGMP is the protocol used to handle channel changes in an IPTV system. In response to remote control commands, a series of IGMP commands to leave the current multicast and join a different service are issued. The time that it takes to execute these commands has a direct impact on channel change times. Middleware providers are working on a variety of different schemes to improve channel change response times.

Network Evolution

Before we begin to talk about some of the key system level issues, there is a need to begin with a brief discussion on where network architectures have been, and more importantly where they are heading. Ten to fifteen years ago, the first push by telcos into IP was primarily done so as an adjunct network to their already existing PSTN networks. For telcos, voice was still offered over PSTN, while data was sent over the IP networks.

On IP networks, there were many different technologies in use, such as Frame Relay, ATM, x25, etc. From a network operator's perspective, they had the ability to offer IP data services, but at the expense of managing a host of networks that were driving up operational and maintenance costs. Consequently the focus turned towards collapsing the networks that could offer IP services into a single IP network. Initially the primary application was data, but recently, voice has been added in the form of Voice over IP (VoIP), and now video over IP services are evolving.

From an "ideal network perspective" the long-term vision is to have an all IP network that can offer converged video, voice and data services over a single network. In this environment, network operators will be able to offer bundled services at lower costs, while at the same time lowering their costs to manage and deploy new services. Total convergence has yet to materialize. In most cases, IPTV is currently being deployed over a separate network and not yet part of a converged all IP network. The reality is that although the services may appear bundled from an end-users perspective (such as billing), most service providers are not yet at the point of having triple-play services over a single all IP converged network.

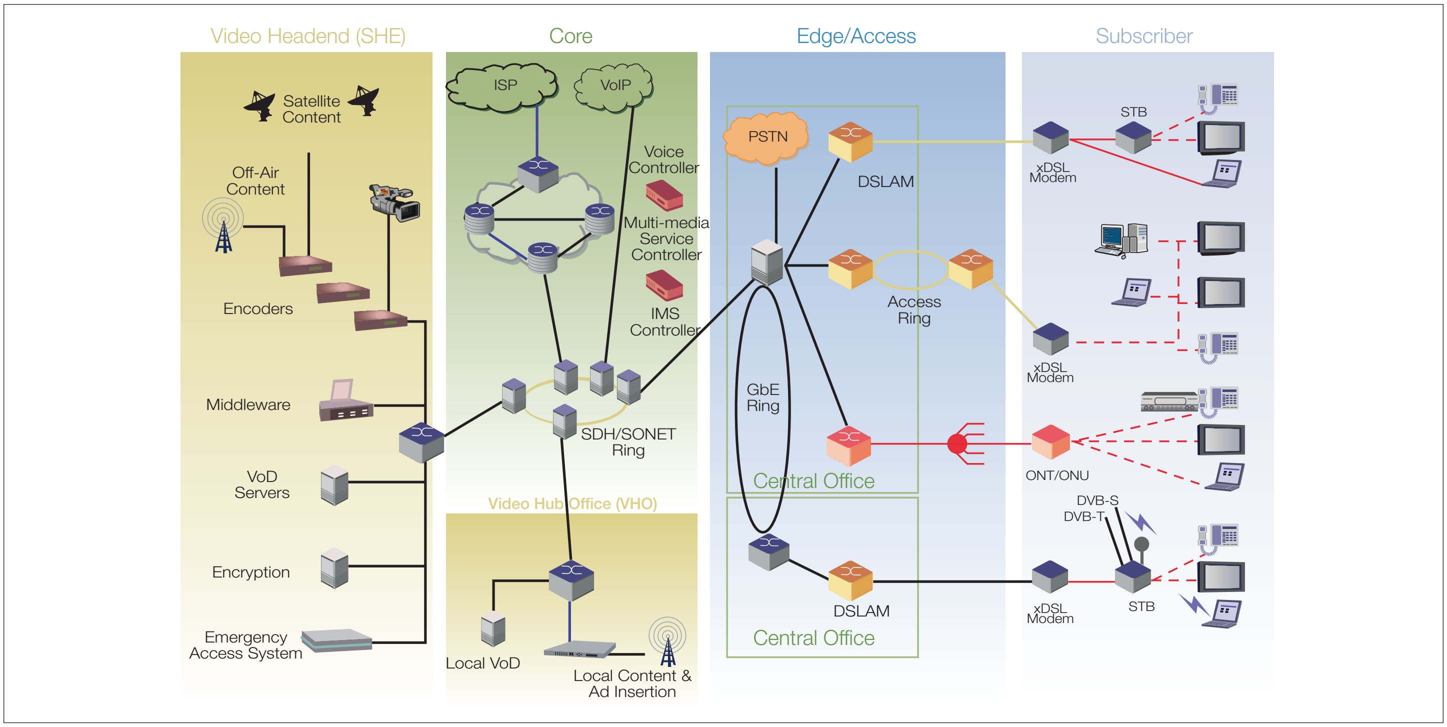


Figure 7. Network Architecture.

Network Architectures

Figure 7. Network Architecture shows an example of a typical IP network structure. Content is ingested from the left side of the diagram at the Video Headend. Video content can be delivered into the Headend in a variety of formats (both compressed and uncompressed) over a number of different delivery mechanisms including Satellite (normally from national stations) and Terrestrial transmissions (normally from local stations). From here, the data is encoded, packetized, and multiplexed appropriately for receipt by the CPE (normally in the form of an MPEG Transport Stream) and then sent to the Core network.

The data and VoIP subsystems are usually connected into the Core network. The Core network is used for the transmission of services at a national (or even global level). The system services (Voice, Video and Data) are then passed to the Access Network for distribution over the "last mile" to the consumer.

There is a variety of Access network technologies used to reach the subscriber, dependent on what sort of connectivity is required at the subscriber site. Telcos, for example, may rely on legacy ATM subsystems delivering data over xDSL on copper cables to the subscriber. Newer 'green field' developments may install direct 'Fiber to the Home' or 'Fiber to The Curb' (FTTx) delivering 100 Mbps direct to the subscriber. Legacy cable networks may use Hybrid Fiber Coax (HFC) to deliver new services over existing plant. All of these additional technologies add complexity to the distribution models and depending on which access technology is used, bandwidth may need to be carefully managed to ensure good QoS and QoE for the subscribers.

The technologies used in Access networks are reviewed in more detail on the next page.

Technology	Downstream	Upstream	Reach
'Original' ADSL	8 Mbps	~684 kbps	~10,000 feet
ADSL2	Up to 14 Mbps	Up to 800 kbps	Limited to 16,000 feet
ADSL2+	Up to 24 Mbps	800 kbps	< 5,000 feet
VSDL	> 50 Mbps > 13 Mbps	> 2 Mbps < 1 Mbps	1,000 feet 5,000 feet

Figure 8. xDSL.

Active FTTP (Point to Point)	PON
■ Point to Point architecture	■ Point to multi point architecture
■ Active components are needed at the end of each fiber and in the outside plant	■ Passive optical couplers replace regenerator and amplifiers. Cheaper and more reliable
■ Each subscriber requires an optical port at the Central Office	■ Can couple up to 64 Optical Network Units (ONUs) onto a single fiber
■ Expensive components are dedicated to a single subscriber	■ Active components like lasers are shared over many subscribers

Figure 9. FTTx Characteristics.

Access Network Technologies

xDSL

DSL is a distance sensitive technology that makes use of the existing copper twisted pair infrastructure. In general, the greater the distance from the exchange, the slower the data rate needs to be to ensure reliable delivery of the service. Typical data rates for given distances are shown in Figure 8. xDSL.

It can be seen that bandwidth, both upstream (data from the consumer) and downstream (data to the consumer), can vary considerably depending on distance. This needs to be carefully considered when designing the Access networks to carry video content. Each subscriber connected to the exchange does so through a Digital Subscriber Line Access Multiplexer (DSLAM) which terminates the DSL circuits and aggregates them. It also separates out the VoIP components.

HFC

Hybrid Fiber Coax (HFC) systems combine the use of high-speed fiber backbone to deliver data out to the edge of the network, using coaxial cable to run the 'last mile' connecting the subscriber to the backbone. Cable networks have used this type of system since the early 1990s, and similar to xDSL, it is again possible for the operators to re-purpose existing plant whilst providing new services for the consumers.

HFC is a more efficient medium than xDSL over copper twisted pair, being able to provide greater bandwidth over greater distances.

FTTx

Fiber optic cable is capable of carrying high bandwidth data over great distances. Consequently, Fiber to the Home (FTTH), Fiber to the Premises (FTTP) or Fiber to the Curb (FTTC) it is possible to deliver 100 Mbps or higher to the home.

The two main fiber systems being deployed today are Active FTTP and Passive Optical Networking (PON). The cost of installation tend to be much higher than other Access technologies but FTTx does have the advantage of providing a single, broadband pipe capable of delivering simultaneous video, voice and data services under the control of the network operator.

WiMAX

WiMAX is not a description of a specific technology, but rather an indication of conformance and interoperability for equipment built to the IEEE 802.16 family of wireless standards. WiMAX could be used to provide last mile broadband connectivity and could offer lower cost alternatives to other Access technologies (e.g. Fiber or xDSL). Potential use scenarios include rural areas, to non-traditional service providers (e.g. electrical utility companies providing triple play services) who lack Access networks, and the satellite providers who lack a back channel or easy access to IP technologies.

In this environment, WiMAX Forum Certified™ systems could typically provide bandwidths of up to 40 Mbps per channel, for fixed and portable access applications at a cell radius of between 3 and 10 kilometers. Pure mobile network deployments could provide up to 15 Mbps of capacity within a typical cell radius of up to 3 kilometers.

For more information: http://www.wimaxforum.org/home/

Control and User Planes

The Control Plane is the portion of the network that carriers control information, sometimes referred to as signaling, whereas the Data Plane, or User Plane, is the part of the network that carriers the actual content.

In the case of IP networks, the Control Plane's primary function is to set up the pathway across which content can be delivered. For multicast deliveries, this involves IGMP signaling that is used to set up and maintain sessions across the network. In the case of unicast transmissions, RTSP signaling is used to establish one-to-one connectivity and allow control commands to be transmitted from the STB.

Whereas the control plane sets up the pathway, the User Plane has to handle the content carried across the established path. This is the provisioning of the service within the network, along with the actual video, audio and other data required for the service itself.

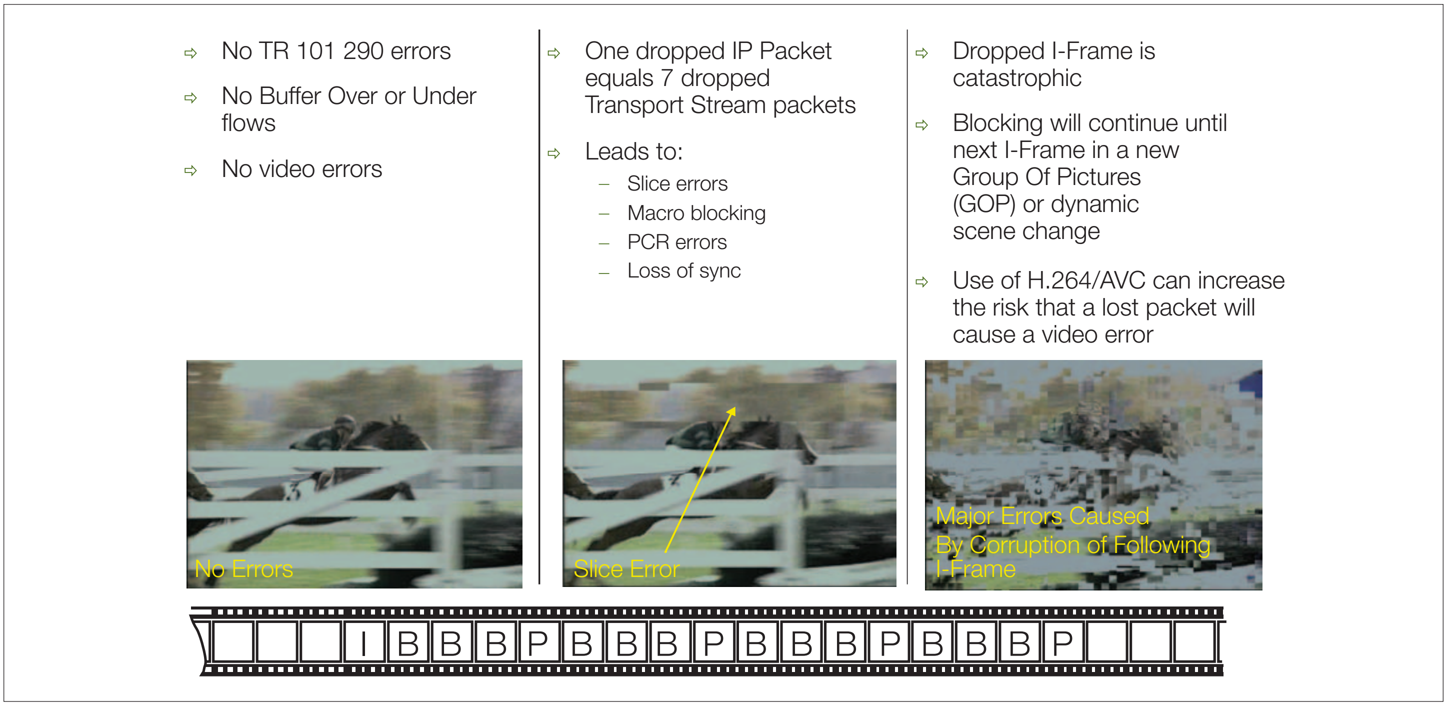


Figure 10. Effects of Dropped Packets.

IPTV Network and Transmission Errors

Video Problems

As previously mentioned, the successful transmission of video through an IP network requires:

1. High availability and sufficient guaranteed bandwidth to allow the successful delivery of the service.
2. Low transmission delay through the network.
3. Low network jitter.
4. Low network packet loss.

Of these, packet loss has by far the greatest impact on the QoE. To understand why this is the case it is necessary to understand how MPEG encoding works.

MPEG encoding compresses the video frames into three different types of frames, I-frames, B-frames, and P-frames. An I-frame contains all the information in one frame of the video stream such that an MPEG decoder can recreate the original frame using only the information from the I-frame, i.e. it contains 100% of the information required to recreate the picture.

To achieve the required video compression, special spatial and temporal encoding techniques (see Tektronix MPEG Primer for a full description) are used to create B and P-frames that contain partial information associated with the I-frame. The picture is recreated using the I-frame and the compressed information in the B and P-frames. A B-frame is an incrementally encoded video frame than can only be decoded using the information in its associated I-frame. A P-frame is an incrementally encoded video frame that can only be decoded using the information in its associated I-frame and B-frame.

These I, B and P-frames are carried across the network in 188 byte MPEG Transport Stream (TS) packets which are encapsulated in IP packets. A single IP packet is capable of containing approximately seven TS. Dropping any packet, but particularly those that contain I-frames, can lead to serious QoE issues. Figure 10. Effects of Dropped Packets illustrates the impact on the video as a result of dropped packets at the network level.

The sequence moves from left to right. On the left there are no dropped packets and all other quality indicators (as defined by ETSI TR 101 290) are good, and therefore there are no issues with decoding the picture. The ETSI TR 101 290 document describes a test and measurement methodology to ensure repeatable results in Digital Video Broadcasting (DVB) based systems.

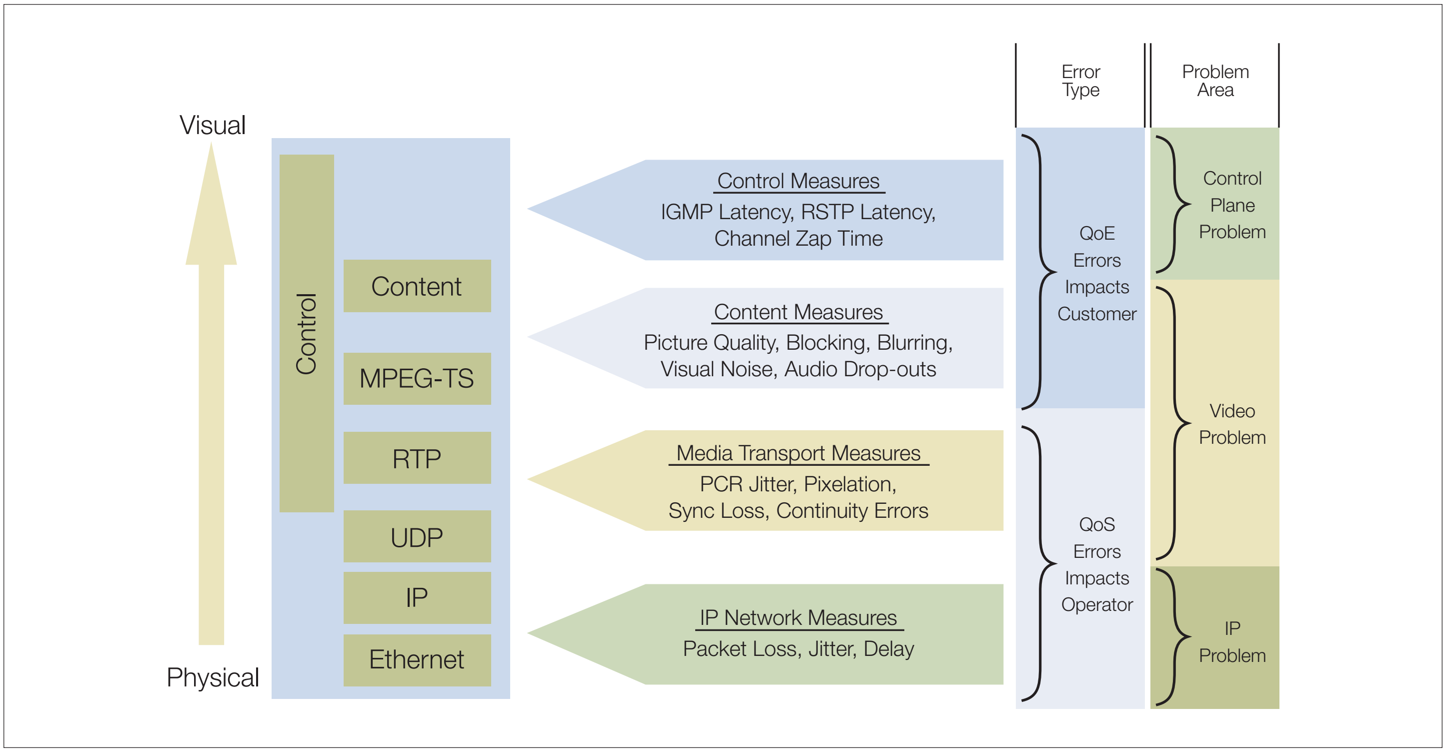


Figure 11. Physical Layer and Protocol Stack Problems.

The next picture in the sequence shows a picture error. This error has occurred as a consequence of a dropped Transport Stream packet. In this case the result is an image that contains a slice error but other potential symptoms including blockiness, blurring, and stuck or frozen frames. Symptoms of this type can continue and worsen until the reference is reset with the arrival of a new I-frame. The effect could be visible for a very short time or last several seconds. The timing of the next I-frame will depend on the length of the encoded group of pictures (GOP). This may be 15 frames, as in MPEG-2 or it may be 60, 100 or 300 frames when using more advanced codecs such as H.264/AVC. At 25 frames per second the error could take a significant and visibly noticeable time to correct.

The final picture indicates the impact of losing a complete I-frame. This has a catastrophic effect on picture quality. As a consequence of losing the I-frame the decoder in the STB has completely lost its reference from which to decode the relevant B and P-frames. This situation will only recover upon the correct receipt and decode of the next uncorrupted I-frame.

Not all packet loss will result in unacceptable video quality. Long-term stability of the network and establishing a steady state environment will depend on Engineers determining if an IP disturbance will cause unacceptable video performance in the network environment. This is an iterative process and requires test tools that allow cross layer (MPEG TS to IP layers) measurement correlations to be made.

Physical Layer and Protocol Stack Problems

Figure 11. Physical Layer and Protocol Stack Problems shows a conceptual representation of the IPTV multilayer model. The drawing provides an indication of the type of errors that can occur, their potential causes and their impact.

From Figure 11 it is clear errors that occur in the lower levels of the stack generally present themselves as QoS errors and consequently have the greatest impact on the Operator. In the physical IP layer the errors manifest themselves as IP packet loss, jitter and delays (or latency). In the media layer (sometimes referred to as the User Plane) errors are typically caused by excessive PCR jitter, sync loss, continuity errors and pixelization. Although some of these errors could result directly from IP physical layer problems it is worth noting that they could also be introduced as part of the video encoding process.

Progressing toward the top half of the stack, the errors become more visible to the consumer and consequently are categorized more as QoE issues. Problems here are normally associated with either the content or the signaling and control used to set up the session. A list of potential problem areas is shown above.

Figure 11 provides an indication of the type of parameters that need to be measured and monitored in a network, or as equipment is being developed and deployed.

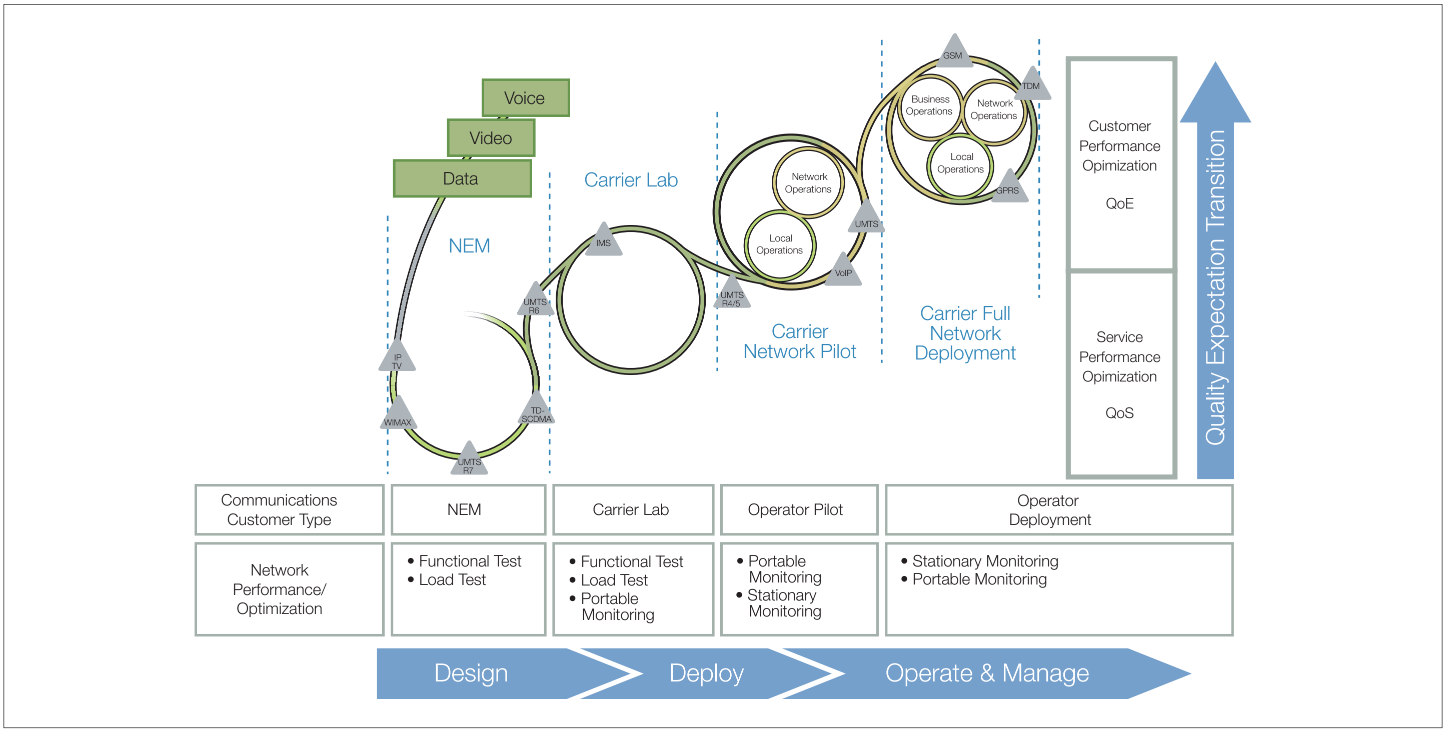


Figure 12. Technology Lifecycles.

Testing IPTV Networks

The Technology Lifecycle

The deployment of new technologies tends to follow a similar lifecycle that begins with early research, development and standardization and ends with full deployment and service management. Not all technologies make it to full deployment for commercial, technological, or even political reasons. This is illustrated above in Figure 12. Technology Lifecycles.

The lifecycle diagram shows a number of different technologies moving through the design and deployment stages to a point where the services are fully operational and managed. The lifecycle itself consists of four distinct rings and it is important to note that any one technology may go round each ring a number of times as the technology advances and matures (or in some cases fails). As the technology progresses, the type of customer (e.g. Network Equipment Manufacturers (NEM)) and location (e.g. telecom Carrier Lab) tends to change, and at each stage the test requirements and type of test equipment evolve (as illustrated in the table along the bottom of the diagram).

It should be noted that as the technologies progress through the lifecycle there is a transition in the quality expectations delivered (shown on the right of the diagram). In the early stages, QoS parameters are significant as early researchers strive to get the base technologies to work (either standalone or interoperating with other equipment) and deliver the service. As the technology matures, and many of the QoS issues are resolved, the focus switches to optimizing the QoE delivered to the subscriber since it is this that will provide a differentiator between service providers.

As illustrated the test tools required through the lifecycle go from those that provide deep diagnostic capabilities (particularly in the early stages) to those capable of monitoring national and even global networks. Consistency of measurement becomes an important factor throughout the process if operators are to develop and deploy services quickly.

Engineers need to rapidly identify, diagnose and remedy problems with components and infrastructure and should not spend time trying to interpret different measurements from different test tools. Tektronix has strived for consistency of measurement across a broad portfolio of test products built from deep expertise in both the broadcast and telecommunications world. Measurements taken on diagnostic analyzers will match those taken by monitoring equipment.

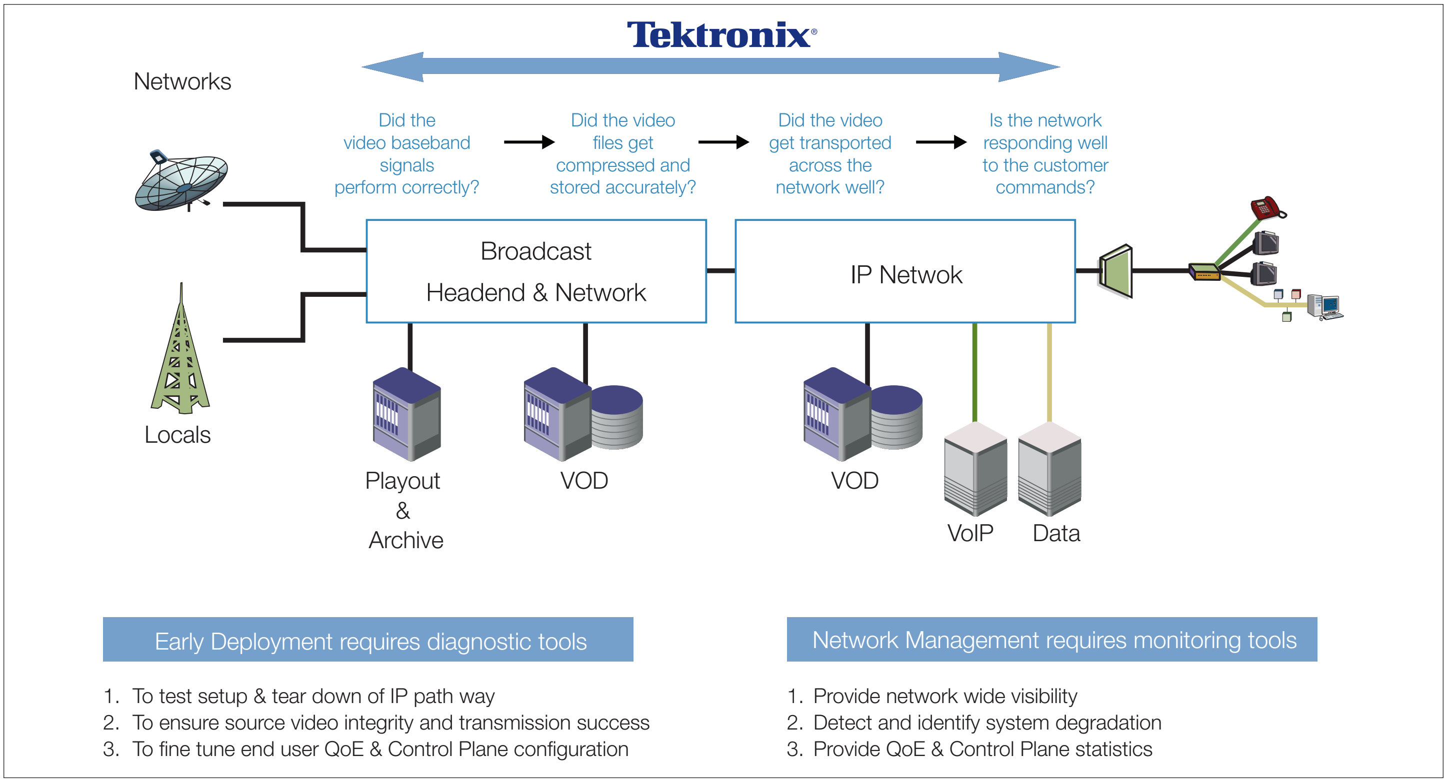


Figure 13. Simplified IPTV Network.

IPTV Test Methodology

Figure 13. Simplified IPTV Network shows an IPTV network reduced to its simplest form. In effect these systems consist of two key subsystems:

1. The video headend, where video is ingested and made ready for transmission to…
2. The IP network, the transmission system used to distribute the video along with voice and data services.

There is a continuum along which the deployment of IPTV is moving (see Figure 12. Technology Lifecycles). This goes from design and manufacturing to full deployment. At each stage the test objectives and needs are different, and in order to move efficiently and cost effectively through each stage the need for testing and the selection of the right test equipment is essential.

Design requires in-depth tests for standards compliance and interoperability of video and network infrastructure equipment. Manufacturing requires consistent, rapid and repeatable functional test and results logging. Early deployment and trials requires tools that provide rapid fault identification and diagnosis of IP, video and voice faults. This requires best-in-class point-monitoring and analysis solutions that can look at the content (User Plane), control and setup (Control Plane), and physical layer.

For full deployment and ongoing system management the emphasis will shift from monitoring and testing parts of the network to 24/7 real-time monitoring of the whole network (which could be global). Network-wide monitoring cannot not be trusted to small niche players. Best-of-breed system suppliers must be used.

Today, most of the major IPTV rollouts are in an early deployment phase. The key objective during this phase is for operators to “get it working”.

There are three key steps that need to occur during this phase of deployment.

1. **Can the “IP Pathway” be reliably set up and torn down?** A triple play network needs to assure availability of network resources and bandwidth to deliver video services. However, video is bandwidth intensive, so it is equally important to ensure that pathways that are no longer needed can be torn down successfully. This requires test equipment capable of establishing and testing the IP pathway and providing statistics on network jitter and packet loss. These needs also apply to the provision of VoIP (Voice over IP) services.

2. **Is the video right at the source and destination?** Once the IP pathway is established it is then essential that the video data pushed into and received from the pathway is correct. This requires the monitoring and analysis of the Transport Streams at the output of encoders, multiplexers at the headend. At the receiver end, similar monitoring and analysis is required to ensure there has been no degradation of the video as it passes through the system.

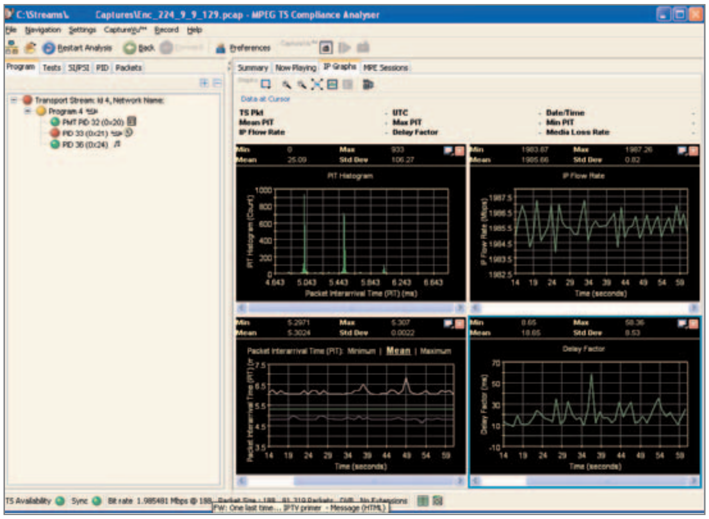


Figure 14. IP-MPEG TS Cross Layer Correlation.

Two areas often overlooked when discussing IPTV are the ingest and storage facilities within the Headend. Significant amounts of content to be distributed through the system are ingested over RF downlinks (e.g. QPSK for Satellite, and COFDM or 8VSB for terrestrial broadcasts). The integrity of these links is as important as any other in the chain. Being at the front end of the transmission chain, any errors introduced at this stage will be propagated throughout the system.

Much of the video content is stored on servers prior to playout into the system (be it Broadcast TV or VoD). Corruption of the stored video can similarly lead to the propagation of poor video through the system.

3. **Is it a great subscriber experience?** The final stage of the early deployment phase is to configure the system to deliver a high QoE. This requires the optimization of Control Plane (IGMP and RTSP) parameters. Engineers are required to ensure that the requested channels are actually delivered and there is access to the EPG, etc.

   This can be an iterative process and will have to be repeated as new services are added, and the system scales up to cater for larger numbers of subscribers.

Throughout these stages engineers require test equipment that brings the broadcast and telecom's worlds together. The equipment is required to perform comprehensive Quality of Service (QoS), e.g. network jitter, lost packet, and QoE Control Plane and User Plane (e.g. IGMP response, PCR jitter) measurements. To aid rapid fault isolation and diagnosis the ability to see and correlate errors across the different layers of the network is essential. Not all the errors that occur on the IP layer cause video errors. It's important to understand which do and which don't.

Test Tools and Technologies

In general terms, IPTV is another version of general broadcast TV, and so existing test methodologies evolved over many years still work well.

MPEG test and measurements such as referenced in the DVB TR101 290 standard can be used to detect timing impairments at the Transport Stream layer and to detect lost or out of order packets using Continuity Counter tests. However, for a more complete solution, it is better to perform these in conjunction with IP tests, throughout the network.

Monitoring can tell the Service Provider how big an issue is occurring in the network, but knowing when or why a video service is off air or delivering poor quality is critical. A Service Provider may or may not know there is problem, but the subscriber will. Reputation, quality and business will suffer with persistent problems.

Cross Layer Measurement and Test

Cross layer testing can be employed in both the IP and MPEG domains, to trace and track performance degradation, so that action may be taken before the problem gets too serious.

Viewing graphical plots, such as those in Figure 14. IP-MPEG TS Cross Layer Correlation, helps correlate events over time, as they happen on one layer, and to see if it has effect on another. This can help isolate the root cause to a specific layer, allowing a fault to be traced and then corrective action taken.

The ability to collect MPEG Transport Stream statistics like Sync Loss, Continuity Count, PID error and PCR statistics, and transport MOS and video elementary stream statistics pre- or post-FEC can prove to be invaluable to engineers during fault diagnosis. During fault situations, alarms, tracer files and fault logs can be time stamped allowing intermittent faults to be tracked to see if an MPEG stream had a Transport Stream layer fault, or if it was related to an IP event.

The DVB TR101 290 test recommendations contain many MPEG layer parameters that test equipment can flag. The specific ones that get affected by dropped and out-of-order packets on the IP layer are the Sync Loss, Sync Byte error, Continuity Counter error (a 4-bit rolling counter on a PID-packet identifier basis) and packet checksum, or CRC.

Most MPEG streams contain a built-in timing packet – Programmable Clock Reference or PCR. Graphing of these parameters (i.e. PCR Inaccuracy and PCR Overall Jitter) gives a good indicator of a stream suffering timing distortions due to packet bunching or network jitter. Together with a simultaneous graphical display of IP Packet Arrival Interval (PIT), it is possible to time correlate with PCR, and even with PTS (Presentation Time Stamp arrival) on elementary IP streams with no embedded PCRs.

PCR Overall Jitter (PCR_OJ) or PCR Frequency Offset (PCR_FO) can be compared with PIT stability to assess the source of IP or MPEG introduced jitter. There are standards for PCR, but none for IP interval or jitter. These are user defined and test equipment should allow user limits to be set to aid fault diagnosis. With some MPEG-4 transports that don't strictly need PCR, some operators are re-establishing PCR feeds, as they take up little bandwidth and give fast indication of timing health.

Finally, cross-layer test can extend to RF layers, where off-air content is acquired for delivery over IP networks. Test probes at the RF layer can not only give indications of RF signal quality, but can also demodulate the signal and perform MPEG tests to detect any issues already present on the down link.

To summarize, the advantages of Cross Layer testing are:

- Being able to Graph Packet Arrival Interval (i.e. to show burstiness).
- Time correlate Packet Arrival, PCR and PTS arrival interval graphs.
- Identify underlying IP layer errors like CRC, dropped packets or out-of-order packets.
- Identify all DVB TR 101 290 errors.
- Time correlate errors at IP, TS and even RF layers to identify root causes.
- Errors can be time stamped, with layers identified in the error logs.

Distributed Multi-layer Monitoring Tools

As IPTV deployments mature and move into the Operate and Manage phase of the deployment lifecycle the test emphasis will shift from deep analysis and diagnostics to one of 24/7 monitoring. These monitoring solutions are required to provide operators with wide visibility and information about their systems.

These monitoring systems should have a combination of the following characteristics:

- Layer-specific probes that detect the different types of errors seen in digital television systems.
- Extended monitoring capability to give operators advanced notification of system degradations before they become quality problems.
- Multi-layer monitoring that lets operators quickly isolate the root cause of a quality problem.

Layer-specific Probes

In a monitoring system, each monitoring device can be considered a probe, monitoring quality at a particular point and layer in the distribution and transmission chain.

Operators need to use different probe types for quality control at different layers.

At the Formatting layer, digital waveform monitors help operators detect many quality problems essentially checking adherence to broadcast and colorimetry standards. They belong to a larger collection of Formatting layer probes which include:

- Digital audio monitors.
- Picture quality monitors for detecting blockiness and other picture impairments.
- Audio/video delay monitors.
- MPEG protocol monitors.

At the Distribution layer, operators need probes to detect quality problems in a wide variety of distribution and transmission channels. Probes in this group include devices to monitor Cable, Satellite and Terrestrial RF transmissions which can have their own national schemes (e.g. DVB, ATSC, or ISDB formats). In IPTV systems they will also include probes for monitoring information sent through either telecommunication Core or Access networks.

Extended Monitoring Capability

Monitoring probes can also be distinguished by the level of monitoring they offer.

Basic confidence monitoring probes track a small set of key quality parameters. They act as an “indicator light,” telling the operator when something has gone wrong. However, basic confidence monitoring probes generally do not offer a complete set of in depth precision measurements. While they can enhance the operators’ ability to react to a quality problem, they do not offer the information needed to proactively address system degradation before it becomes a quality problem.

Extended confidence monitoring probes use more sophisticated analysis to make additional measurements of quality parameters. They act as “indicator gauges” telling the operator when something is going wrong.

RF transmission monitoring offers a good example of this distinction. Basic RF confidence monitors measure bit-error-rate (BER). BER will remain low until the transmission approaches the digital cliff, then increase dramatically as the transmission falls off the cliff. This gives operators only slightly more time to react than they would have by watching the transmission on a picture monitor.

Extended RF confidence monitors add additional measurements like Modulation Error Ratio (MER) or Error Vector Magnitude (EVM). These measures will noticeably change as system performance degrades, giving operators early warning of potential quality problems, and an opportunity to make adjustments or seamlessly transition to backup systems.

Multi-layer Monitoring

To have confidence their facilities are performing correctly and efficiently, operators will generally need to probe at multiple layers in their systems. Probing at only one layer can give a misleading picture of system health.

Watching a broadcast on a picture monitor is a simple example of this methodology. In this case, operators are probing quality at the Formatting layer. While this offers significant information on system health in an analog system, it offers little information in digital systems. Similarly, monitoring just the MPEG protocol or the RF transmission will only yield partial information. To gain a complete picture of system quality, and to quickly detect and isolate quality problems, operators will need multi-layer monitoring solutions.

A small sample of test/monitoring opportunities is shown in Figure 15. Examples of Multi-Layer Test Points.

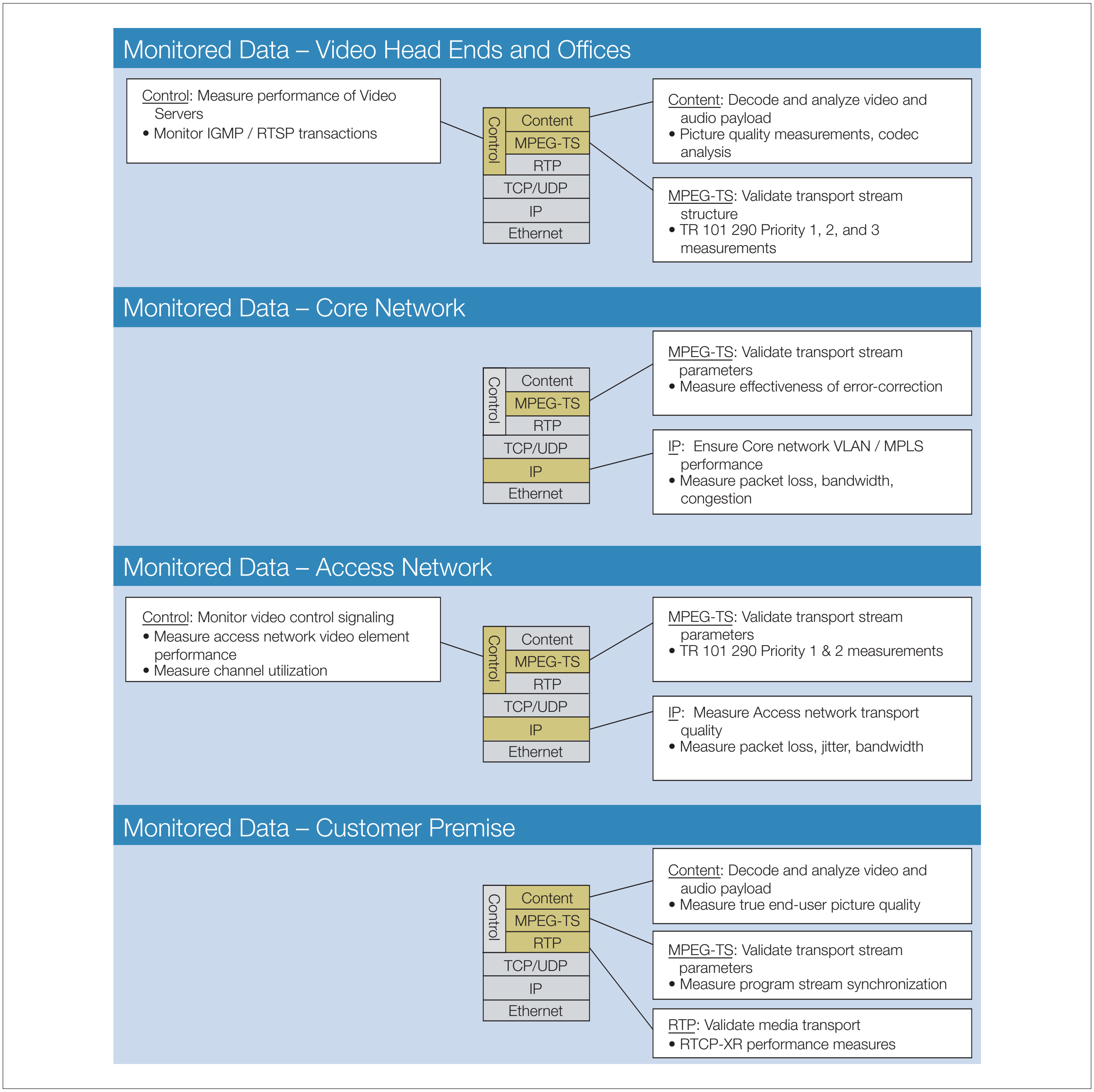


Figure 15. Examples of Multi-Layer Test Points.

Picture Quality and Quality Indicator Tools

Subjective and PQ tests

Mean Opinion (MOS) scores for VoIP are in common use in telecommunications systems and therefore there is a natural expectation that the video MOS should be made available. However, audio is Pulse-code Modulated (PCM) and degradation parameters for video are quite different. Objective measurements must be designed to correlate directly with a real subjective test database to be of any value. In audio this was done with deployment scenarios such as multiple languages, cultural groups and Codecs. There is no video equivalent for these scenarios.

True Picture Quality Analysis (PQA) can be expensive to implement and the more meaningful double-ended test (the received picture is compared to an unimpaired reference) has the problem of how the reference stream is routed to the point of test, often requiring a dedicated test feed.

Single-ended MOS tests may be more attractive, but there is no “ideal” algorithm available for this. To be successful these systems need to apply combinations of algorithms since they are very dependent on external factors such as the STB decoders, the number of dropped packets, encoder type and bit rate.

However PQ tests require the picture to be “in the clear”. If the operator is using encryption technologies such Conditional Access (CA) or Digital Rights Management (DRM), MOS will not work.

It should also be noted that these PQ metrics tell you when something has already gone wrong and, although they can give some indication of the cause (some TR 101 290 tests), they do not provide a complete test methodology.

Quality Control for Stored File Based Content

MPEG or IP transport monitoring is unlikely to pick up lower layer elementary stream coding artifacts. Picture Quality tests may detect the issues but cannot fix them. It is better to prevent coding errors from being transmitted or correct them at source.

Modern broadcast networks make extensive use of server technology for archive and playout purposes. There is a new breed of test tools that provide automated QC (quality control) capabilities for file based video, on the video file server before it is broadcast. Here, checks can be carried out on encoding profiles and baseline payload quality all in a wide variety of formats (SD-SDI, H.264 etc).

QoS Measurement Using MDI

Media Delivery Index (MDI) is defined by IETF RFC 4445. It is defined as a single figure of merit used to quantify two IP transport impairments, namely Packet Jitter or Delay and Packet Loss. These two test parameters are defined as Media Delay Factor (MDI-DF) and Media Loss Rate (MDI-MLR).

The Delay Factor indicates how long a data stream must be buffered (i.e. delayed) at its nominal bit rate to prevent packet loss. It does capture impairments on the IP layer, and will give you a general idea of network jitter from the DF measurement.

However, there are two important points to note; firstly, MDI is not payload aware. That is, it cannot separate video traffic from other data and VoIP packets. Secondly, the raw UDP protocol does not provide any means to detect packet loss. So for raw UDP, the packet loss portion of MDI is not measured directly, it must be extrapolated by using the MPEG Continuity Count error counts. For RTP flows, DF is measured using the timestamps from the received packets. The presence of RTP sequence numbers also allows RTP packet loss to be measured and displayed as part of the MDI.

The MDI-DF can give a measure of congestion in a network, by showing utilization level, and detect if queuing is happening in network components, but does not indicate how much of this is due to video packet bunching. The MPEG buffer model is not as simple as MDI assumes.

The Media Loss Rate is the number of packets lost during a period of 1 second.

MDI is expressed as a ratio:

“Delay Factor : Media Loss Rate”, e.g. “MDI = 150:14”

This example shows a Delay Factor of 150 ms & 14 packets lost per second.

A good MDI does not mean a faultless IP transmission, and a bad MDI can be the result of non-IP related issues. A more complete solution is to use MDI in conjunction with MPEG layer protocol test, i.e. cross-layer measurements.

Testing carried out at ingest can ensure good content is entering the network before being handed off onto the IP network. These video streams are then encapsulated and passed into the Core network along with the VoIP and high speed data traffic. Owning the entire network provides major advantages at this point, since quality of service (QoS) tools and network management protocols (e.g. PIM) can be used to prioritize the video traffic to prevent delay or fragmentation.

Tektronix in the Converged World

Tektronix has broad application experience and deep domain knowledge in both video and telecommunications network management and diagnostics – making them the only vendor that plays a role on both the video and communications side of the convergence of voice, video, and data. Tektronix is focused on blending their video and communications expertise to deliver the right test methodologies and solutions to help bring triple play, including IPTV, to market.

From the broadcast video side, Tektronix' industry leadership offers the broadest (across multiple standards and video layers) and deepest (in depth fault diagnosis and analysis) solution for compressed video test. Tektronix video test portfolio offers products to address triple play video applications.

Our monitoring tools provide extended confidence MPEG transport stream monitoring for broadcasters and network operators who need a scalable solution to detect signal degradation caused during transmission and distribution, ensuring QoE for the subscriber. These tools enable operators to easily and cost effectively perform rapid multilayer fault detection and diagnostics. This will minimize downtime and improve the reliability of networks.

Our MPEG analysis tools were the world's first Compressed Digital Video Debugger/Analyzers that could be applied anywhere at any level, to see and solve the most subtle, complex, and intermittent DTV problems in the minimum time. These tools provide Real Time Video over IP analysis and recording.

The delivery of VoD services requires extensive use of server technology to store and stream the video to the subscriber. The Tektronix automated content analysis tools provide the ability to automatically verify the quality of stored video and audio content before it is transmitted. This equipment tests video and audio stored in compressed and uncompressed formats and is designed to detect errors that are likely to be missed by human monitoring.

On the Communications side of the equation, the Tektronix Internet Protocol Diagnostics (IPD) product portfolio contains products designed for IP analysis. These solutions are targeted at users who require in depth IP test, measurement and statistics about the IP layer and the Control Plane. These solutions extend beyond IPTV into the Triple Play environment with VoIP test capabilities.

As full IPTV system deployment occurs the focus of operators will shift from diagnostic tools towards 24/7 real time monitoring capability and service assurance. Tektronix is a leading supplier of Service Assurance solutions for next generation networks providing products capable of monitoring complete global networks.

As a consequence of our expertise in both broadcast and telecommunications, consistency of measurement across our portfolio becomes a major differentiator for our customers. This is important as it allows engineers more time to diagnose the real fault rather than trying to explain differences in measurement results between diagnostic and monitoring equipment.

Conclusion

The delivery of video over Telecom networks is not a new idea. It was tried in the early nineties but failed to gain traction. Telcos drivers to deliver IPTV services have intensified. They are finding that the saturation of voice services (both fixed and mobile) are limiting growth opportunities, and that the revenues earned from their voice and data services are dropping as these services become commoditized.

To compound this situation the telcos are faced with intense competition from non-traditional competitors. For example, MSO's now providing voice and data services over their cable infrastructures, MVNO's (e.g. Virgin Mobile) delivering voice services over the telco infrastructure, and even the Mobile companies persuading the population to shift their fixed line services to mobile.

Coupled to the commercial drivers, the technological enablers have improved. Next Generation Core networks have been widely deployed, and there are several Access network technologies now available which are capable of delivering the required bandwidth over the local loop or last mile. These include xDSL, FTTx, WiMAX and HFC.

Compression technologies such as H.264 and VC-1 have further alleviated the bandwidth issues in the local loop and are continuing to evolve, becoming increasingly more efficient at video compression. Trusted Watermarking and DRM technologies are available, providing the content owners with some confidence that their assets will be protected in what is considered an inherently insecure network environment.

IPTV operates in a very complex environment with many opportunities for errors to be introduced. These errors could be introduced at almost any layer in the system, be it the RF layers at ingest, the protocol layers in the network, the Transport Stream (or media layer) of the system, or the physical layer in the network. This requires a range of tools capable of providing early visibility of the fault in order to achieve rapid diagnosis and remedy.

As IPTV moves through the technology deployment lifecycle the test needs change. Initially diagnostic and load test tools are required to prove the technology and "get it working". Beyond this the test requirements evolve towards 24/7 monitoring to assist in the management and operation of the service. The early stages require best in class diagnostic tools, while the later stages require best in class system solutions.

As these systems mature, the focus shifts from QoS as the systems are deployed and commissioned, to QoE during the Operate and Manage phase. QoE will be a critical component of success for IPTV. Traditional broadcast systems have already set the expectations of this experience and there are a number of technical issues within IPTV systems that need to be dealt with to achieve the same levels of QoE. The most obvious is channel change time. The need to provide both this level of QoS and QoE requires visibility of both the User Plane (the video content) and the Control Plane (the signaling layers that provide the control for the subscriber, etc).

IPTV represents the convergence of the broadcast and telecommunications worlds. Successful deployment requires tools and expertise from both worlds. Tektronix provides a wide portfolio of products designed to address the converging world, those products having been derived from our long experience in both Video and Telecommunications test and measurement.

Glossary

Active Fiber to the Premises (Active FTTP) – Active FTTP networks utilize powered electronic equipment in neighborhoods, usually one equipment cabinet for every 400-500 subscribers. This neighborhood equipment performs local switching and routing off loading the central office. See also PON.

Asymmetric Digital Subscriber Line (ADSL, ADSL2, ADSL+, ADSL2+) – ADSL is the most prevalent DSL implementation in North America. The others are enhanced versions of ADSL. ADSL+ is the same as ADSL2+.

Asynchronous Transfer Mode (ATM) – A digital signal protocol for efficient transport of both constant bit rate and bursty information in broadband digital networks.

Bit Error Rate (BER) – Also sometimes referred to as the Bit Error Ratio.

Conditional Access System (CA, CAS) – A system which controls which content can be viewed by the subscriber. For example, it prevents a user from viewing Pay Per View content until the subscriber has agreed to pay for it.

Consumer Premises Equipment (CPE) – A term used in telecommunications that refers to telephones, cable modems or purchased set top boxes utilized by the subscriber to access the network.

Control Plane – The portion of the IPTV network that carries control or signaling information.

Cyclic Redundancy Check (CRC) – Used to produce a checksum that is used to detect errors after transmission or storage.

Digital Rights Management (DRM) – DRM controls subscriber usage of the delivered content, i.e. view once, unlimited view during calendar window, etc.

Digital Subscriber Line Access Multiplexer (DSLAM) – A network device that connects multiple DSLs to a high speed Internet backbone.

Digital Video Broadcasting (DVB-C) – Generally refers to the European initiated consortium of broadcasters, manufacturers, regulatory bodies and others that created standards for delivery of digital television and data services. Includes DVB-C (cable), DVB-H (handheld/mobile), DVB-S (satellite) and DVB-T (Terrestrial) versions.

Digital Video Broadcasting (DVB-H) – Generally refers to the European initiated consortium of broadcasters, manufacturers, regulatory bodies and others that created standards for delivery of digital television and data services. Includes DVB-C (cable), DVB-H (handheld/mobile), DVB-S (satellite) and DVB-T (Terrestrial) versions.

Digital Video Broadcasting (DVB-S) – Generally refers to the European initiated consortium of broadcasters, manufacturers, regulatory bodies and others that created standards for delivery of digital television and data services. Includes DVB-C (cable), DVB-H (handheld/mobile), DVB-S (satellite) and DVB-T (Terrestrial) versions.

Digital Video Broadcasting (DVB-T) – Generally refers to the European initiated consortium of broadcasters, manufacturers, regulatory bodies and others that created standards for delivery of digital television and data services. Includes DVB-C (cable), DVB-H (handheld/mobile), DVB-S (satellite) and DVB-T (Terrestrial) versions.

European Telecommunications Standards Institute (ETSI) – An independent, non-profit standards organization.

Error Vector Magnitude (EVM) – A measure used to quantify the performance of a digital transmitter. EVM is a measure of how far the constellation points defined by a MER calculation are from ideal locations.

Frame Relay – Frame relay is a transmission technique used to digital information from one or many sources to one or many destinations. Commonly implemented for voice and data encapsulation. IP-based networks are replacing Frame Relay in many applications.

Fiber to the (FTTx) – A series of fiber optical network architectures including: FTTP (Fiber to the Premises), FTTC (Fiber to the Curb), FTTH (Fiber to the Home), FTTS (Fiber to the Subscriber), FTTN (Fiber to the Node / Neighborhood), FTTCab (Fiber to the Cabinet),FTTB (Fiber to the Building), FTTEx (Fiber to the Exchange).

Group of Pictures (GOP) – Group of Pictures. In transmission order a GOP starts with an I-picture and ends with the last picture before the next I-picture.

Hybrid Fiber Coax (HFC) – Hybrid Fiber Coax (HFC) systems combine the use of high-speed fiber backbone to deliver data out to the edge of the network, using coaxial cable to run the 'last mile' connecting the subscriber to the backbone.

IEEE 802.16 – An IEEE Working Group on Broadband Wireless Access.

Internet Engineering Task Force (IETF) – Develops and promotes Internet standards.

Internet Group Management Protocol (IGMP) – Protocol used to manage membership in multicast groups.

Media Delay Factor (MDI-DF) – Indicates how long a stream must be buffered to at its nominal bit rate to prevent packet loss. This measurement is not payload aware; it does not separate video traffic from other data and VoIP packets.

Media Loss Rate (MDI-MLR) – The number of packets lost during a period of 1 second.

Modulation Error Ratio (MER) – A measure used to quantify the performance of a digital transmitter. MER is closely related to EVR but is calculated form the average power of the signal.

Middleware – Software and hardware that connects IPTV components together. This includes customer EPG, application control and back office/billing.

Mean Opinion Score (MOS) – A subjective indication expressed in a single number from 1 to 5 of the perceived quality of media.

Multi Protocol Label Switching (MPLS) – A data carrying service for both circuit based clients and packet switching clients. It can be used to carry different kinds of traffic including IP packets.

Multiple Service Operator (MSO) – An operator of multiple cable television services.

Multicast – Simultaneous delivery of information to a group of destinations.

Mobile Virtual Network Operator (MVNO) – A company that resells wireless services using the network of another mobile phone operator.

Pulse Code Modulation (PCM) – A technical term for an analog source waveform, for example audio or video signals expressed as periodic, numeric samples. PCM is an uncompressed digital signal.

Program Clock Reference (PCR) – The sample of the encoder clock count that is sent in the program header to synchronize the decoder clock.

Protocol Independent Multicast (PIM) – A family of multicast routing protocols which provides one-to-many and many-to-many data distribution over the Internet and includes PIM-DM (dense mode), PIM-SM (sparse mode) and PIM-SSM (Single Source Multicast).

Passive Optical Network (PON) – An optical point-to-multipoint access network that uses no optical repeaters or other active devices in the outside plant. PON is the underlying technology behind most FTTH services. There are multiple implementations of PON including ATM PON (APON), BPON (Broadband PON), Gigabit PON (GPON), and Ethernet PON (EPON) which differ in speed, distance, and services which may be deployed. See also Active FTTP.

Picture Quality Analysis (PQA) – Picture Quality Analysis provides repeatable, objective measurements that directly replicate subjective human visual assessments.

Public Switched Telephone Network (PSTN) – The network of the world's public circuit-switched telephone networks. Originally a fixed line analog network it is now almost entirely digital and includes mobile telephones.

Quality of Experience (QoE) – A subjective measure of a customer's experience.

Quality of Service (QoS) – Refers to the capability to prioritize traffic and accord best treatment to delay-sensitive traffic such as voice and video.

Requests for Comments (RFC) – RFC documents encompass new research applicable to Internet technologies. Engineers can publish discourse for peer review or to convey new concepts. RFC4445 describes a proposed Media Delivery Index (MDI).

Real-time Transport Protocol (RTP) – A standardized packet format for delivery video and audio via IP. It is used in multicast and unicast applications.

Real-time Streaming Protocol (RTSP) – Allows a user to remotely control a streaming media server using commands similar to VCR commands, i.e. play and pause.

Session Initiation Protocol (SIP) – A Control Plane protocol for creating, modifying or terminating sessions with one or more participants, i.e. in a multicast application.

Set Top Box (STB) – The Consumer Premises Equipment (CPE) used to interface with the user and the IPTV services provided via the network.

Transmission Control Protocol (TCP) – One of the core protocols if the IP protocol suite. This is the intermediate layer between the IP below it and an application above it.

TR 101 290 – An ETSI Technical report that defines a set of standard evaluation tests for digital video systems.

User Datagram Protocol (UDP) – One of the core protocols of the IP protocol suite.

Unicast – Sending information from a single source to a single destination.

User Plane – The portion of the IP network that carries the data.

Very High Speed Digital Subscriber Line (VDSL, VHDSL) – An xDSL technology providing faster data transmission over single twisted pair.

Video on Demand (VoD) – A system in which television programs or movies are transmitted to a single consumer only when requested.

X.25 – A protocol suite for wide area networks that uses the phone system. X.25 providers are commonly referred to as part of the "packet switched network".

xDSL – See Asymmetric Digital Subscriber Line and Very High Speed Digital Subscriber Line.

Contact Tektronix:

ASEAN / Australasia (65) 6356 3900
Austria +41 52 675 3777
Balkans, Israel, South Africa and other ISE Countries +41 52 675 3777
Belgium 07 81 60166
Brazil & South America (11) 40669400
Canada 1 (800) 661-5625
Central East Europe, Ukraine and the Baltics +41 52 675 3777
Central Europe & Greece +41 52 675 3777
Denmark +45 80 88 1401
Finland +41 52 675 3777
France +33 (0) 1 69 86 81 81
Germany +49 (221) 94 77 400
Hong Kong (852) 2585-6688
India (91) 80-22275577
Italy +39 (02) 25086 1
Japan 81 (3) 6714-3010
Luxembourg +44 (0) 1344 392400
Mexico, Central America & Caribbean 52 (55) 5424700
Middle East, Asia and North Africa +41 52 675 3777
The Netherlands 090 02 021797
Norway 800 16098
People's Republic of China 86 (10) 6235 1230
Poland +41 52 675 3777
Portugal 80 08 12370
Republic of Korea 82 (2) 6917-5000
Russia & CIS +7 (495) 7484900
South Africa +27 11 206 8360
Spain (+34) 901 988 054
Sweden 020 08 80371
Switzerland +41 52 675 3777
Taiwan 886 (2) 2722-9622
United Kingdom & Eire +44 (0) 1344 392400
USA 1 (800) 426-2200
For other areas contact Tektronix, Inc. at: 1 (503) 627-7111
Updated 12 November 2007

For Further Information
Tektronix maintains a comprehensive, constantly expanding collection of application notes, technical briefs and other resources to help engineers working on the cutting edge of technology. Please visit **www.tektronix.com**

1/08 EA/ 25W-20277-1

Tektronix
Enabling Innovation