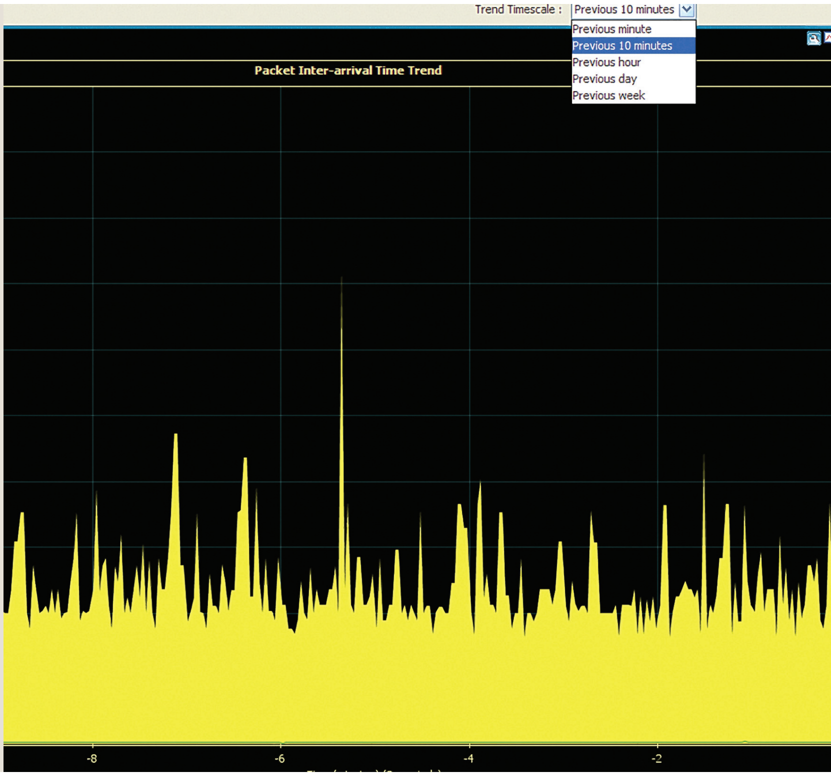
Application Note



Video Over IP Cross Layer Measurements —

Delivering Superior Quality of Service in IPTV Networks

Summary

Along with well known technologies such as MPEG-2 Transport Streams, more recently introduced technologies have accelerated the rollout of IPTV systems across the world. Despite the maturing of these enabling technologies, the deployment of IPTV presents many technical challenges to those required to successfully provide these services. This document explores some of these challenges and how Test and Measurement equipment can be used to facilitate the design, rollout and management of these systems.

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The key operational challenge for telecommunications operators is how to efficiently deliver superior quality of service (QoS) levels to maintain differentiation in this competitive market. There is therefore a requirement to provide an intuitive and simplified presentation of video quality and diagnostic information, to enable delivery of these superior QoS levels in an increasingly complex broadcast environment. In order to achieve these high QoS levels, we need to provide accurate and timely information on system performance to both operations and engineering staff.

Use of test equipment in this environment is essential and correctly placed monitoring probes across the network can provide important data in the form of Key Performance Indicators (KPIs). This empowers operators and engineers to efficiently manage network systems in order to prevent degradation of signal quality which may result in errors which affect the end users experience. Using correctly configured test equipment to perform essential cross-layer monitoring, it is possible to predict system problems long before critical revenue earning services go off the air, rather than cure them after they have happened.

Introduction

Along with well known technologies such as MPEG-2 Transport Streams, more recently introduced technologies have accelerated the rollout of IPTV systems across the world. These include advanced 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 VDSL and ADSL.

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

IP networks provide bi-directional interactive capabilities which traditional TV technologies lack. Theoretically this allows one to one distribution allowing individual viewers control of their chosen content along with 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 feed-back and other services such as online shopping and gaming. The two way nature of these networks enable Video on Demand 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.

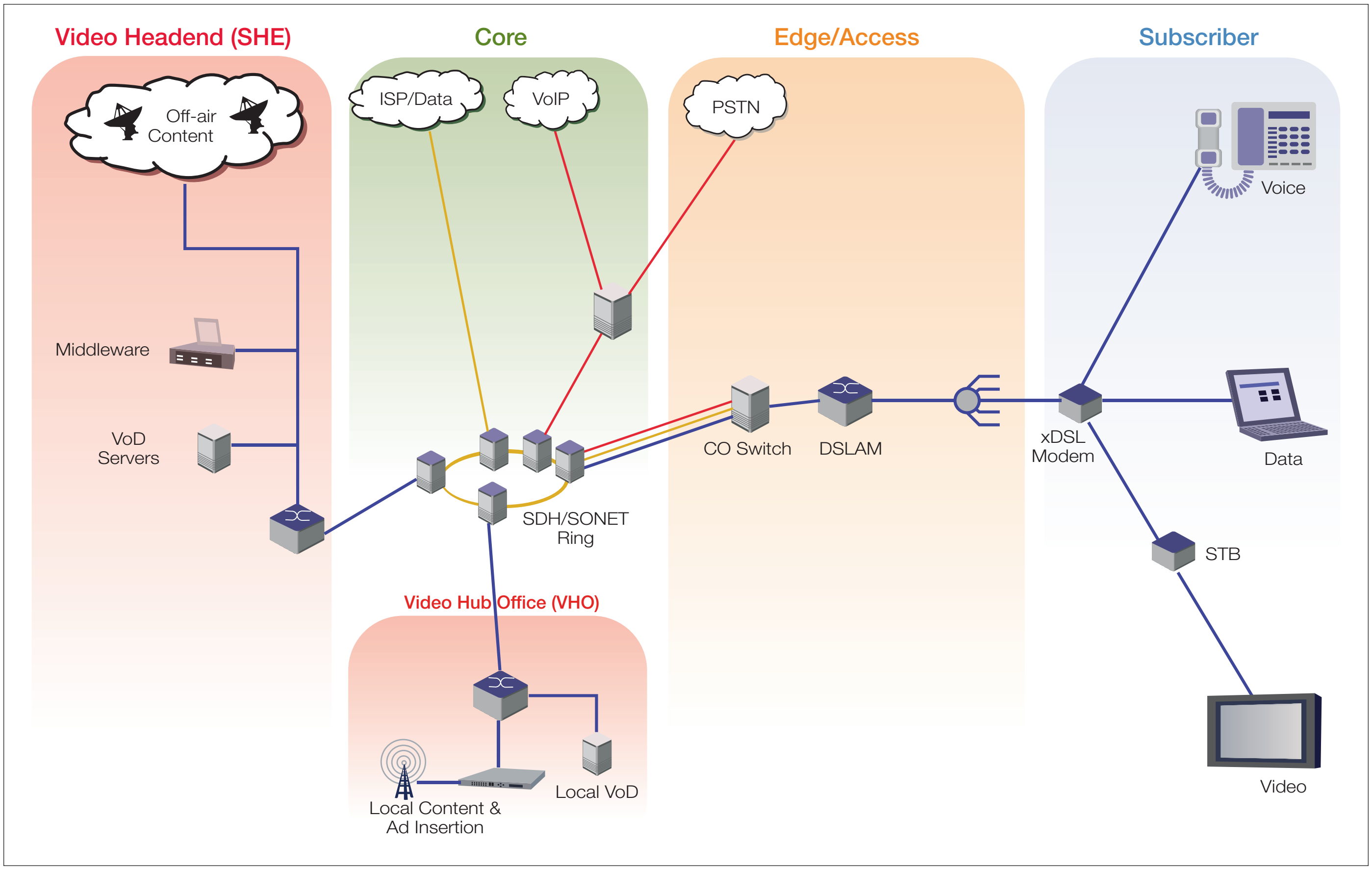


Figure 1. Network Architecture.

Figure 1, Network Architecture shows a schematic layout for a typical architecture for an IPTV based system which includes broadcast video content and VoD services along with both voice and high speed data services.

These linked technologies allow Telco s to balance the demise of their traditional fixed line business by reengineering their existing plant to carry IPTV, High Speed Data and Voice over IP, the so-called Triple Play Services. 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.

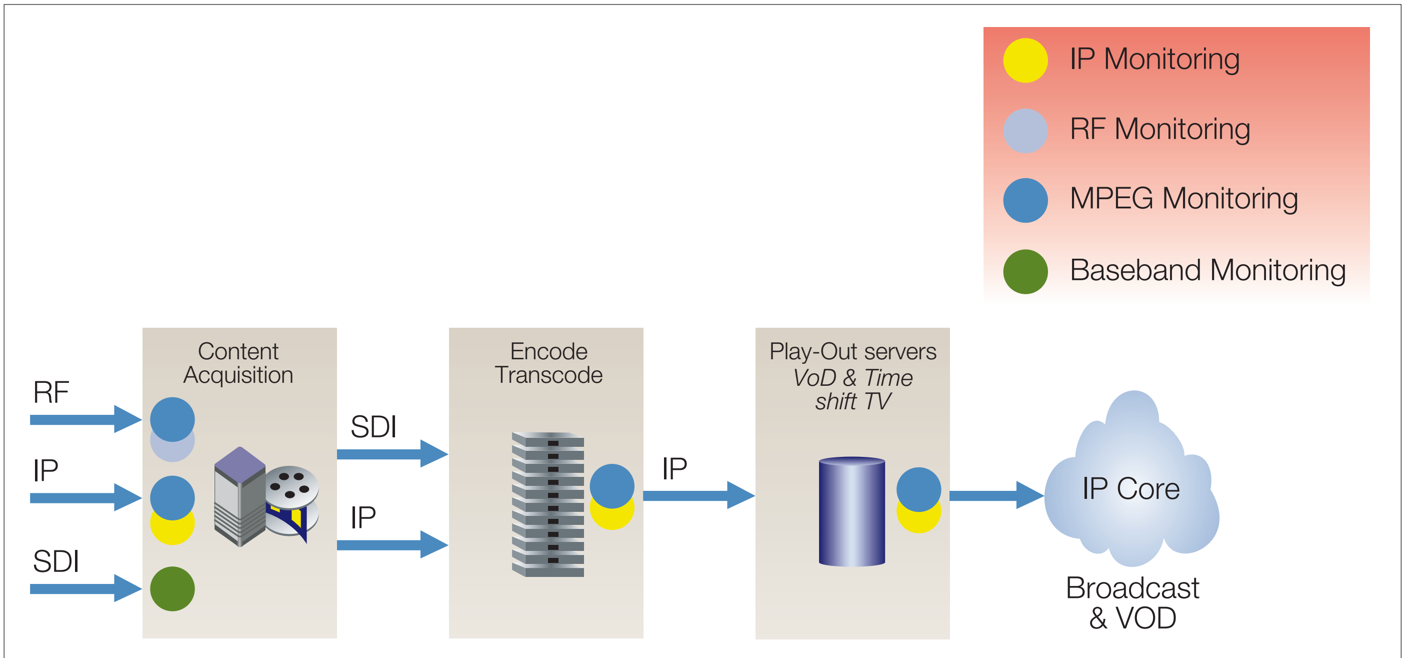


Figure 2. Key Monitoring Points.

IPTV Headend Overview

The focus of this document is the IP headend system. The primary functions of a IP headend are as follows:

Digital program acquisition: content from the satellite or terrestrial sources, and the preparation of that content for digital delivery (National or regional).

Digital program storage: storage and insertion of additional, non-live broadcast programming like local content, video-on-demand or advertising.

Digital program distribution and delivery: encompasses program preparation and aggregation, rate-shaping, modulation, encapsulation (encoding), encryption and other technical process s for program delivery.

In these systems, ingest for the headend can be largely taken from various RF sources, whether they be cable, satellite or off-air terrestrial TV feeds and possible also using SDI or IP feeds. Therefore, the secret to maintaining reliable and high-quality IPTV services when using input formats such as IP, SDI and RF is to focus on critical factors that may compromise the integrity of the system. It is therefore essential to monitor QoS at the ingest before signals are processed through the headend for output onto the Telco network. In order to maintain quality, comprehensive monitoring can be utilized and key monitoring points can be seen in Figure 2, Key Monitoring Points.

Whilst these 3 primary functions can be considered separately in this example, in reality, multiple processes could be undertaken by a single block of hardware. Since a significant proportion of the content inputs to our system are from terrestrial and satellite RF sources, we need to consider how we establish and maintain the quality of these ingested signals. The following section therefore describes those critical RF measurements which help to detect such problems before viewers lose their service and picture completely.

RF signal strength	How much signal is being received
Modulation Error Ratio (MER)	An early indicator of signal degradation, MER is the ratio of the power of the signal to the power of the error vectors, expressed in dB
Error Vector Magnitude (EVM)	EVM is a measurement similar to MER but expressed differently. EVM is the ratio of the amplitude of the RMS error vector to the amplitude of the largest symbol, expressed as a percentage
Bit Error Rate (BER)	BER is a measure of how hard the Forward Error Correction (FEC) has to work BER = Bits corrected / Total bits sent
Transport Error Flag (TEF)	The TEF is an indicator that the FEC is failing to correct all transmission errors TEF is also referred to as “Reed-Solomon uncorrected block counts”
Constellation diagram	Characterizes link and modulator performance

Table 1. Key RF parameters.

RF Ingest — Key Monitoring Parameters

Modern digital TV systems behave quite differently when compared to traditional analogue TV as the signal is subjected to noise, distortion, and interferences along its path. Today s consumers are familiar with simple analogue TV reception. If the picture quality is poor, an indoor antenna can usually be adjusted to get a viewable picture. Even if the picture quality is still poor, and if the program is of enough interest, the viewer will usually continue watching as long as there is sound. Digital TV (DTV) is not this simple. Once reception is lost, the path to recovery isn t always obvious. The problem could be caused by MPEG table errors, or merely from the RF power dropping below the operational threshold or the cliff point. RF problems can include any of the following: satellite dish or Low-Noise Block Converter (LNB) issues terrestrial RF signal reflections, poor noise performance, or channel interference; and cable amplifier or modulator faults. There are a couple of ways to solve DTV reception problems. One solution is to make receivers more tolerant to degraded signals. A better solution is for the network to maintain a clean, high-quality RF signal. Key RF parameters are detailed in Table 1 above.

It is not the intention of this Application Note to give in-depth details regarding RF measurements; however we will look at some specific parameters in order to demonstrate their usefulness in maintaining RF signal quality. If you require more information on this subject, please refer to Tektronix Application Note Critical RF Measurements in Cable, Satellite and Terrestrial DTV Systems # 2TW-17370-1 available from www.tektronix.com.

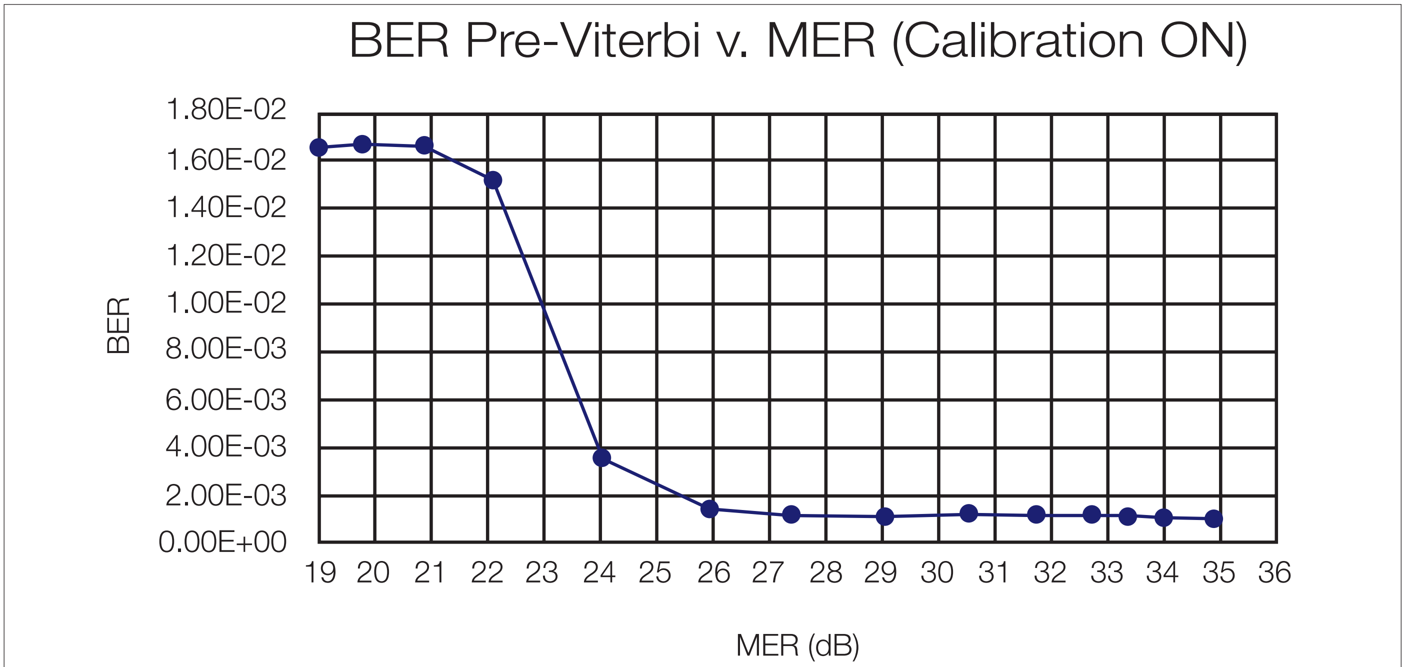


Figure 3. MER/BER.

For ingest monitoring, it has already been stated that good signal quality at the headend input is critical. Using some or all of the above measurement parameters, operators and engineering can ensure that high levels of quality are maintained. Use of the right measurement equipment is therefore essential. These key parameters can be used not only to find problems that exist right now, but to pro-actively manage the system in order to prevent service affecting degradation.

The TR 101 290 standard describes measurement guidelines for DVB systems. One measurement, Modulation Error Ratio (MER), is designed to provide a single figure of merit of the received signal. MER is intended to give an early indication of the ability of the receiver to correctly decode the transmitted signal. In effect, MER compares the actual location of a received symbol (as representing a digital value in the modulation scheme) to its ideal location. As the signal degrades, the received symbols are located further from their ideal locations and the measured MER value will decrease. Ultimately the symbols will be incorrectly interpreted, and the Bit Error Rate (BER) will rise; this is the threshold or cliff point.

Figure 3, MER/BER shows a graph, which was obtained by connecting the MER receiver to a test modulator. Noise was then gradually introduced and the MER and pre-Viterbi BER values recorded. With no additive noise, the MER starts at 35 dB with the BER near zero. Note that as noise is increased the MER gradually decreases, while the BER stays constant. When the MER reaches 26 dB, the BER starts to climb, indicating the cliff point is near. MER indicates progressive system degradation long before reaching the cliff point.

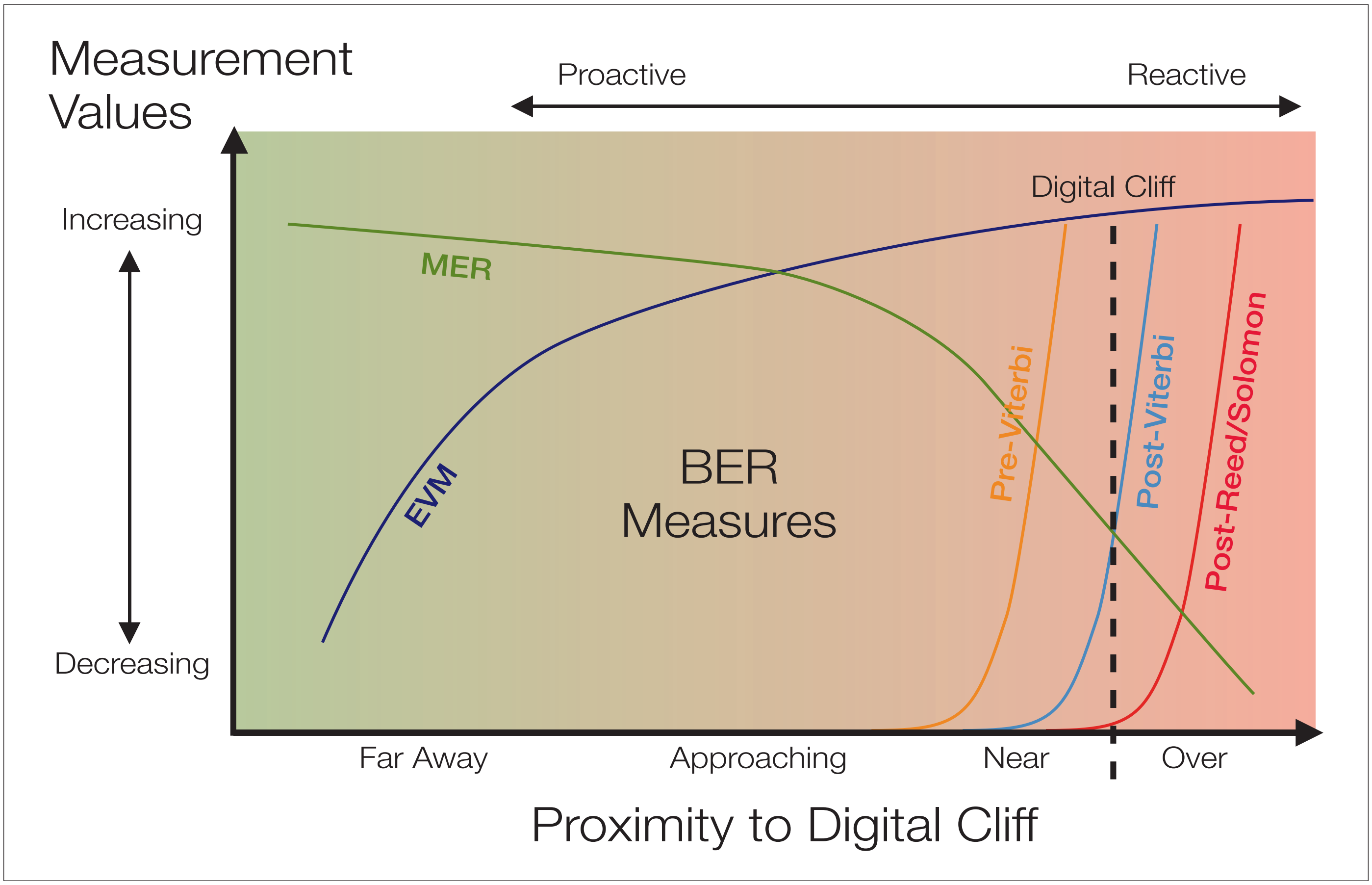


Figure 4. The Digital Cliff.

In practical terms, this means is that monitoring Bit Error Rate alone would not give any indication of impending problems as noise increases in the 26 - 35 dB MER region. In this case, use of MER measurement can provide valuable insight to the signal degradation before it is customer impacting. Looking at Figure 4, the Digital Cliff it can be seen more clearly that falling MER and increasing Error Vector Magnitude can provide early indication of signal degradation. It can be seen that as MER levels decrease and EVM increases, the signal can be seen to degrade well before Forward Error Correction (FEC) is required. As the signal becomes noisier, MER decreases to such a point that FEC can no longer correct errors and BER levels start to rise, eventually leading to Transport Error Flags and MPEG Continuity Counts errors, which in turn leads to visible video artifacts. Once we get to the point where the FEC is overwhelmed, the digital cliff is hit, the signal is corrupted and customers start to pick up their phones.

It is worth noting that the requirement to deliver a competitive grade of digital video service requires that the access network be designed to deliver a Bit Error Rate of at least 10^{-9}. Even at this high rate, the end user will still see a visible picture artifact every 6 minutes on a 3 Mbps SD transport stream. It is in everyone s interests to prevent this happening and proactive monitoring can help.

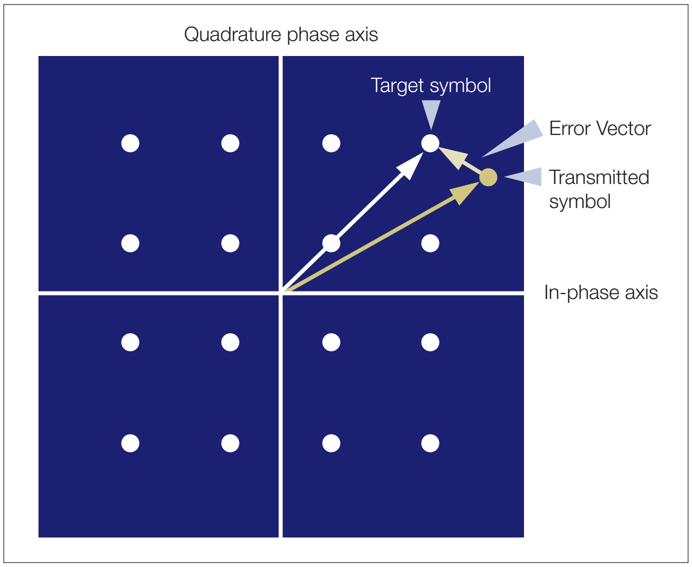


Figure 5. EVM.

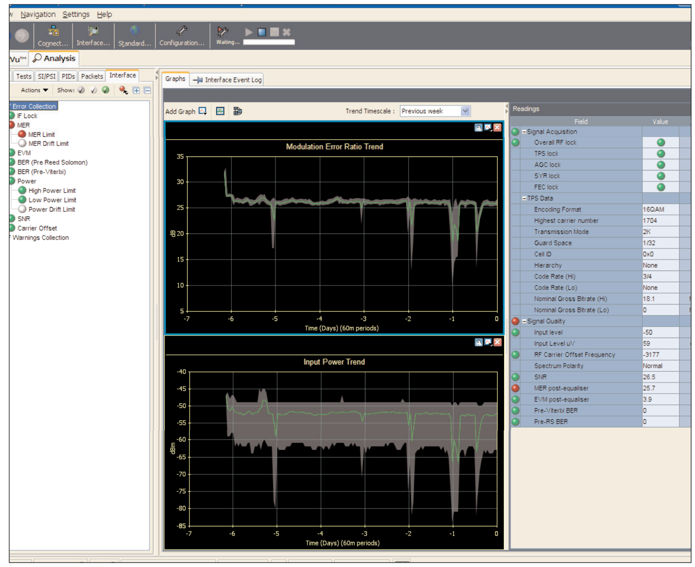


Figure 6. Dual Level Alarms & Trending.

It has been demonstrated that MER and EVM measurement can be very useful in maintaining RF signal quality, but there are also other methods for engineers to get a health check on their RF signals. A more visual method is the Constellation Diagram.

Ideally, a transmitted signal would show all constellation points precisely at their ideal locations; however imperfections in the system (for instance, excessive phase noise) can cause the actual constellation points to deviate from their calculated ideal locations. EVM is therefore a measure of how far the points are from the ideal locations. It can be seen from Figure 5, EVM that measurement of both EVM and MER can be very useful in predicting quality issues on the incoming RF signals.

Absolute measurement of EVM and MER values is one method of seeing what is happening to your signal quality any point in time. However, the ability to measure the rate of change of these values can give more accurate prediction of any degradation in signal quality. The ability to measure the absolute values and the rate of change of those values allows operators to pro-actively manage the systems and therefore prevent signal quality dropping to such a level that bit errors start to occur. This dual level measurement and alarming is key, and coupled with the ability to provide trending of this data over periods up to 7 days, is a powerful tool in aiding operators and engineers in maintaining signal quality. An example of this is shown at Figure 6, Dual Level Alarms and Trending.

It is better to predict system problems long before critical revenue earning services go off the air, rather than cure them after they have happened. MER measurements are able to measure small changes in transmitter and system performance and are one of the best single figures-of-merit for any RF transmission system. EVM and more traditional BER are useful for standard cross-equipment checks and as an aid to identify short-term signal degradation.

Constellation displays help provide a reliable health check for RF transmission systems by indicating artifacts, distortion, or equipment drift. By combining these critical RF measurements with comprehensive MPEG transport stream monitoring and alarming in a single probe, system problems can be detected at an early stage, before viewers are affected. With the MTM400A, Tektronix is able to provide all the critical RF measurements and interfaces, integrated with MPEG measurements in a single cost-effective monitoring probe.

IP Broadcast Output — Key Monitoring Parameters

Hopefully, with efficient use of monitoring equipment at the Telco headend ingest, we can now establish and maintain RF signal quality into the plant. We now need to look at how we maintain that signal quality as the multi-program transport streams are de-multiplexed and groomed into single program transport streams into the headend output over the core network.

The key operational challenge for telecommunications operators is how to efficiently deliver superior quality of service (QoS) levels to maintain differentiation in this competitive market. There is therefore a requirement to provide an intuitive and simplified presentation of video quality and diagnostic information, to enable delivery of these superior QoS levels in an increasingly complex broadcast environment. In order to achieve these high QoS levels, we need to provide accurate and timely information on system performance to both operations and engineering staff.

We have established that is may be necessary to improve our systems QoS in order to maintain high performance levels and there reduce customer churn. The question is, what do we really mean when we say Quality of Service in an IP environment?

What is QoS ?

Quality of Service, or QoS, in the field of telephony, was defined in the ITU standard X.902 as "A set of quality requirements on the collective behavior of one or more objects. In network traffic engineering, QoS can be used provide various priorities to differing data flows, or guarantee a certain level of performance to a data flow. In IPTV systems, this prioritization is critical to achieve good quality video delivery.

According to a Cisco Whitepaper from 2006 Quality of Service (QoS) refers to the capability of a network to provide better service to selected network traffic over various technologies, including Frame Relay, Asynchronous Transfer Mode (ATM), Ethernet and 802.1 networks, SONET, and IP-routed networks that may use any or all of these underlying technologies. [1]

The primary goal of QoS is to provide priority including dedicated bandwidth, controlled jitter and latency .. and improved loss characteristics. [1]

Key Performance Indicators

Expanding on this definition, it becomes apparent that QoS therefore refers to the ability of a service provider to support users requirements with regard to at least 4 service categories;

- Bandwidth
- Latency or delay
- Jitter
- Traffic Loss

Looking at these 4 categories in more detail, they can be further defined as;

- Bandwidth
  —The network should be able to sustain sufficient capacity to support the users throughput requirements.
- Latency or Delay
  —The time taken to send any packet from a given transmit node to a given receive node.
- Jitter
  —The variation in the delay between the arrival of packets at the receive node.
- Traffic or Packet Loss
  —How often are packets lost?
  —How many packets are affected?

These 4 items can be considered as KPI s to be used in measuring the performance of the system. So now we know what we can measure, the question would be, why do we need to measure?

IPTV systems are run on best effort networks because the IPv4 and IPv6 protocols are, by definition, best effort delivery systems. Both protocols rely on other supporting protocols such as TCP, in order to provide QoS services to the user and the simple fact is that data and voice services can normally cope with jitter and delays, video cannot. Video is normally carried over UDP/RTP and requires high availability (in bandwidth and time) which requires implementation of robust network management policies. It cannot be successfully delivered by a best effort network where IP packets carrying video do not arrive on time and in the right order. This therefore brings us to how we can efficiently measure what is happening in our IPTV delivery network.

Referring back to Figure 2, Key Monitoring Points above, we now need to consider what is being output to the Core Network, so we must use suitably placed monitoring points. In any environment, monitoring probe placement should be non-service impacting. This can be ensured on IP feeds by placing probes on router/switch mirror ports. A mirror port is a passive method which gives equivalent measurement to in-line measurement. In most cases, in-line measurement is not recommended as it can be potentially service impacting, as the monitoring probe is effectively part of the broadcast chain.

Dependent on the size of the network, it may be prudent to place probes at the IP output onto the core network, and then at any access network inputs. This gives comprehensive monitoring of the headend output and the output from the core network/input to the access network. Using this methodology, any degradation of the IP feed caused by the core network can be identified quickly by comparing the probe at the headend output to the probe at the access network input. This can be achieved by having the monitoring probes connected to a overall Network Management System which could also control some or all of the headend acquisition and Transport Stream processing systems. In this way, a system-wide view is possible and therefore, by using pre-emptive techniques highlighted above in the RF section, errors can be identified and remedied before they become customer impacting.

We have already categorized the 4 main KPIs which can be used to monitor QoS of the system. It could be argued that bandwidth is something that is managed during system design as suitable provisioning and traffic management policies should be designed into the network at the outset. Operation measurements such as the number of sessions present at any network node and the bandwidth of each or all those sessions can be useful as indicators of potential system overload. Any overload could be predicted by monitoring other parameters which could be symptomatic of provisioning issues. These could include the 3 other KPIs mentioned above.

The first 2 are inextricably linked — Latency/Delay and IP Jitter should be monitored across all sessions on the link. As an example, let us consider a single session of 4.7 Mbps carrying a single Transport Stream:

Assume 7 MPEG packets per frame =
7 x 188 bytes = 1316 bytes

Ethernet header	14 bytes
IP header	20 bytes
UDP header	8 bytes
RTP header	12 bytes
Total	54 bytes overhead + 1316 byte payload (4.1% overhead)

Assume the Ethernet frame has IP/UDP/RTP encapsulation

Therefore the Ethernet frame size is 1370 bytes which gives an Ethernet flow rate of 4.886 Mbps:

Ethernet flow rate (Ethernet frames per second) = 4,886,000 [bitrate] / (8 [bits per byte] * 1370 [bytes per frame]) = 445.80 frames per second

The interval between frames is 1 / 445.8 [Ethernet frame rate] = 0.00224 seconds

The ideal packet arrival time should therefore be 2.24 mS. Any variation away from this ideal could cause buffer issues on any receiving device. A fixed variation could be an issue, but variable timing between IP packets, otherwise know as IP Jitter, can cause major issues if not diagnosed and rectified. The effects of packet jitter on the end user can be variable as network design elements such as router buffers sizes and consumer equipment design can have significant effects on the perceived QoS. Consumer set top boxes designed with large input buffers can largely negate most network jitter effects but that improved design will almost certainly come at a great cost than less tolerant designs. It is therefore preferable to be able to measure and counteract any excessive IP packet jitter in the network. A measurement probe should be capable of measuring and displaying Packet Inter-arrival Times (PIT) over extended periods to ensure that no underlying issue is degrading to a point where a customer affecting situation arises.

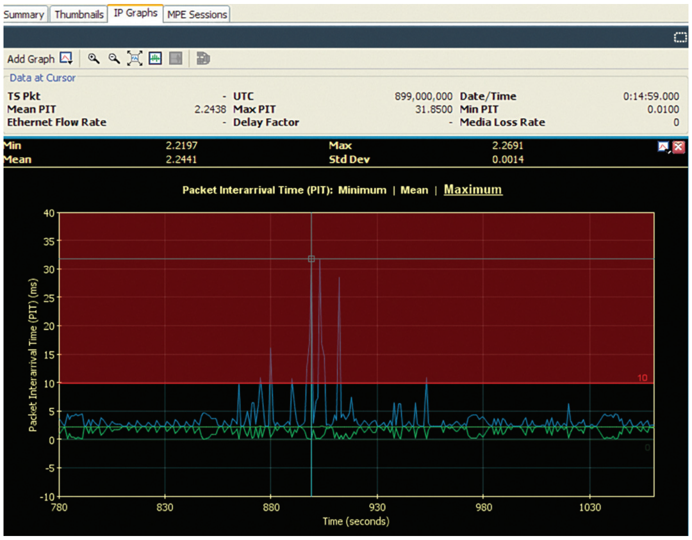


Figure 7. Packet Interarrival Time Graph.

The ability to accurately measure and display PIT is one of the keys to good network diagnostics and management and an example is shown at Figure 7, Packet Interarrival Time Graph.

It can be seen that limits (red area) have been set on the monitoring probe and this limit has been exceeded at several points over the recent past. A simple graphical display gives the operators or engineers a fast, convenient method to get important QoS details and act accordingly.

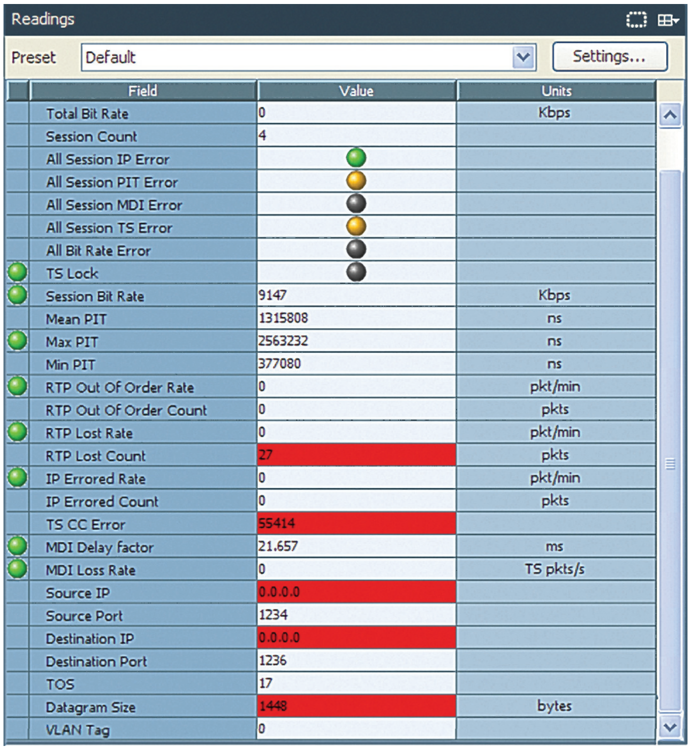
Cross Layer Timing Issues

There is also an associated, but sometimes overlooked effect of carrying an MPEG stream over an IP network The transport stream packets are packetized into IP packets (more specifically, UDP or RTP over UDP) which is nominally seven TS packets per IP packet. As this IP packet is processed, it has the effect that all seven TS packets arrive at the same time into the MPEG decoder buffer. Since the TS packets are given the same timestamp on arrival, at the buffer input, the timestamp for any PCR carrying packets will be wrong, therefore affecting the PCR timing measurements.

PCR accuracy (PCR_AC) is independent of arrival time, so will be unaffected, but PCR drift rate (PCR_DR) , frequency offset (PCR_FO) and overall jitter (PCR_OJ) do depend on arrival time. Video over IP decoders have a buffered output of the IP packets into the MPEG decoder, which restores the constant bit rate. This can be emulated within the Tektronix MPEG TS Compliance Analyser by switching PCR Interpolation on, which spreads the TS packets inside an IP packet over the distance between the two IP packets.

It is also worth noting that none of the PCR measurements will work on a variable bit rate (VBR) stream. This is because the time of arrival of the TS packet cannot be reconstructed without further timing information for each TS packet.

Even maintaining correct PCR timing may not be enough to ensure good video quality. Whilst the system time clock can be synchronized from encoder to decoder by the PCRs, frame synchronization is typically accomplished through the Presentation Time Stamp (PTS) inserted in the Packet Elementary Stream (PES) header. The PTS is a 33-bit value in units of 90 kHz, (27MHz clock divided by 300.) The PTS value indicates the time that the frame should be presented by the decoder. Since the PCR and PTS/DTS, keep the encoder and decoder in synchronization, any variation between these PCR and PTS values can cause buffer underflow or overflow issues, thereby causing decoding problems such as color loss, obviously visible to the viewer. Cross layer, time correlated timing measurements such as PIT, PCR and PTS timing can therefore prove valuable in tracing systematic timing problems.



Readings

Preset: Default | Settings...

Field	Value	Units
Total Bit Rate	0	Kbps
Session Count	4	
All Session IP Error		
All Session PIT Error		
All Session MDI Error		
All Session TS Error		
All Bit Rate Error		
TS Lock		
Session Bit Rate	9147	Kbps
Mean PIT	1315808	ns
Max PIT	2563232	ns
Min PIT	377080	ns
RTP Out Of Order Rate	0	pkt/min
RTP Out Of Order Count	0	pkts
RTP Lost Rate	0	pkt/min
RTP Lost Count	27	pkts
IP Errored Rate	0	pkt/min
IP Errored Count	0	pkts
TS CC Error	55414	
MDI Delay factor	21.657	ms
MDI Loss Rate	0	TS pkts/s
Source IP	0.0.0.0	
Source Port	1234	
Destination IP	0.0.0.0	
Destination Port	1236	
TOS	17	
Datagram Size	1448	bytes
VLAN Tag	0	

Figure 8. RTP packet statistics.

The next KPI to consider is packet loss. We have mentioned buffering issues at the router previously and it is these buffer issues on the output ports of network routers that can cause packet loss. If a router at a network aggregation point gets near its maximum input capacity, there may be packet losses at the output interface as the routers buffers reach overflow. This may not be an instantaneous event, but may be an effect of a gradual increase in traffic, maybe at a point in the early evening where prime time TV comes online. If suitable traffic management and provisioning has not taken place or has been overwhelmed, packet losses could proliferate through the network and result in poor end user experience. It is therefore an essential feature of a network monitoring system to be able to detect both lost or out-of-order packets. This is shown in Figure 8, RTP packet statistics below. Network induced delays may result in out-of-order packets but the end user effect could again be negated by consumer equipment design, with larger buffer sizes giving the set top box time to re-order packets Nevertheless, monitoring equipment should have the capability to detect the out-of-order events and provide timely diagnostic information to operators and engineers in order that the situation can be isolated and rectified before customers complain.

Media Delivery Index

Both packet delay and packet loss have been taken into account by IETF RFC 4445. This RFC describes Media Delivery Index (MDI) and it is defined as a single figure of merit used to quantify 2 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.
- The Media Loss Rate is the number of packets lost during a period of 1 second.

Whilst MDI has been broadly accepted by the industry as the de-facto measurement for packet delay and loss, it is not without issue. One key issue is that MDI does not take into account the payload of the IP packet it measures. Therefore it treats audio, data and video in the same way. This comes to the fore when basic UDP (i.e. not RTP) traffic is being carried. Raw UDP protocol does not provide any means to detect packet loss. So for raw UDP, the packet loss portion of MDI has to be extrapolated from MPEG Continuity Count errors. Therefore any other error, such as Transport Stream syntax errors, cannot be detected by MDI.

The MDI Delay Factor (MDI-DF) is transport stream bit rate based, derived from the Transport Streams Program Clock References (PCRs) and is used to measure packet jitter on the network. However this relies on accurate PCR values, which may not be the case. Therefore, a bad PCR from a multiplexer could trigger an MDI error even though there is no network issue. It is therefore important to consider that, a good MDI does not mean a faultless IP transmission, and a bad MDI can be the result of non-IP related issues. MDI is not the answer — it simply complements other measurements.

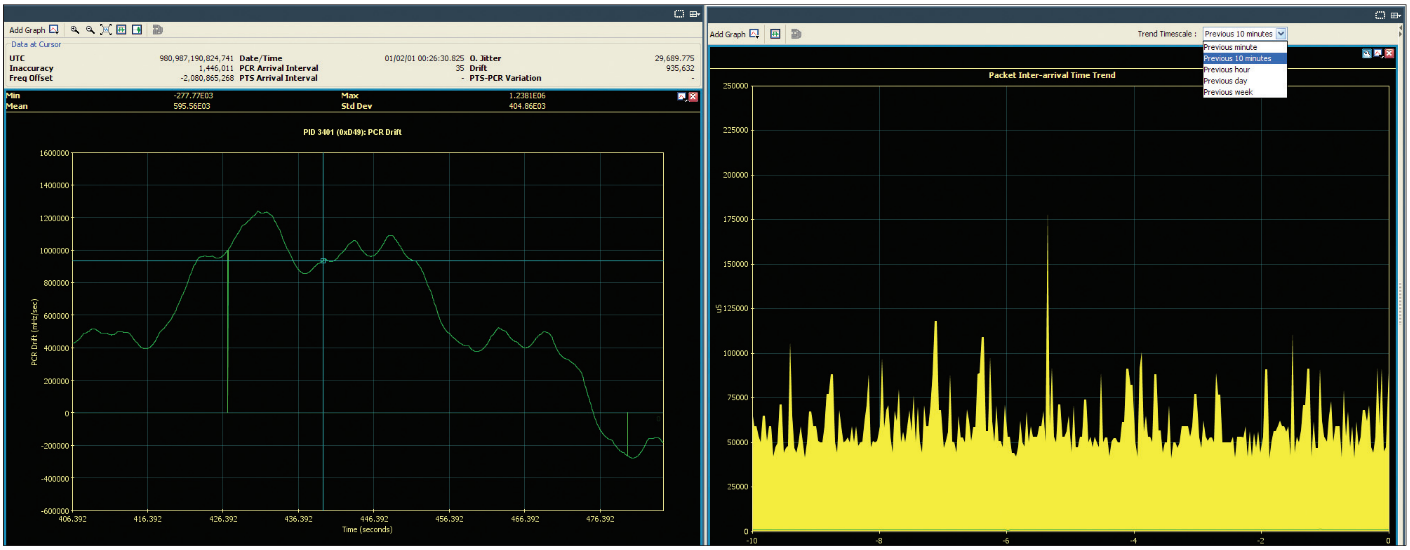


Figure 9. Cross Layer Measurements.

Cross Layer Measurement — Case Study

In terms of total QoS measurement, it is important to consider all aspects of the content being carried, whether it is Transport Stream over RF or IP carriers. The ability to monitor and measure across all these layers should be part of an overall monitoring strategy.

One example of how useful these cross layer measurements can be was shown when a customer reported intermittent set top box drop-outs on their IPTV system. Further analysis showed that only a certain model of STB was affected. Deeper investigation then proved that the cause was a combination of the 27MHz PCR clock drift (PCR_DR measurement) rapidly changing due to an overworked multiplexer and at the same time, the Packet Network entering a period of extreme delay (Packet Inter-arrival Time or PIT measurement) at the edge of the network - the STB would lose lock.

The STB was being affected due to an already rapidly changing PCR_DR and sudden network delay artificially speeded up the PCR_DR even more, causing these customer affecting STB issues. Independently these PCR_DR and PIT problems could be handled by the STB, but not simultaneously. In-depth PCR analysis with graphical results views enable high accuracy timing and jitter measurements to be made to ensure correct operation of the network. These measurements are shown in Figure 9, Cross Layer Measurements.

The ability to carry both IP and MPEG layer measurements on all sessions simultaneously from a single probe is a very powerful additional to the engineers toolkit whether they are maintaining system QoS diagnosing system problems. Using multiple probes connected across the system, from ingest to the access networks can give operators and engineers the ability to access key information to enable signal quality to be maintained, along with the ability to respond quickly and diagnose a system failure.

Conclusions

It is clear that carrying high quality digital video across IP networks is a challenging task. Differentiated IP services such as High Speed Data, VoIP and video all high differing bandwidth and QoS requirements. Video requires high availability (in bandwidth and time) which requires implementation of robust network management policies, along with suitable monitoring tools to ensure those policies are maintained 24/7. It has been shown that IP video cannot survive in a Best Effort environment - video packets need to arrive in sequence and with no losses.

Use of test equipment in this environment is essential and correctly placed monitoring probes across the network can provide important data in the form of KPIs. This empowers operators and engineers to efficiently manage network systems in order to prevent degradation of signal quality which may result in errors which affect the end users experience.

The Tektronix MTM400A Transport Stream Monitor provides a complete solution for real-time transmission monitoring of MPEG transport streams over RF, IP, and ASI interfaces. It combines powerful confidence monitoring capability with deep diagnostic measurements within a single integrated solution. Its FlexVuPlus“ User Interface uniquely presents simplified presentation of video quality and diagnostic information, to enable delivery of superior QoS levels in an increasing complex broadcast environment.

References

1. Cisco Systems. 2006. Quality of Service. Available at (http://www.cisco.com/univercd/cc/td/doc/cisintwk/ito_doc/qos.htm) [Accessed September 2007]

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For Further Information

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10/09 JS/WOW 2AW-21920-1

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