Delivering Superior Quality of Service in IPTV Networks

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Agenda

- Overview – What is enabling IPTV and Video over IP ?
- IPTV System - Headend Ingest
  - Key Monitoring Points
  - RF Ingest – What should we monitor ?
  - Maintaining Signal Quality
- IPTV System – Broadcast Outputs
  - Overview
  - What is QoS ?
  - The Issues
  - Cross Layer Timing and how it helps….
- Conclusions
- Question & Answer

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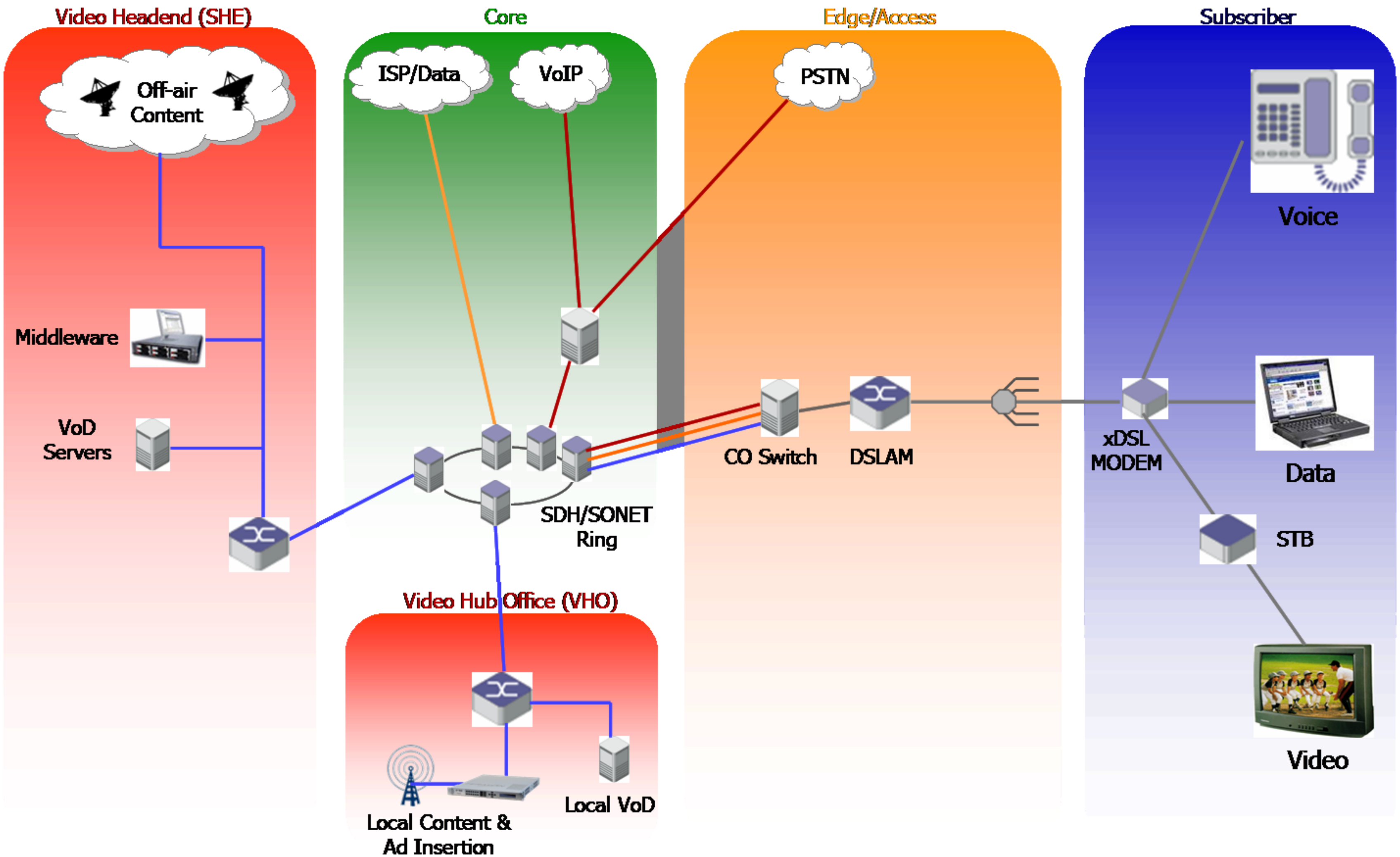
Overview

- Along with well known technologies such as MPEG-2 Transport Streams (ISO/IEC 13818-1), more recently introduced technologies have accelerated the rollout of IPTV systems across the world.
- These include advanced compression technologies like H.264/AVC (ISO 14496-10) and VC-1 (SMPTE 421M) allowing more efficient use of the limited bandwidth links to the home.
- Despite the maturing of these enabling technologies, the deployment of IPTV presents many technical challenges to those required to successfully provide these services.
- This presentation explores some of these challenges and how Test and Measurement equipment can be used to facilitate the design, rollout and management of these systems.

Typical Network Architecture



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IPTV System - Headend Ingest

IPTV Headend Overview …

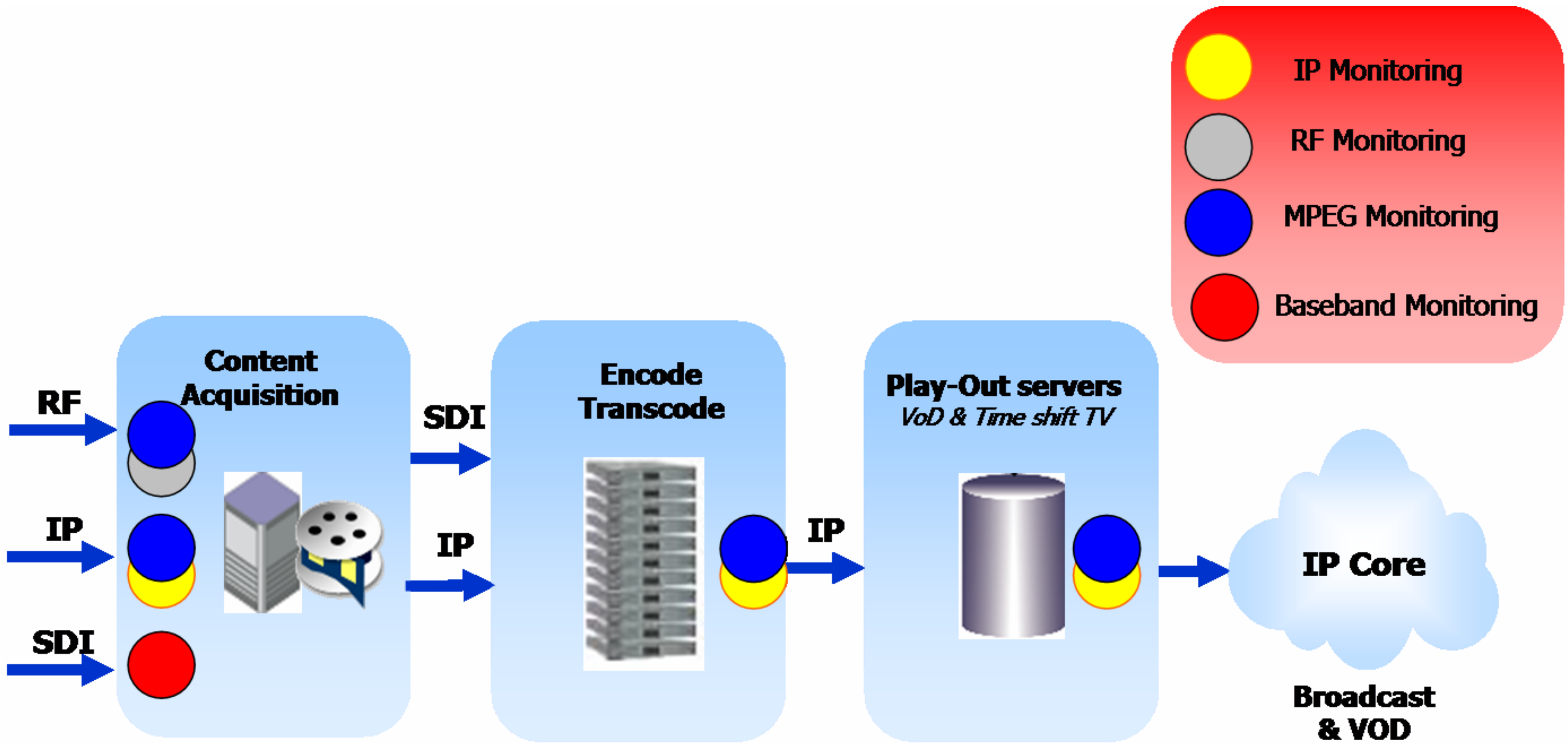
- The primary functions of a IP headend are as follows:

- **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, encapsulation (encoding), encryption and other technical processes for program delivery

IPTV Headend Overview (cont.)

- Ingest for the headend can be largely taken from various RF sources, whether they be 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 Quality of Service (**QoS**) at the ingest before signals are processed through the headend for output onto the Telco network.
- In order to maintain quality, comprehensive monitoring throughout the network should be utilized.

IPTV Headend – Key Monitoring Points



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.
- 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.
- It is therefore important for the network to maintain a clean, high-quality RF signal. This can be done by looking at key parameters in order to maintain signal quality.

RF Ingest – Key Monitoring Parameters

RF signal strength	How much signal is being received
Modulation Error Ratio (MER)	A early indication 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 = Transmitted bits corrected / Total bits transmitted
Transport Error Flag (TEF)	The TEF is an indicator that the Forward Error Correction (FEC) is failing to correct all transmission errors in a 188 byte packet.
Constellation diagram	Characterizes link and modulator performance

RF Ingest – Key Monitoring Parameters

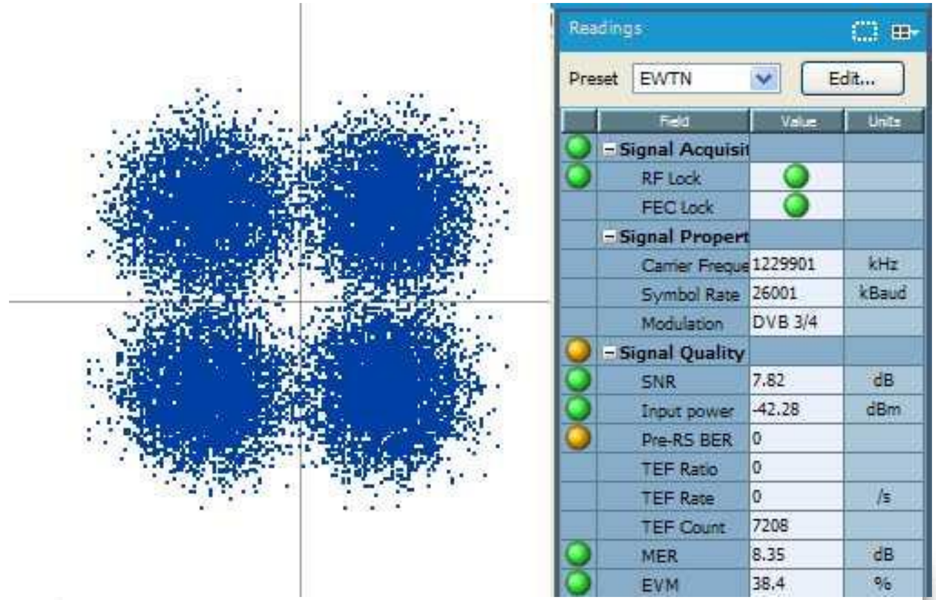
- MER – Modulation Error Ratio
  - Provides a “Single Figure of Merit” for received signal
  - Ratio of “Ideal signal vectors” to “Errored signal vectors” in dBs
  - Similar to SNR measurement
  - Defined in TR 101 290

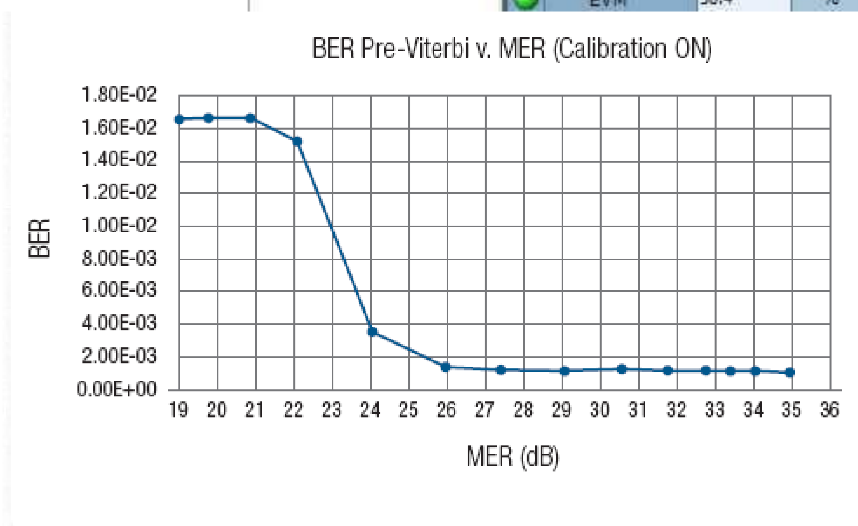
- TR 101 290 expresses preference for MER

- EVM – Error Vector Magnitude
  - Closely related to, and can normally be calculated from MER
  - Ratio of “Errored signal vectors” to max. signal amplitude in %
  - Defined in annex of TR 101 290

RF Ingest – Key Monitoring Parameters: MER

- 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.
- Noise is gradually introduced and the MER and pre-Viterbi BER values recorded.
- 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 **<u>cliff</u>** point is near.





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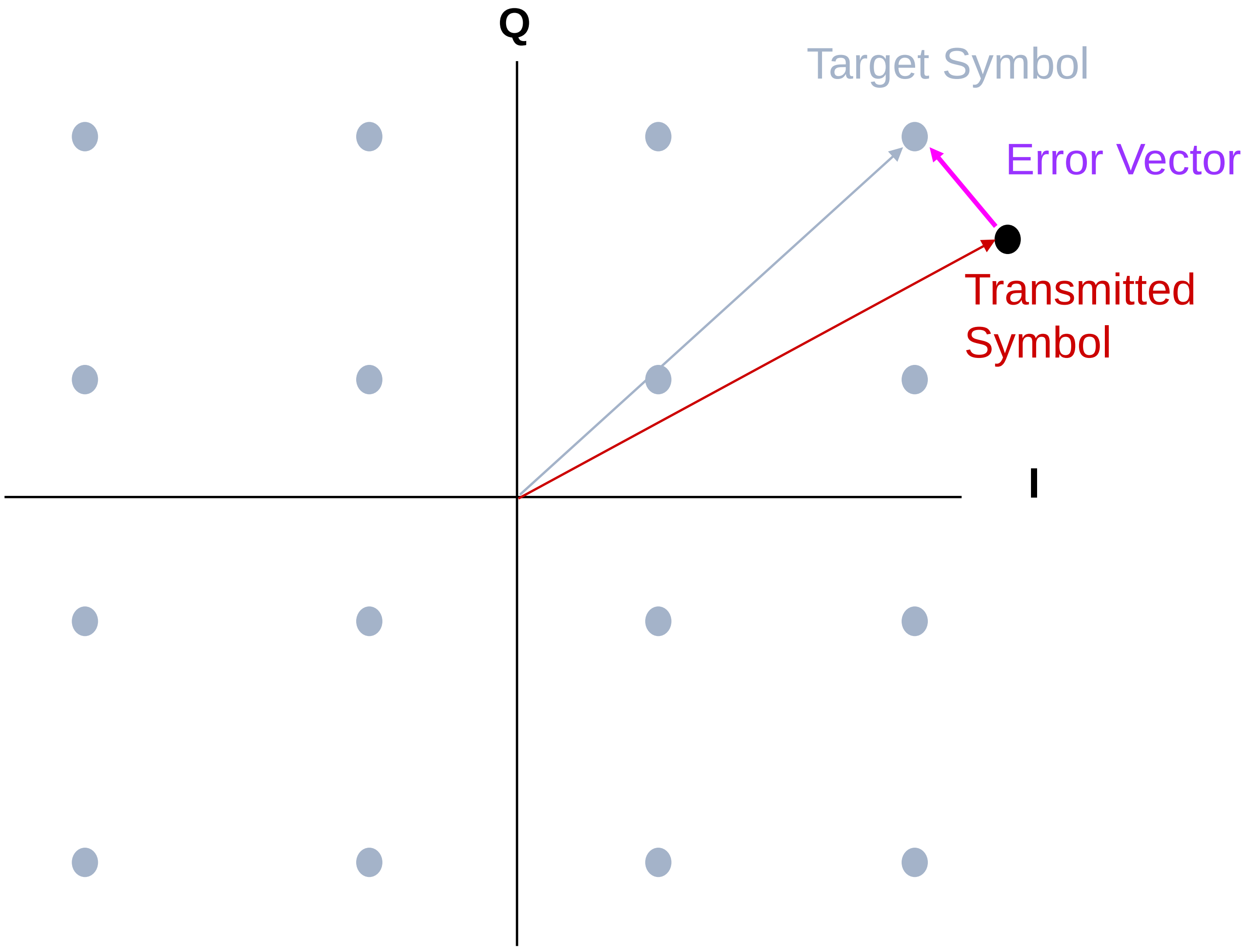
RF Ingest – Key Monitoring Parameters: EVM



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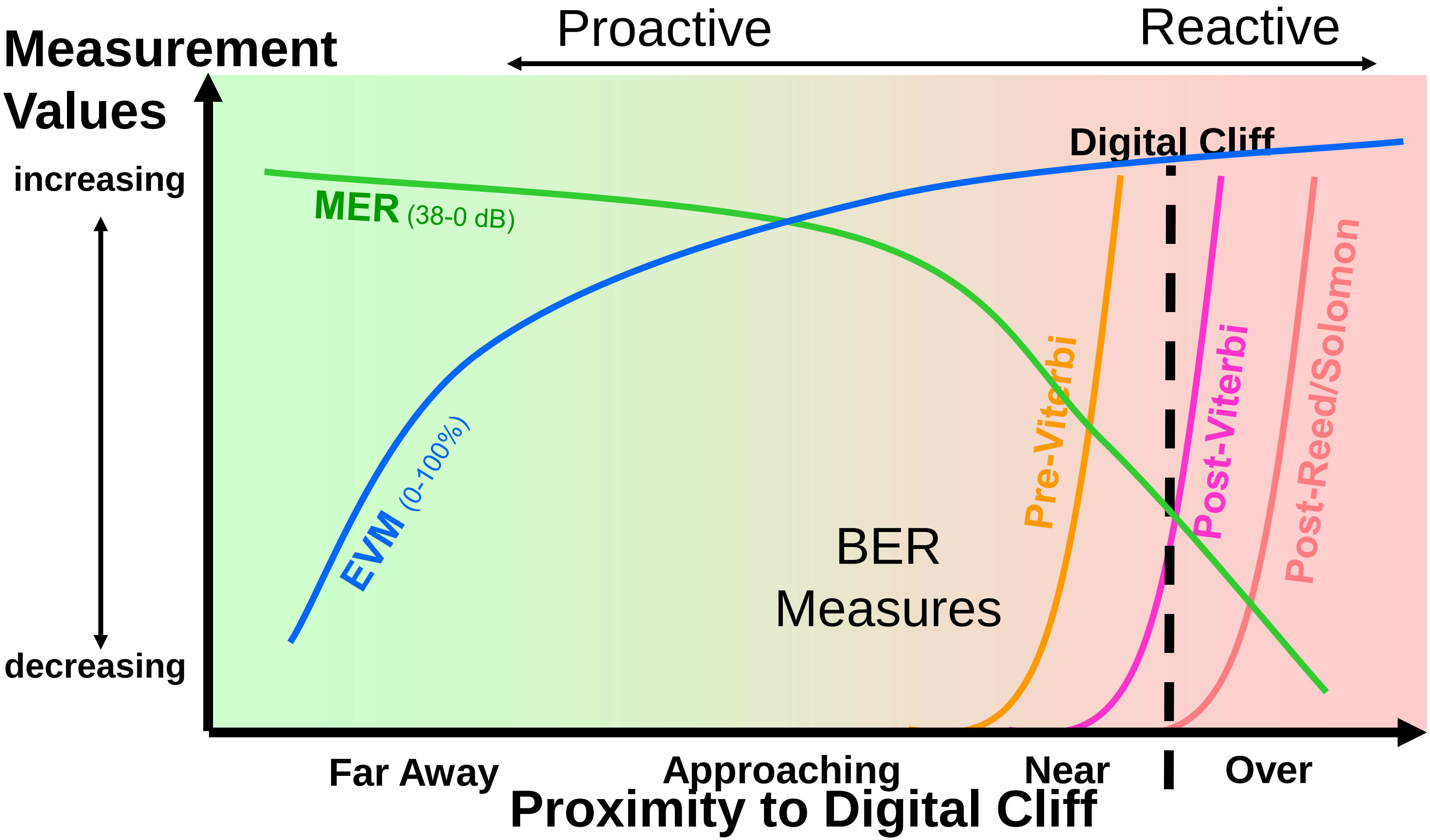
RF Ingest – Key Monitoring Parameters: MER Vs EVM



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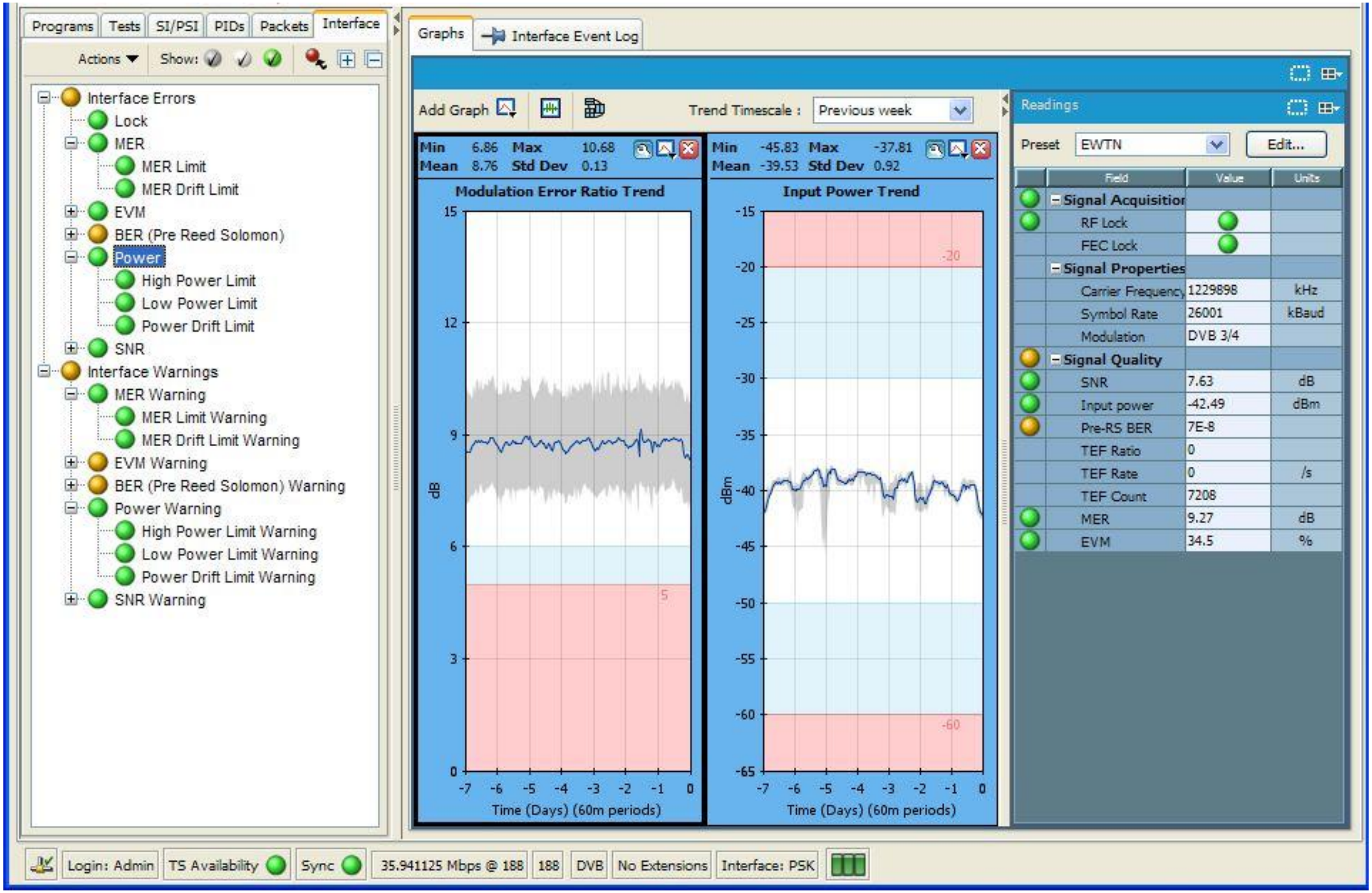
Key Monitoring Parameters – RF Trending

Recent Telco examples: Terrestrial Interference



RF Ingest – Key Monitoring Parameters

- 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.
- 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.
- Proactive measurement such as rate of change measurements of MER & EVM can provide better Quality of Service and reduce outages.

IPTV System – Broadcast Output

Overview

- The key operational challenge for IPTV system operators is how to efficiently deliver superior Quality of Service (QoS) levels to maintain differentiation in this competitive market.
- 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.
- 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.
- '*The primary goal of QoS is to provide priority including dedicated bandwidth, controlled jitter and latency….. and improved loss characteristics*.' [1]

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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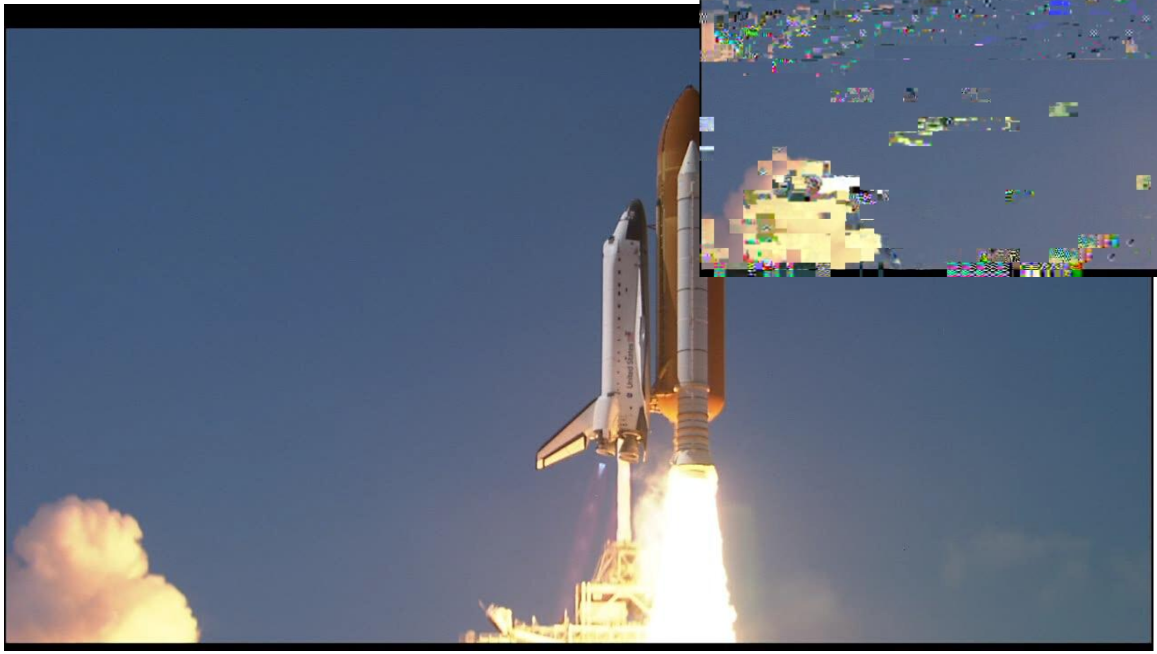
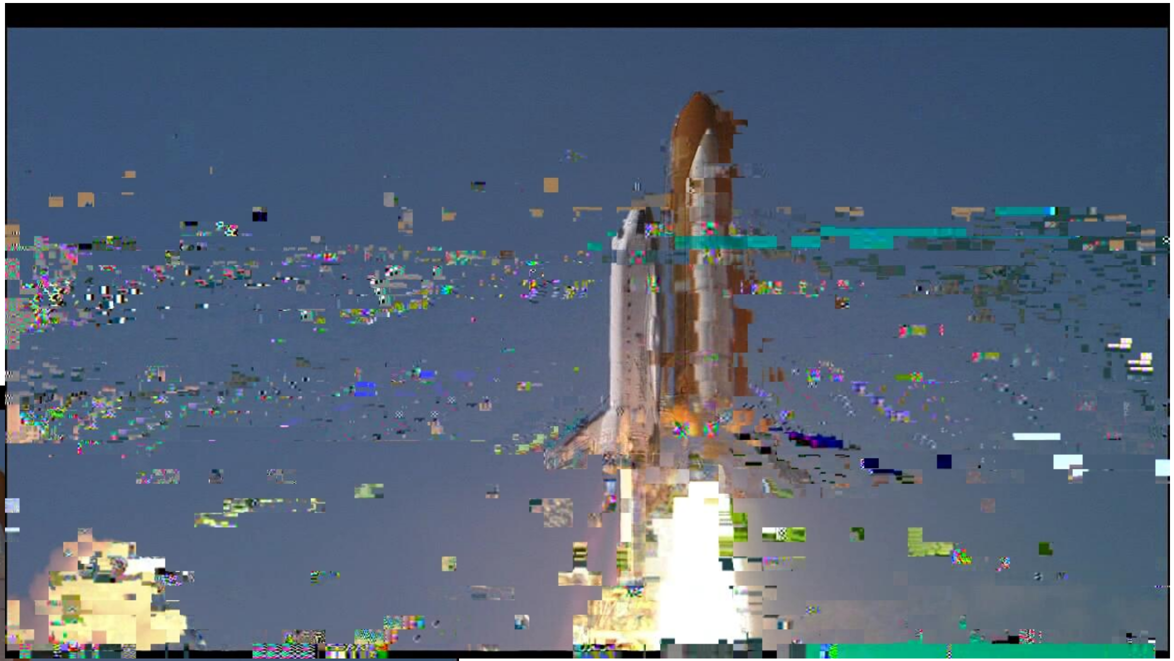
What is QoS ?

- QoS therefore refers to the ability of a service provider to support user-requirements with regard to at least 4 service categories;
  - Bandwidth
  - Latency or delay
  - Jitter
  - Traffic Loss

No IP loss:





Some IP loss:

What is QoS ? –

- 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 Key Performance Indicators or KPI's

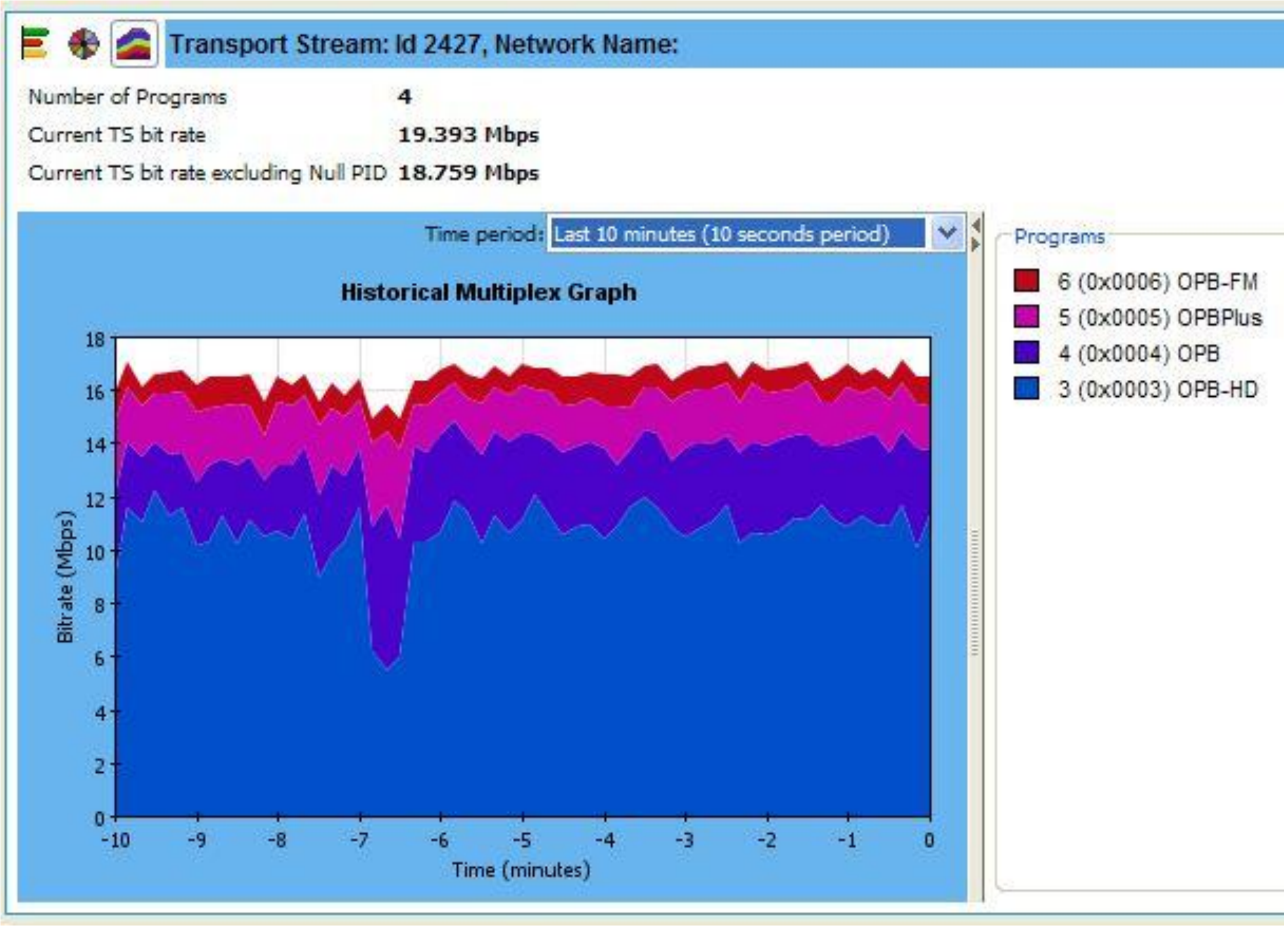
The Issues…

The issues…..

- IPTV systems are run on ‘best effort’ networks
- IPv4 and IPv6 protocols are, by definition, ‘best effort’ delivery systems
- They rely on other supporting protocols in order to provide QoS services to the user
- Data and voice services can normally cope with jitter and delays, video cannot.
- Example: Home Router QoS setting

The issues - Bandwidth

- Video Bit Rate and Multiplex Efficiency trends
  - Example: VBR HD, SD, in a Multi-program transport



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The issues - Latency/Delay and IP Jitter

- Latency or delay and IP Jitter are inextricably linked……
- They 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
  - Assume the Ethernet frame has IP/UDP/RTP encapsulation

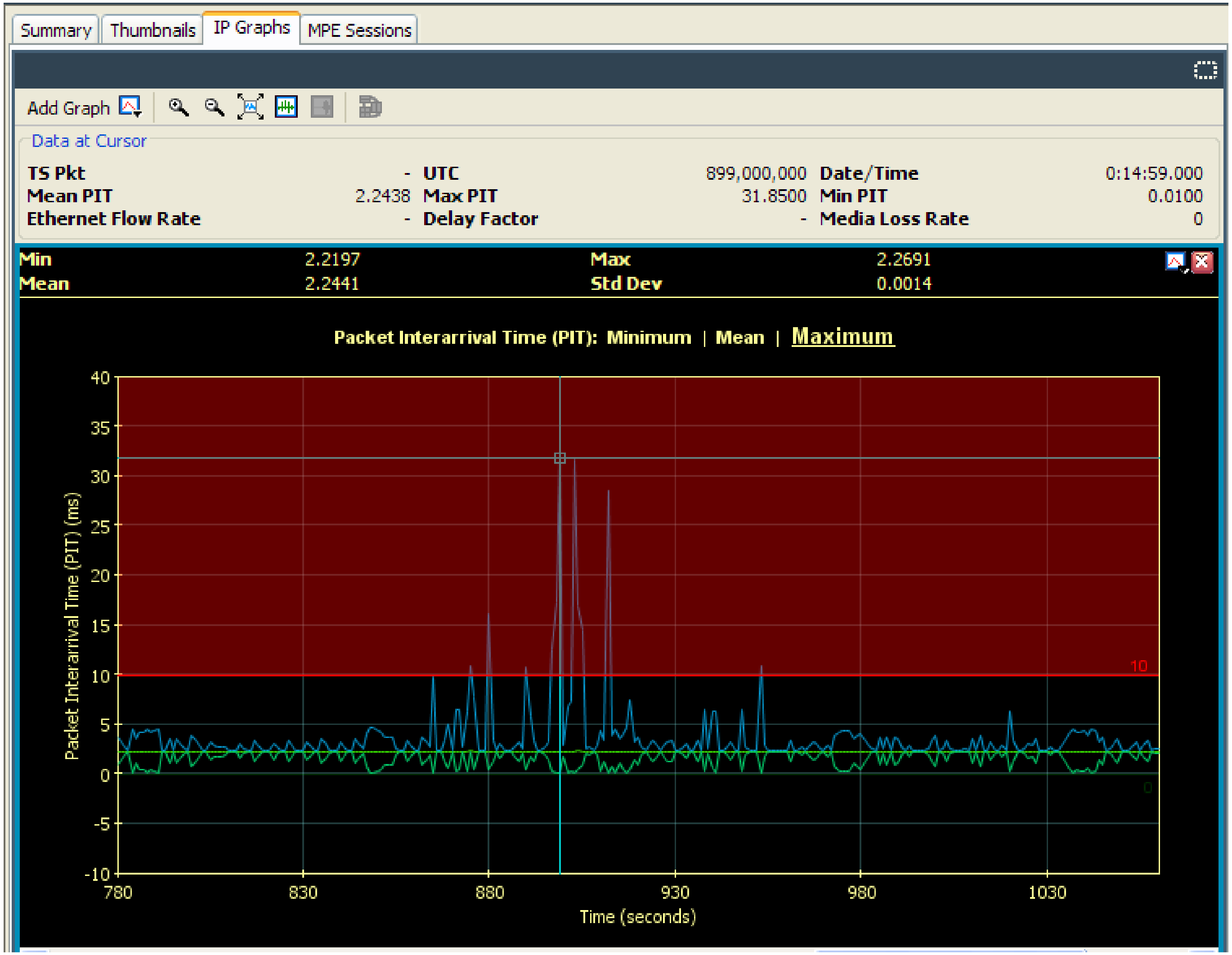
Ethernet header	14 bytes
IP header	20 bytes
UDP header	8 bytes
RTP header	12 bytes
Total	54 bytes overhead + 1316 byte payload (4.1% overhead)

  - 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 issues - Latency/Delay and IP Jitter

- The ideal packet arrival time should therefore be 2.24mS.
- 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.
- But……..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.
- It is therefore preferable to be able to measure and counteract any excessive IP packet jitter in the network.
- Measurement and display of Packet Inter-arrival Times (PIT) over extended periods is critical to ensure that the network is degrading to a point where a customer affecting situation arises.

The issues - Latency/Delay and IP Jitter



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The issues – Out of Order or Lost Packets

- Buffering issues on the output ports of network routers 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.

The issues – Out of Order or Lost Packets

Dropping one IP packet can heavily impact picture quality

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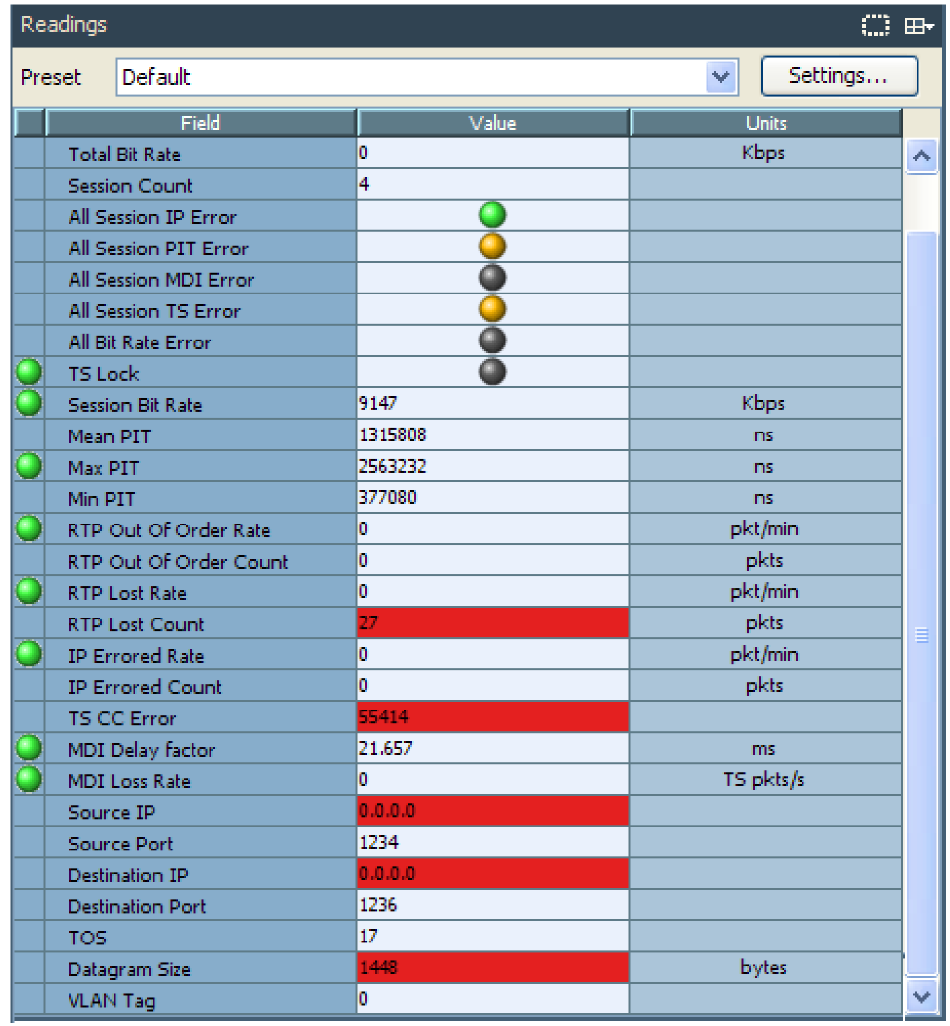
The issues – Out of Order or Lost Packets

- 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.
- Monitoring equipment needs the capability to detect the out-of-order events and provide timely diagnostic information so that the situation can be isolated and rectified before customers complain.



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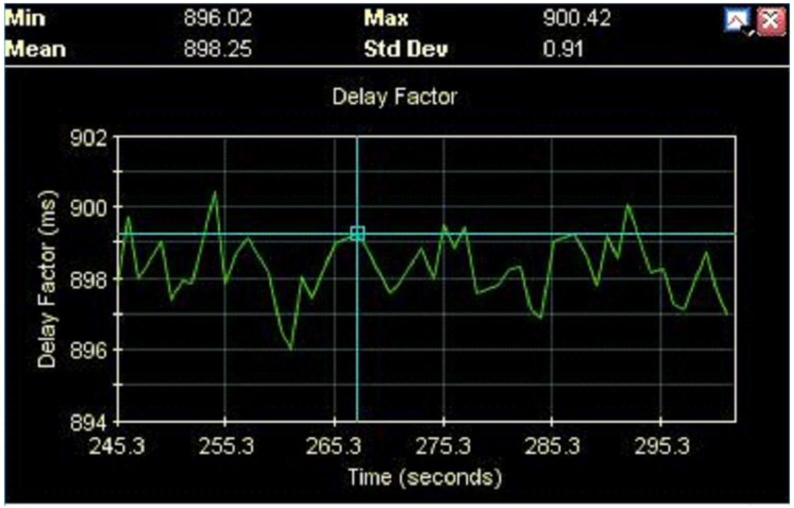
The issues – Out of Order or Lost Packets

- Recent Telco example: Over 300 Video Programs, some bad, 1 very bad

Session Alias	Has TS	TS Sync Error	TS CC ...	Packet Error	MDI:DF	PIT Error	Bit rate (Kbps)	MDI:MLR	Bit rate Error	
Funnimation	✔	●	183933	0	9.69	●	3650	2647	●	UDP
WEALTH	✔	●	6	0	10.305	●	3661	0	●	UDP
Public Acc	✔	●	3	0	14.826	●	3698	0	●	UDP
F.S.B.A.	✔	●	2	0	7.929	●	3668	0	●	UDP
MTV	✔	●	2	0	5.026	●	7467	0	●	UDP
Oxygen	✔	●	2	0	6.219	●	3672	0	●	UDP
Spike	✔	●	2	0	7.001	●	3850	0	●	UDP
VH1	✔	●	2	0	8.391	●	3857	0	●	UDP
TWC	✔	●	2	0	7.608	●	3857	0	●	UDP
GRN	✔	●	2	0	9.05	●	3675	0	●	UDP
TNT	✔	●	1	0	8.675	●	3861	0	●	UDP
Toon Disney	✔	●	1	0	10.231	●	3857	0	●	UDP
AMC	✔	●	1	0	51.106	●	3668	0	●	UDP
SciFi	✔	●	1	0	6.606	●	3886	0	●	UDP
HRTV	✔	●	1	0	10.516	●	3886	0	●	UDP
Comedy	✔	●	1	0	10.1	●	3861	0	●	UDP
E	✔	●	1	0	9.444	●	3857	0	●	UDP
C-Span 2	✔	●	0	0	9.797	●	3861	0	●	UDP
Disney Ch.	✔	●	0	0	104....	●	3461	0	●	UDP

The issues - MDI

- Media Delivery Index (MDI) is defined by IETF RFC 4445. 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 **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



The issues – MDI

- MDI is not payload aware. That is, it cannot separate video traffic from other data and VoIP packets.
- Raw UDP protocol does not provide any means to detect packet loss. So for raw UDP, the packet loss portion of MDI is calculated using MPEG Continuity Count errors.
- 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.

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The issues – MDI summary

- The MDI Media Loss Rate (**MDI-MLR**) is based on Continuity Counters – measures packet loss
  - Therefore any other error, such as TS syntax errors, cannot be detected by MDI
- The MDI Delay Factor (**MDI-DF**) is transport stream bit rate based, derived from PCRs – measures packet jitter on the network
  - A bad PCR from a multiplexer could trigger an MDI error even though there is no network issue.
- 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

The issues - Cross Layer Timing

- 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.

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The issue - Cross Layer Timing

- 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)
- 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.

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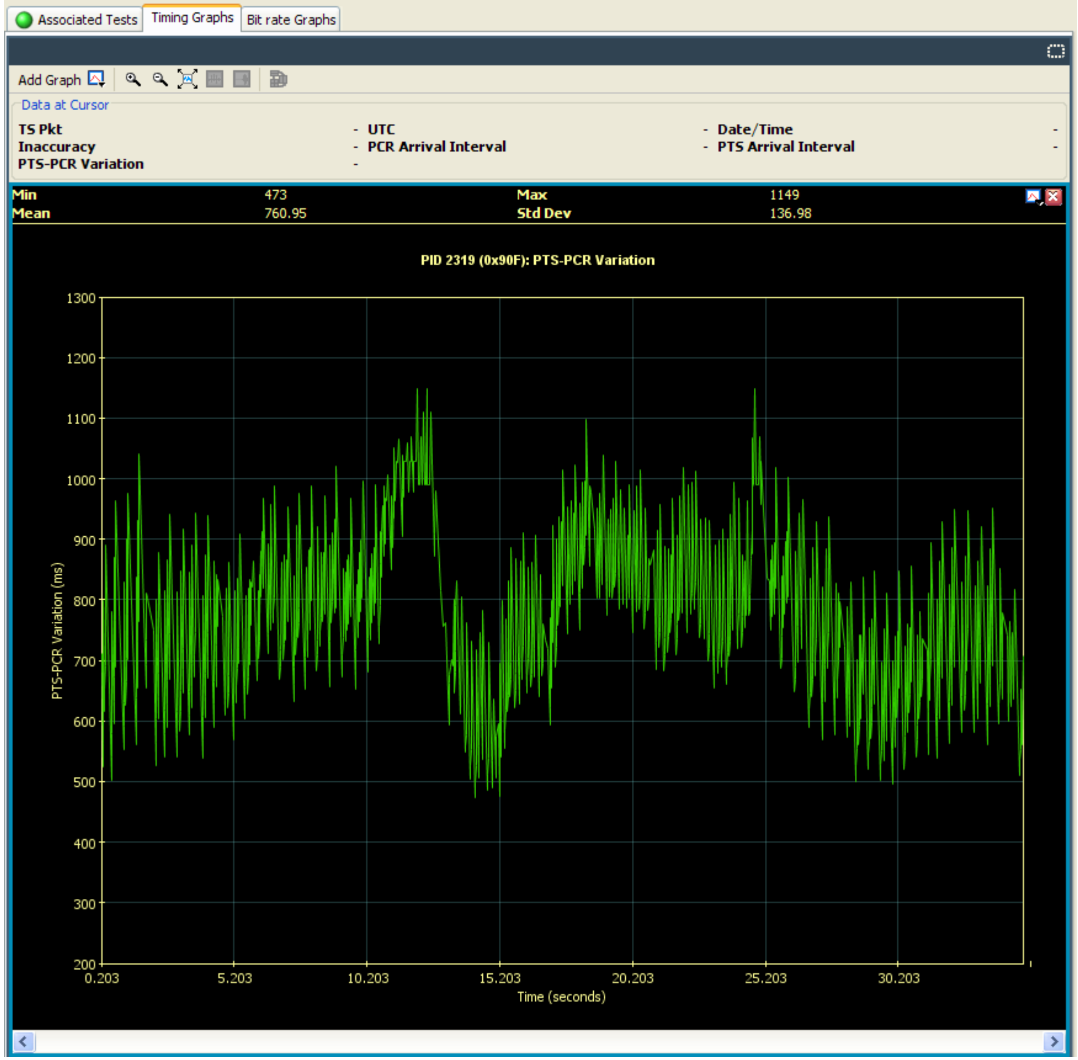
The issue - Cross Layer Timing: PTS/PCR Variation



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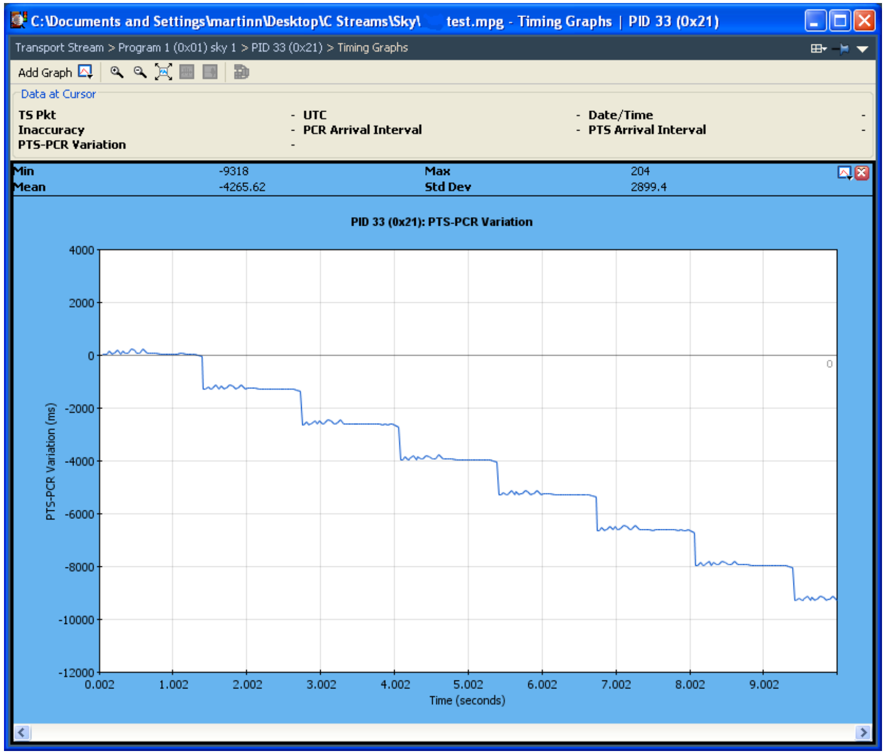
The issue - Cross Layer Timing: PTS/PCR Variation

The PTS is associated with a PES packet and essentially defines when the contained frame (or AU) should be presented to the user. Note that a PES packet may contain > 1 AU.

The PTS-PCR relation shows the time that an AU spends in the Elemenatary buffer and re-order buffer.

For audio, it is simply the time spent in the Main Buffer. The graph is a measure of the T-STD buffer model.



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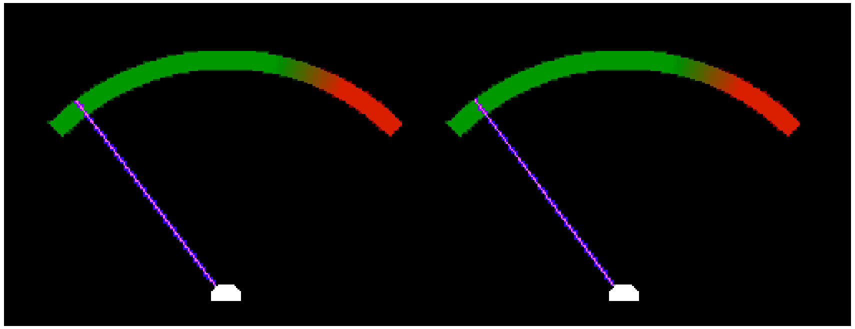
Video Backhauling from Remote MTM400A

- Monitoring content via thumbnails
- Redirecting video and audio from a remote MPEG probe to a Network Operations Center PC. Decoded using VLC



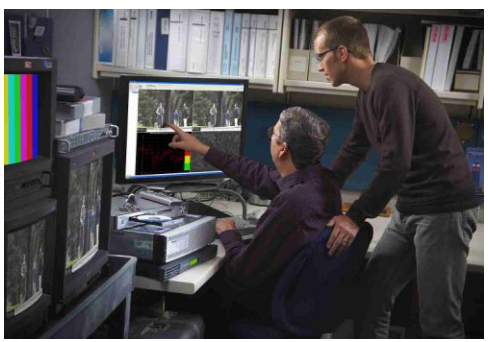
Conclusions

- 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 using differing bandwidth and QoS requirements.
- 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.
- Operators and engineers can therefore 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 IPTV Product Portfolio

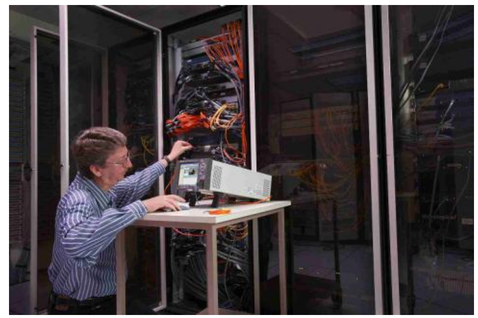
Solving today's digital video delivery and quality challenges

MPEG/IP Test Systems
MTS415/430



Picture Quality Analysis
PQA500

Design



MPEG/IP Generator/Analyzer
MTX100B/MTS400P

MPEG Transport Stream Analysis
MTS4SA

Rollout

MPEG/IP Monitor
MTM400A

Triple Play
Unified Assurance

Monitoring

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Question Time

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