IPTV CHALLENGES AND METRICS

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IPTV, one of the components of triple-play services, is being used by traditional wireline carriers as a way to leverage their DSL customer base, in order to quickly and efficiently enter the broadcast TV and other video-content delivery markets. Even the cable MSOs see the value in IPTV as they migrate their networks from the legacy analog and digital channel overlay technologies towards all-broadband IP networks.

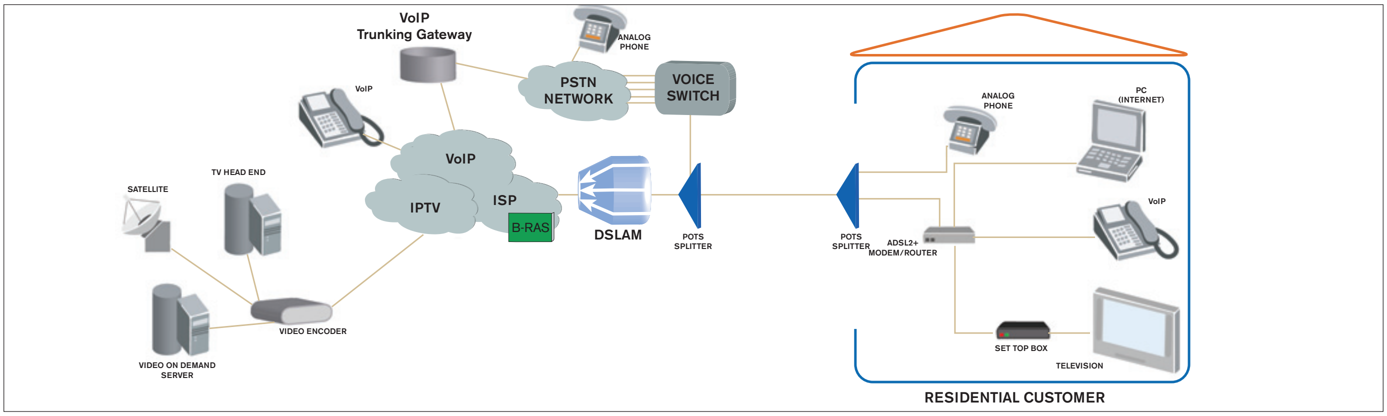

Figure 1. Triple-play network architecture

IPTV was initially designed and optimized to work over DSL circuits. Traditional cable TV coaxial networks worked on the premise that all channels were delivered to all subscribers all of the time. In contrast, the IPTV system optimizes bandwidth by delivering a small number of channels to each consumer, without reducing channel availability. When customers change channels or order a pay-per-view video, they automatically send a request to a server upstream to activate the new channel. As a bandwidth-saving measure, channels that are no longer being watched are no longer delivered. Through these mechanisms, the consumer cannot differentiate between the several hundred channels offered by their cable TV provider and their new IPTV service provider.

Factors Affecting IPTV Quality of Service

There are four main factors that affect IPTV services, each resulting in poor image quality to the subscriber.

Network Bandwidth

The total amount of video-stream data that can be sent is ultimately limited by the bandwidth provisioned over the access network. Any increase in bandwidth demand that is above the maximum capacity of the link will result in video packets being lost, causing impairments on the screen.

Impulse Noise

The copper-loop plant is susceptible to short impulse noise caused by external sources. These impulses can lead to large bursts of errors, which could have a significant impact on the video picture quality.

Packet Loss

IP packet loss can represent various levels of image impairments, ranging from a single, unnoticeable missing point of the video sequence to a long period of degraded, pixilated or unavailable images.

Jitter

MPEG2 or MPEG4 video streams are affected by a phenomena called jitter. The transport stream carrying a program clock reference (PCR) may be affected by jitter as well. This condition will have a direct impact on the decoding process performed by the customer's set-top box.

IPTV Metrics

In order to avoid quality degradation caused by these occurrences, several parameters can be tested so as to correct the situation before service is affected.

Packet Loss

Packet loss occurs when one or more packets of data traveling across the access network fail to reach their destination. Retransmission of packets is not intended due to the real-time nature of IPTV and, as a result, every lost packet will have an impact on the quality of the video stream delivered to the customer. If the missing packets were related to the reconstruction of I frames, there is a good chance of losing the video signal for a short period of time. If the missing packets are related to B or P frames, the impact is less severe but image-quality issues could still be experienced. When testing the real-time transport protocol (RTP) packet integrity, the measurement must be taken in real time and any packet loss must be displayed as it happens.

Packet Delay

Every RTP packet is synchronized and time-stamped locally at the time of transmission. In a packet-based network, it is quite common that the route for transporting the packets is not always the same; packets may arrive at different times and out of order; the RTP protocol allows out-of-order arrival of packets. Since every RTP packet has a sequence number, as long as the delay does not exceed the size of the receiving decoder buffer, the packet can be processed and placed in the right position for decoding. If the delay exceeds the buffer, the packet is dropped and considered lost. When testing for delays, the RTP packet measurement must be taken in real time, and any packet delay must be registered as it happens. Test results should display a maximum, minimum and average delay time.

Packet Interarrival Jitter

Packet interarrival jitter is an estimate of the statistical variance of the RTP data packet interarrival time, which is measured based on the RTP time stamp. The interarrival jitter is defined to be the mean deviation of the difference in packet spacing at the receiver compared to the sender for a pair of packets. The interarrival jitter should be calculated continuously as each data packet is received from the same source. The jitter calculation must allow profile-independent monitors to make valid interpretations of reports coming from different implementations.

PCR Jitter

Transport stream (TS) is a format specified in MPEG-2 Part 1 of the ISO/IEC standard 13818-1; it contains seven packets of 188 bytes each (184 bytes of payload and 4 bytes of packet header). TS can also be utilized to transport MPEG-4 encoded video.

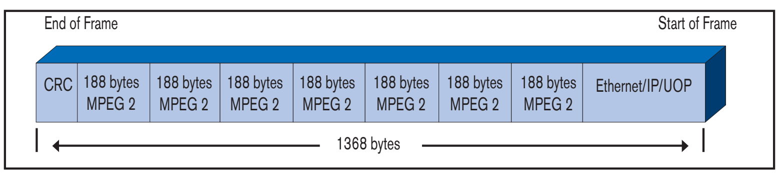

Figure 2. Content of transport stream

Included in this TS are clock-synchronizing parameters that are sent at regular time intervals. These clock-synchronizing fields, called program clock reference (PCR), are the instantaneous values or a sample of the 27 MHz system time clock (STC), located at the MPEG video encoder. The PCR in the TS permits the MPEG decoder to recreate the encoder's STC. This recreated clock guarantees that the decoded video output operates at the same rate as the video signal input to the MPEG encoder.

The ITU-T Recommendation H.222.0/ISO/IEC 13818-1 indicates a maximum interval of 100 ms between consecutive PCR values. The Digital Video Broadcasting (DVB) organization recommends that all DVB-compliant systems transmit the PCR values with a maximum interval of 40 ms. However, all receivers should work properly with intervals as long as 100 ms. The standards do not insist that the interval be constant.

At the receiver, the regeneration of the 27 MHz system clock for the program under the decoding process is controlled by a signal that makes use of each of the PCR values corresponding to such a program at the time of arrival to introduce corrections when needed. It is assumed that the stability of the clock regenerator is such that the phase does not unduly drift from one PCR value to the next over intervals as long as 100 ms. However, it is the responsibility of the TS to provide the correct PCR values (with an error no greater than 500 ns from the instantaneous phase of the system clock). The limit of 500 ns may be exceeded as an accumulated error over many PCR values, and it should be considered in terms of its drift contribution. This value, however, specifically excludes the specifics or impairments of the transport layer.

The MPEG TS is transmitted over any real network being exposed to certain effects caused by the network components, which are not ideally transparent. One of the pre-dominant effects is the acquisition of jitter in relation to the PCR values and their position in the TS. If the PCRs do not arrive with sufficient regularity, then this clock may jitter or drift. Recovery of the PCR allows the decoder (STB) to synchronize its clock to the same rate as the original encoder clock. High PCR jitter levels may cause the receiver/decoder to go out of lock, which will affect the video displayed on the TV.

The Packet Identifier

TS packets have a fixed length of 188 bytes with a minimum 4-byte header and a maximum 184-byte payload. Within the 4-byte header, there are multiple data fields, some of which are used for management and control purposes.

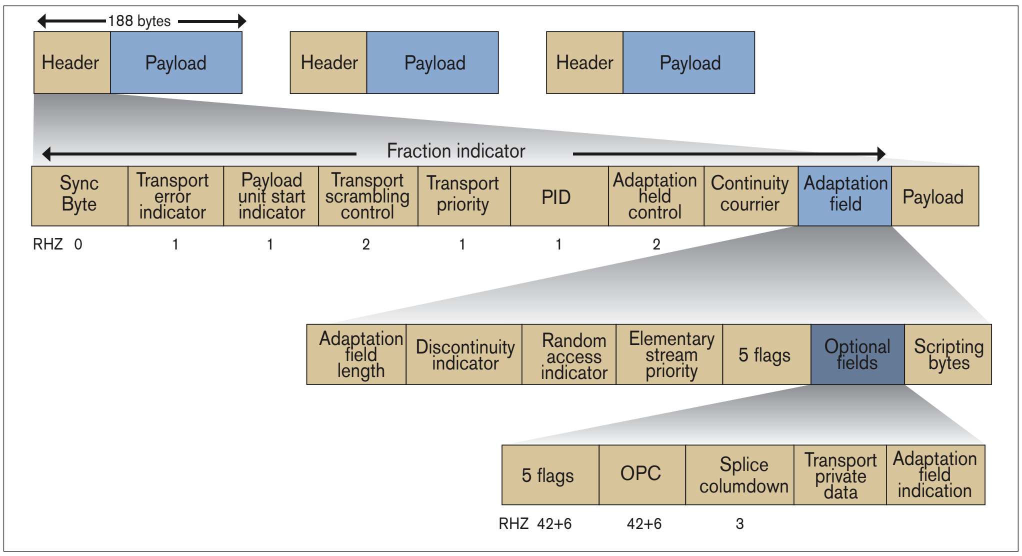

Figure 3. Transport stream structure (source: iec.org)

The packet identifier (PID) is part of that 4-byte header and represents a unique address identifier for the type of packet or payload carried by the TS. Video or audio packets in the stream need to have a unique PID; this allows the decoder or STB to process the packets accordingly. The PID value is provisioned during the MPEG multiplexing stage. Certain PID values are reserved as indicated in the following table:

Reserved PID Values
8,191: null packet
0: program association table (PAT)
1: conditional access table (CAT)

It is necessary to ensure that the PID assignment is done correctly and that there is consistency between the payload structure identifier (PSI) tables content (set of tables required for demultiplexing of the MPEG and sorting out which PIDs belong to which programs) and the associated video and audio streams in order for STBs to reconstruct a media stream from all its video, audio and table components. This is one of the main components of MPEG monitoring and testing.

Finding and correctly decoding a specific PSI table will determine whether or not the STB will subsequently be able to identify and decode video and audio information for the specific stream.

The ability to see the PID configuration and consistency, as well as the ability to monitor packet loss and rates for specific video packets, audio packets or PSI tables, allows service providers to isolate and troubleshoot problems affecting IPTV service.

Media Delivery Index (MDI)

MDI (RFC 4445) is a scoring mechanism that combines jitter and packet loss in order to determine the ability of a network to transport high-quality video. It does not take into consideration the encoding method. MDI measurements can be used as a diagnostic tool or as a quality indicator for monitoring a network intended to deliver applications, such as streaming media, IPTV, VoIP or other information sensitive to arrival time and packet loss.

- Delay Factor – The delay factor (DF) is the maximum difference observed at the end of each media-stream packet between the arrival and the drain of media data. This assumes that the drain rate is the nominal constant traffic rate for constant-bit-rate streams or the piece-wise computed traffic rate of variable-rate media-stream packet data. The DF is the maximum observed value of the flow-rate imbalance over a calculation interval. This buffered media data in bytes is expressed in terms of how long, in milliseconds, it would take to drain (or fill) this data at the nominal traffic rate to obtain the DF.

  The DF gives a hint of the minimum size of the buffer required at the next downstream node. Greater DF values also indicate that more network latency is necessary to deliver a stream due to the need to pre-fill a receive buffer before beginning the drain to guarantee no underflow.

  Refer to RFC 4445 for details on the DF calculation algorithm.
- Media Loss Rate – The media loss rate (MLR) is the count of lost or out-of-order flow packets over a selected time interval where the flow packets carry streaming application information.

MDI test results therefore combine the DF and MLR results and are displayed in the following format: DF: MLR.

Zap Time

Zap time refers to channel-changing, and this process is done through Internet Group Management Protocol (IGMP), which is the protocol responsible for requesting to join or leave a video stream. In IPTV, the selected TV channel is associated to one video stream. In order to change the channel, it is necessary to leave the channel that is no longer required and join a new video stream associated with the new channel.

In traditional cable TV, all channels were present and changing channels meant switching to a new frequency. Changing channels was never a challenge and definitely not a source for user dissatisfaction.

In the IPTV environment, a deployed system must be able to perform predictably in a realistic network load condition. In case of overloaded network conditions, the system must still be able to deliver user satisfaction.

The following parameters related to channel changing can be tested:

- IGMP Join Latency – This parameter represents the time required to join a multicast group up to the point that the first video stream data is available for decoding.
- IGMP Leave Latency – Similarly, this parameter represents the time required to leave a multicast group up to the point that the last video stream data is available for decoding.
- Zap Delay – This parameter represents the time required for changing channels and is calculated as the time between the sending of a channel leave request and the receiving of the first video stream data for the new just-joined video stream.

Conclusion

The channel-changing delay measurement is not sufficient to characterize the performance of an IPTV network. In order to detect network malfunctions that could affect zap time and video quality, it is important to perform additional network-related testing to locate the source of the problem.

Metrics such as jitter, MDI, packet loss and delay have all been defined to help service providers achieve efficient IPTV transmissions and thus retain and increase their customer base.

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