Quality of Video-over-IP
Measuring the Quality of Streaming MPEG-2
Transport Streams over IP

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SUMMARY

Video-over-IP is a new and emerging technology that combines switched packet networking with streaming video. There are few standards for Video-over-IP today. The integration of these two technologies has lead to several questions as to the measurements that determine the quality of a Video-over-IP stream. Before we can determine what to measure, a firm understanding must be gained about the fundamental properties that define the quality of a Video-over-IP stream.

Unlike data transfers over IP, streaming video quality is measured live and at the end-point, or more to the point the TV. Quality end-point video is not solely a function of network bandwidth nor is it solely a function of MPEG-2. In fact many of the issues that surround quality end-point video are a combination of both the MPEG-2 quality and the level of deterministic IP packet delivery of the network. Unlike data traffic which measures quality by speed of reliable through-put with little attention to the nature of the payload, video (as voice) demands more from network transport. Networks designed to carry streaming video must account for the payload they carry. Furthermore the type of MPEG-2 stream being transported effects the minimum and maximum boundary characteristics that the network packet delivery and the overall system must conform to for quality streaming video at the end-point.

Measuring and monitoring these streams will involve measuring Ethernet packet arrival times at layer 3, the IP layer, average and instantaneous behaviors of these arrival times and finally the boundaries of the system by decoding part of layer 7, the MPEG-2 content. The MPEG-2 content combined with the system buffering limits will impose the boundaries on the transport.

This paper will discuss the current technology of Streaming Video-over-IP using an example of Video on Demand (VoD). Then look at the challenges at each stage of the VoD system and where quality needs to be measured. And finally it will look at the key parameters of determining quality video at the end-point of Streaming Video-over-IP networks using a system approach to network performance measurement.

CURRENT TECHNOLOGY – MPEG TRANSPORT STREAM OVER IP

What is Streaming Video-over-IP? Streaming Video-over-IP is a technology in which a consumer can watch video content, at near real time, over and IP network. Although there are many applications for moving video content over IP, the most common example of Streaming Video-over-
IP is a service that is being provided by some cable companies called Video on Demand. Video on Demand, implemented over an IP network, is an excellent example of Streaming Video-over-IP. In a VoD system a customer can order a movie from his living room that is served from a remote location over an IP network. The video is produced (served from a disk server), wrapped in an UDP/IP packet, transported to the customer’s home and consumed (watched).

The figures below diagram a simplified block diagram of a VoD network, from the video server at a cable outfit to the local cable hub and then finally to the customer’s home. From this basic overview of the technology we can explore the critical parameters of the system.

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**Figure 1 – Simplified Block Diagram of a VoD Network**

**Figure 2 – Flow of a Movie from the Server to the Home**
Each MPEG-2 TS packet has header with unique ID to indicate content of packet.

Format of Typical MPEG-2 TS over IP Packet

<table>
<thead>
<tr>
<th>CRC</th>
<th>MPEG-2 188 Byte Packet</th>
<th>MPEG-2 188 Byte Packet</th>
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</tr>
</thead>
</table>

Figure 3 – MPEG-2 TS Structure at the Server RAID

Movie 1 Wrapped for IP Transport
Packets are then forwarded over the switch network to end destination the QAM

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Figure 4 – Structure of the IP Packet Containing MPEG-2 TS Payload
Using this example many questions can be asked. How do you know this video played on the customers TV without error? If there was error what caused the error? Was the MPEG bad? Did a switch drop a packet or change the stream in some way that caused the video to fail? How long was the stream in failure?

Not to mention the scaling questions that can be asked. What happens if several customers at several homes purchase videos at the same time? Does the customer’s neighbor watching a VoD movie affect the customer’s video quality?

**KEY PARAMETERS TO QUALITY STREAMING VIDEO-OVER-IP**

To start to answer these questions we need to simplify the model of the above example. Figure 6 shows a simple conceptual model of a video stream over a Gigabit Ethernet Network.
Gigabit Ethernet
100s of videos playing simultaneously

Movie 1 Packet of the Stream
188/204 Byte MPEG Packet
188/204 Byte MPEG Packet
188/204 Byte MPEG Packet
188/204 Byte MPEG Packet
188/204 Byte MPEG Packet
188/204 Byte MPEG Packet
188/204 Byte MPEG Packet

Ethernet/IP/UDP
CRC

Inter-Packet Gap

7 – 188/204 Byte Packets

Movie 1 Packet of the Stream

Figure 6 – Packet Structure on GbE

Basically from the VoD server MPEG-2 is wrapped in a packet and shipped out at a constant rate consistent with the rate of the MPEG-2 TS. For example, movie 1 was MPEG-2 encoded at 3.75 Mb/s, meaning the decoder for video must see 3.75Mb every second with a ± 500ns MPEG-2 packet jitter tolerance. So the VoD server groups 7 MPEG-2 TS packets for every Ethernet packet and (theoretically) sending that packet out the Ethernet port at an even and constant rate as to facilitate 3.75 Mb/s at the end point.

Because there are multiple clock domains in this system, buffering is used to help smooth out clocking and speed variations. Figure 7 shows the basic flow diagram for quality Streaming Video-over-IP. As Ethernet packets come from the VoD server and from the switched network, the MPEG-2 TS packets get buffered and streamed to the decoder at a smooth 3.75 Mb/s rate. Then the MPEG-2 is decoded and displayed on the TV.

This model does not implement a particular buffer size, but utilizes and effective buffer size. No matter what the size of the buffer, there is a delivery condition that can make the buffer overflow or underflow, resulting in video quality degradation due to MPEG-2 packet loss. In this model it is important to understand the behavior of the stream. Watching for the min and max cases will determine the effective buffer size and/or delivery behavior required for quality Streaming Video-over-IP.
It is suggested that there are five properties that must be measured and monitored to ensure quality transport of Video-over-IP:

1) Inter-packet arrival jitter causing delay
2) Inter-packet arrival jitter causing burst
3) Ethernet packet loss
4) Ethernet inter-packet arrival average drift/deviation from the MPEG-2 data transport rate
5) MPEG-2 quality due to packet corruption on the network, MPEG-2 encoding errors, or MPEG-2 packet loss

The best way to explain these is to diagram the effect on a system looking at these measurements. The figures below show how these five properties can affect end-point video quality.
Ethernet packet jitter that causes large delays can cause the end-point buffer to run dry, producing segments of time in which the decoder has nothing to decode. This leads to degradation of the video quality observed on the TV. In many cases the TV will show macro blocking video or simply go blank. Jitter delays can be caused by several things including switch QoS settings, switch aggregation, and/or server problems.

Ethernet packet long term rate variations can also cause the buffer to run dry in the same way. This happens when the average Ethernet inter-packet gap timing produces a data delivery rate less than the MPEG video rate, in this case 3.75 Mb/s. So, for example, if the server were to send Ethernet packets of MPEG out on the network such that the sustained data rate was 3.50 Mb/s; the buffer would eventually run dry.

Figure 9 – Effects of Jitter/Burst and Ethernet Inter-Packet Gap Drift

Similar to the prior case, Ethernet jitter that causes bursts of packet can cause buffer overflows. This is a much more difficult case to monitor, because data loss can occur at several points on the network. The causes are significantly different. In this case, faster delivery of Ethernet packets overflows the buffer in the next case Ethernet packets are dropped in the network. This could be either a delivery problem or a potential bandwidth problem. The server may be bursting Ethernet frames or network congestion is causing network switch elements to burst frames. So measuring quality at the TV can not show the entire story. When the network drops packets due to packet burst-induced overflow, the MPEG decoder may actually underflow as a result; regardless of the MPEG decoder’s buffering ability. This happens because the MPEG decoder’s buffer continues to drain while some of the MPEG packets simply did not make it to the decoder’s buffer. Thus, there may actually be both overflow and underflow conditions existing simultaneously on the path between the encoder/server and the decoder.

Also similar to the last case, Ethernet packet long term rate variations can cause the buffer to overflow, leading to MPEG packet loss.
In this figure, Ethernet packets are dropped on the network. This is a simple case to see the effect; if the data does not arrive it will lead to poor quality.

In all these cases there is an underlining issue of quality of the MPEG. If the MPEG is encoded poorly or the MPEG payload is corrupted anywhere along the way, including data corruption right from the RAID, video quality is compromised.

Another effect to consider is the dynamic nature of the network and its influence on other streams. Because Ethernet is a shared network, the more streams the greater the chance network switch elements have to buffer and (re)order traffic, thus creating jitter, delay, burst and packet loss.

**CONCLUSION AND OBSERVATIONS**

Streaming Video-over-IP is a new and emerging technology that has few standards. Quality at the end-points of Streaming Video-over-IP requires measurement and monitoring of the jitter, packet loss and payload quality as it relates to a buffer limit in the system. This is a system approach to video network performance measurement.

As stated, one implementation of Streaming Video-over-IP is Video on Demand. As higher numbers of streams are served, a more comprehensive understanding of the system and its bounds is needed to insure and maintain the quality for the customer at the home.

To insure quality of the transport, measurement and monitoring at several points along the system architecture may be required. Measuring and monitoring of both network behavior and MPEG quality is needed.

There are several observations to make looking at this model. This system approach to measuring end-point video quality of streaming Video-over-IP is focused on the video and sets demands on the network transport. The faster the MPEG rate the more constricting the boundaries of the network. For example, using the same buffer at the end-point for DVD quality MPEG-2 vs.
High Definition quality MPEG-2 may produce considerably different results as the requirements on the network are quite different. This is true even though the increase in data rate is well within the Ethernet network boundaries.

DVD quality MPEG-2 has a data rate of approximately 3.75 Mb/s, High Definition quality MPEG-2 is approximately 19.3 Mb/s. This rate is approximately 5x on the network and means the buffer can handle 1/5th of the network jitter or inter packet arrival time drift. A jitter/delay of 30ms may be fine for the DVD case, as the drain rate from the buffer is less than the HD case. Because the High Definition quality case will process (i.e. drain) the buffer much faster, the same network jitter/delay may be unacceptable.

A specific drain rate of a buffer is neither good nor bad in and of itself. It does, however, impose real limits and behavior profiles on network transport. Likewise the buffer size has the same effect. In fact, the drain rate and the buffer are directly related with respect to the network transport. For example if you increase the buffer by the same factor as the increase in the speed between the rates of HD MPEG-2 over the DVD MPEG-2 in the last example, the behavior differences would be minimized. Note, other system parameters may be affected by faster or slower streams at the server and/or throughout the network.

In conclusion, as this technology matures, more video streams will be able to move over IP but the measurement and monitoring requirements of the system must be accounted for to insure a quality video product at the end-point.