Niagara® Video Corporation

Video Technology Crossroads … What's next?

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A little background & history

The history of digital video technology and transport was first led by the broadcast industry and the service provider industry. Both based their technologies and deployments on their industry standards. An excellent example is how Bellcore led the effort to develop the first MPEG standards and the fact that some portions of that very first standard are still in use today. The Internet segment and its support of video and video transport developed in a very different way and was led by big companies like Microsoft, Adobe and Apple. The only upstart to this was Real Networks who found a niche in providing a bridge between Apple and Microsoft. Today the defacto standards of Internet video and streaming are primarily based on Adobe and Apple (as a side note it was actually Macromedia who first adapted Flash for video and they were acquired by Adobe).

Although at first these two market segments had very different needs more and more, and with the wide spread adoption of OTT, these needs have converged. Although broadcasters will always have some requirements related to production that are specific to their workflows increasingly they are using the same technologies and standards that Internet focused companies use for transport and delivery of content to end users. For a long time, there was not a big driver to try to bring these two worlds together; Broadcast and Service Provider + Internet focused companies. Now the needs of the two worlds are converging and starting to overlap.

Where IP video protocols come from?

Broadcaster's and service provider's video and transport protocols and codecs were designed for meeting their specific requirements and needs. Internet based companies had different needs when streaming first started to take place and also had to deal with many more constraints. When it came to video transport over IP, as it relates to broadcasters, a key an important factor was that they were using dedicated IP networks. As a consequence, they could use UDP for transport. The Internet segment evolved in an environment devoid of dedicated networks. Due to this the Internet segment's video transport protocols were designed for unreliable IP networks and had to use TCP for transport.

In our next section, we will take a look at the differences between TCP and UDP.

TCP versus UDP

It is helpful to have a quick look at the two protocols to see how they are different as this leads to how they are now developing, evolving and being deployed.

UDP is a connection-less protocol. It sends out packets but has no idea if they are being received nor does it know if any are getting lost.

TCP is connection oriented. It has to be first connected to the receiver and if packets are lost then it is sends a command to re-transmit the lost packets.

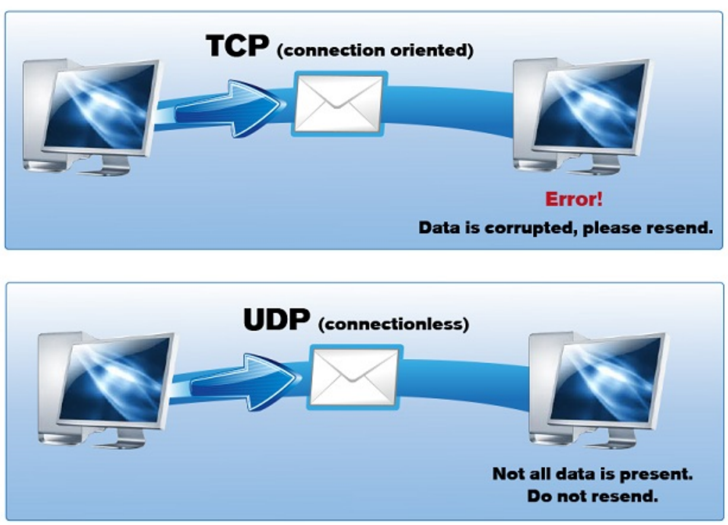

Figure 1 – TCP versus UDP

Two other factors to keep in mind are:

- Since UDP is connection-less it can support unicast, multi-cast and broadcast transport.
- TCP can only support unicast transport.

A further illustartion of TCP versus UDP differences is below.

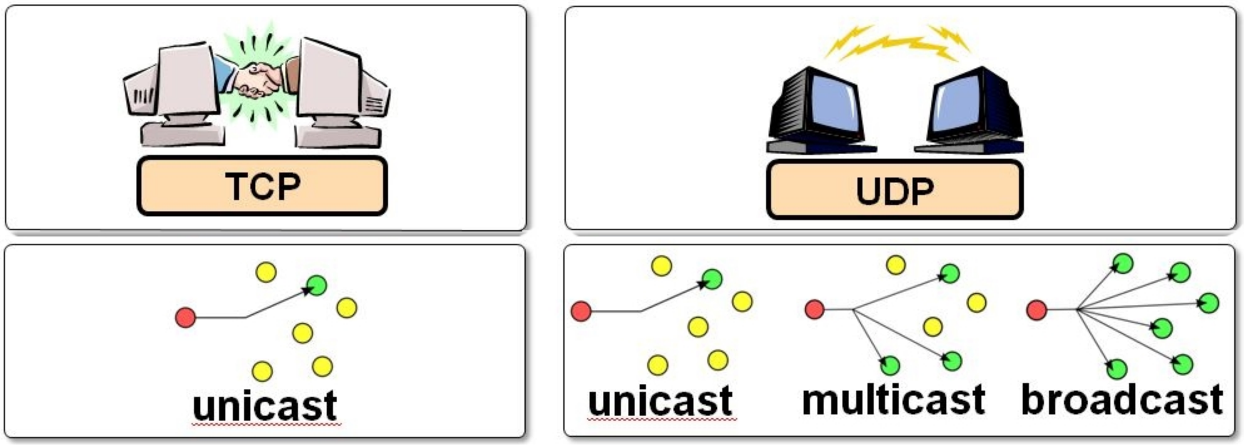

Figure 2 – Transport Comparisons

What are the needs of each segment?

To understand the needs of all the players involved it is helpful to provide a very short history of both. It is important to note that the broadcast and service providers first developed a live delivery system. Even though many programs were taped and tape delayed, the transport was a live, real-time system. In the case of the Internet the history was the opposite. Starting in the dial-up era live streaming really wasn't technically feasible. Early VOD systems relied on down loading and people had to wait until they could play the video. Then progressive down loading was developed so now customers did not have to wait to start playing the file. As broadband was introduced eventually live streaming became more feasible. In those days people were happy just to get something that looked like video. Early VOD content typically were very small resolutions and frame rates. Over time broadband speeds increased and resolutions and frame rates increased to a point today where they rival the quality of the broadcast and service providers. In some cases, they even exceed the quality, an example of this is 4K content available on Netflix.

What new needs are developing?

The Internet market has not always offered the same kind of video service as broadcasters. However, increasingly needs are converging. Closed captioning was developed and is in use by Internet providers today but the systems today can only offer a single audio program. The Internet market is working on improving video latency but these improvements are still not as good as the broadcasters typically can achieve. Live streaming for the Internet still primarily relies on RTMP (Adobe Flash live) and this works very well over the Internet for encoders streaming to a CDN. However, RTMP cannot support multiple audio programs and has little control over latency. It also does not support new codecs like HEVC (h.265). Other markets primarily use UDP from the MPEG standards. This has very low latency, support for multiple audio programs and already supports HEVC however it does not work reliably over the Internet.

The Internet market needs guidance

The Internet market is in a state of confusion about what to do to support new video features and requirements with existing transport protocols. One possibility is to use HLS as it is based on MPEG and has the ability to support all of the above features but has inherently longer latency that if users try to decrease by using smaller chunk sizes you decrease the reliability.

Guidance from broadcast segment will be helpful

For a long time, there has not a big driver to try to bring these two worlds together; Broadcast and Service Provider + Internet focused companies. Now the needs of the two worlds are converging and starting to overlap.

There are already proprietary versions of technology offering reliable Internet transport with low latency … Zixi comes to mind. These proprietray solutions have been in wide use by broadcaster and media compies.

The broadcast industry does have an effort to have a protocol for video transport over the Internet. Broadcasters wish to have one standard so that products from different vendors will operate with each other. The broadcasters push for an Internet standard is being led by the Video Services Forum and is called RIST. RIST stands for Reliable Internet Streaming Transport. The goals of broadcasters are a little different than the Internet segment companies but the resulting standard could be used and has high potential for both groups. The goal of broadcasters is to have a protocol for point to multi-point applications …whereas the Internet only needs a point to point protocol for first mile solutions.

If the Internet segment wants to have a simpler protocol than one that could be considered would be to simply add the TCP protocol to MPEG. RTMP uses TCP so really this approach would take the best of both worlds. The broadcast industry is using UDP/RTP along with an ARQ mechanism for re-transmission. Basically, this adds TCP like qualities to UDP.

Whatever does happen it would be nice to have one standard especially for companies like Niagara Video that make products for and serve customers in both markets. We would welcome your thoughts and comments on the topic. Please feel to reach out to us at any time to discuss!