THE BROADCAST BRIDGE

Connecting IT to Broadcast

Live IP Delivery

Essential Guide

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Introduction from Telestream

Scott Puopolo, CEO, Telestream.

Hello, and thank you for downloading this Essential Guide, covering Live IP Delivery.

Telestream and The Broadcast Bridge strongly feel that there are new business models to be realized and operational efficiencies to be gained with live content distribution regardless if you are a new player in the market or if you are a long-time broadcaster.

For any video business, however, content needs to appeal to the consumer. Today, streaming viewers hold all the cards. They expect to access content at any time, in any place and on any platform. At the same time, research indicates that consumers do not have much more incremental time that they can spend watching video. The consumer demands Quality of Experience (QoE) that provides flawless viewing throughout. If their expectations are not met they will move on to other channels very quickly.

In the fight for eyeballs, agility, efficiency, and quality are the three pillars that support every video enterprise: agility in ways the enterprise offers content to consumers to realize the highest dollar return on content, and efficiency in data processing, preparation, and delivery.

Businesses must automate complex workflows for the capture and preparation of video content, including both file-based and live streaming media. Enterprise operations are consolidating, with content creators now looking to reach customers themselves through their own channels to market. In seeking a direct relationship with consumers, our customers require a seamless way to capture, process, prepare, and deliver a quality of experience that meets or exceeds consumer expectations.

The series of articles in this guide, developed by Broadcast Bridge, provide not only an overview of Live IP Delivery channels but also the intricacies of HTTP and IP technology so broadcast engineers and technical managers will be able to design and support OTT systems more effectively. For content producers to capitalize on live streaming, businesses should consider solutions to increase efficiency, reduce operational and infrastructure costs, and ultimately optimize the customers' quality of experience.

Scott Puopolo, CEO, Telestream.

Live IP Delivery

By Tony Orme, Technology Editor at The Broadcast Bridge

Part 1 - Challenges

Delivering live, high quality broadcast video over the internet has always been an interesting challenge. Broadcast engineers are expected to understand and manage complex video, networking, scale, reliance, and playback to deliver reliable programming to multi-platform viewing devices. In this series of articles, we delve deeper into live-OTT broadcasting, identify some of these challenges, and present strategies for achieving reliable live-OTT distribution.

Streaming over the internet enables broadcasters to reach a much larger audience than traditional terrestrial, cable, and satellite models. Viewers are now watching their favorite programing on a whole plethora of devices including cellular phones, gaming systems, PC's, and Smart-TV's. And to increase their audience base and hence revenue, broadcasters must deliver to these viewers.

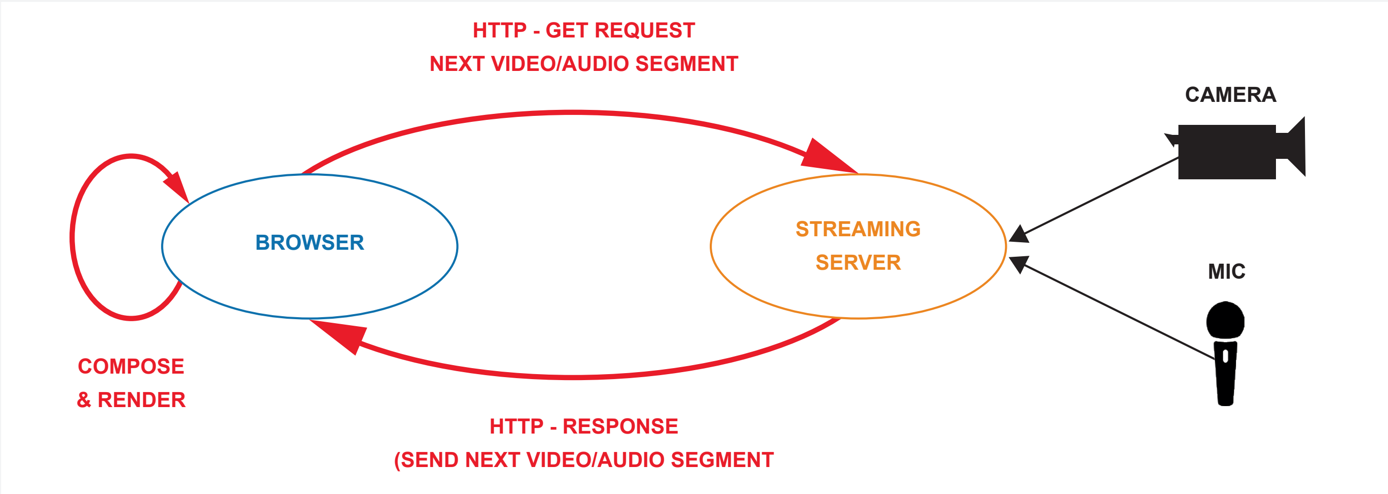


Diagram 1 – OTT uses HTTP's GET command to allow web-browsers to request (pull) consecutive segments of video and audio from the media server.

The public internet opens new opportunities for multi-platform delivery providing audiences with many new viewing options. However, this is not as straightforward as it would first appear as there are three specific challenges that will be faced by most broadcasters; bandwidths vary, latency is unpredictable, and picture sizes are determined by the playback device the viewer is using.

OTT Pulls Data Streams

Fundamentally, broadcasting and OTT delivery differs in one important aspect. Satellite, cable, and terrestrial systems all push data to the set-top-box and TV set. In contrast, OTT playback devices request a stream and pull the data from the broadcaster, giving each audience member a unique view.

The internet was developed to deliver textual documents using a client-server model. To initiate any data transfer, a client often starts by sending a "GET" command to the listening agent, often a web server. Web-servers are in permanent listening mode and when they receive the "GET" command from an authorized client, they will send the requested information back to the browser at the appropriate IP address.

Internet connected devices generally use the HTTP (Hyper Text Transfer Protocol) model to communicate with web servers. HTTP sits on top of a TCP (Transfer Control Protocol), which in turn sits on top of IP datagrams. Although more commands have been added to the HTTP protocol as it has developed over the years, the client-server, demand-supply model, is how most internet connected devices work today. Even if the viewer watches on a dedicated app, the HTTP client-server approach is used.

HTTP Scales

HTTP generally operates on top of TCP/IP to guarantee data is reliably exchanged between the client and server. Although TCP is highly effective in resending lost packets that if not resent would significantly degrade a video feed and affect the viewing experience, there is an overhead associated with TCP that can lead to increased latency and network traffic.

Other systems do exist such as RTMP (Real Time Messaging Protocol) and webRTP (web Real-Time-Protocol). Traditionally, RTMP was used in Flash viewers, but it's use has declined as delivery networks look to consolidate infrastructures to a common delivery method and Flash has become deprecated in many viewing environments.

While not initially developed to stream live video over the public internet, HTTP has become the most commonly used video stream delivery protocol today. Because it is the de-facto language for most web traffic, standards-based infrastructure already broadly exists.

Work Back from the Playback Device

To understand OTT distribution, from a broadcast engineers' point of view, is to start at the playback divide and work back to the playout center.

In IT terms, streaming is the process of breaking a file into segments and making them available to a playback device to facilitate video and audio viewing. The alternative is to download the entire file into the player. While progressive download exists for on-demand, it's not ideal as the long download times will affect the viewer experience and cost.

Fragment Media

VOD and live-OTT are similar in that they both fragment the media, so the playback device can request consecutive chunks of data and play the clips in an orderly fashion. Where they differ is that VOD has all the data available before fragmentation begins, whereas live-OTT does not, and must compress and fragment video, audio, and metadata on the fly.

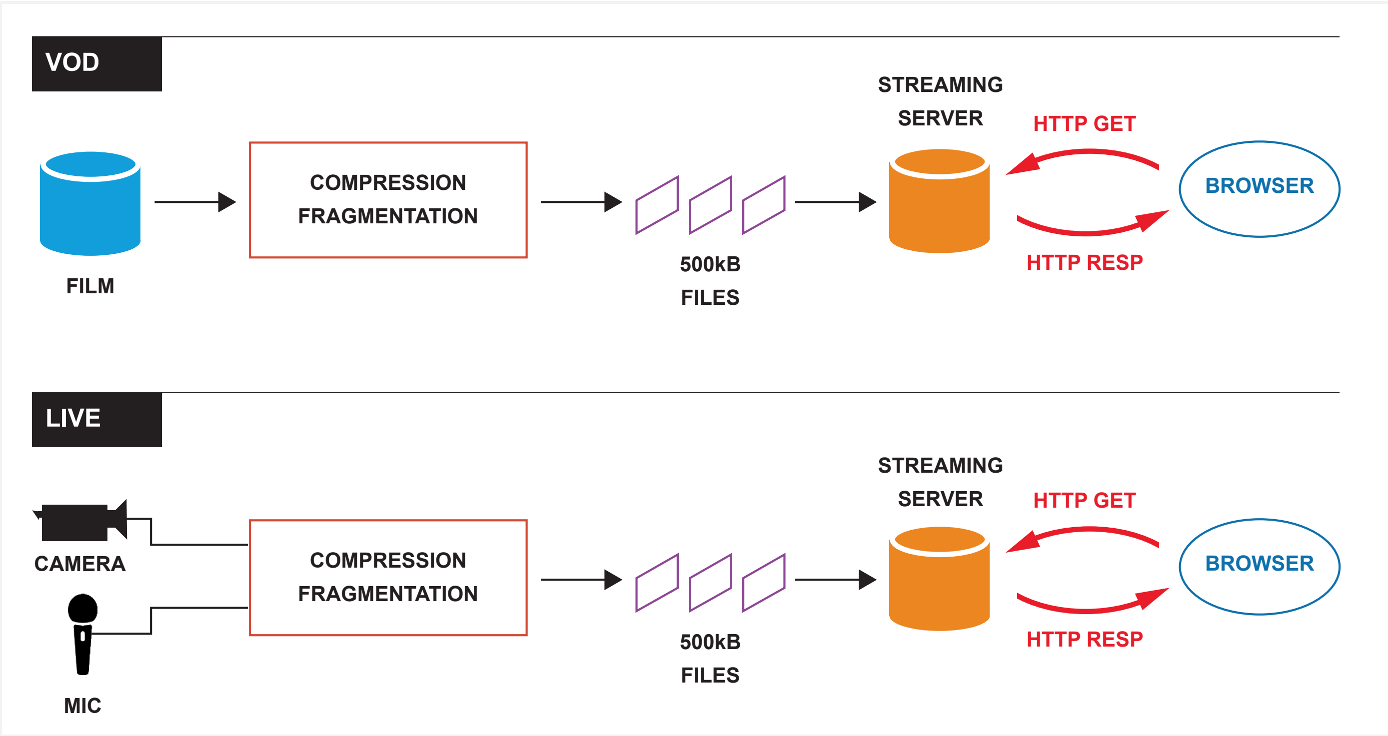


Diagram 2 – VOD and live-OTT both segment media into small segments so the browser can generate HTTP-GET commands to retrieve the data and then assemble it for viewing.

In a live-OTT single client-server model, this works well as the viewers' playback device will be sending HTTP-GET commands to the web-server approximately once every second to retrieve consecutive chunks of video and audio data. However, life gets interesting when the number of people watching the event increases to national and international volumes. If 10 million people are watching the event, then 10 million HTTP-Get requests will be sent every second.

Scalable Systems are Critical

Before the IP revolution, broadcasters had to predict audience and viewing capacity requirements years in advance during a system installation. Although SDI broadcast systems have proved their worth and reliability, they have resulted in proprietary, rigid, infrastructures, that were incredibly expensive to build and largely only applicable for video and audio delivery.

IT scaling allows us to build a template system at a relatively modest cost, and then duplicate it as customer demand increases, thus giving CEO's the perfect business model where profits are directly proportional to operational costs. This model is available in live-OTT using distributed streaming-servers.

Quality of Experience is Paramount

Many modern streaming architectures now include just-in-time packaging (JITP). JITP consists of a resilient origin-server that hosts the live content, and a multi-CDN approach with optimized caching layers that scale on-demand as audience size grows. Robust monitoring systems are placed throughout the delivery chain to enable broadcasters to proactively guarantee viewers' quality of experience (QoE).

An added benefit of the HTTP model is that it allows broadcasters to gather statistics directly from all viewers enabling them to not only improve QoE, but also better target and evaluate monetization opportunities.

Viewers are on the Move

Multi-platform viewing introduces new challenges as consumers watch events on mobile devices. By their very nature, cellular phones and notepads are moved from location to location as viewers go about their business. In doing so, the reliability and quality of the network varies. Watching a live event on a cellular phone while on a train journey is a good example of this.

As a viewer moves between cellular networks, environmental conditions cause variations in the phone signal resulting in periodic changes in data rates and latency. Without intervention, viewers would suffer a poor QoE as the phone-app would be constantly buffering data. The picture would stutter, sound would break up, and the "please wait" icon would be regularly displayed.

To resolve this, most modern streaming platforms deliver adjustable bitrate (ABR) packages of content. As HTTP requires the video streams to be packetized, streaming platforms take advantage of this and provide multiple bit rates for each program. Manifest files are periodically sent enabling playback devices to choose the most optimum bit rate for the quality of network available to them.

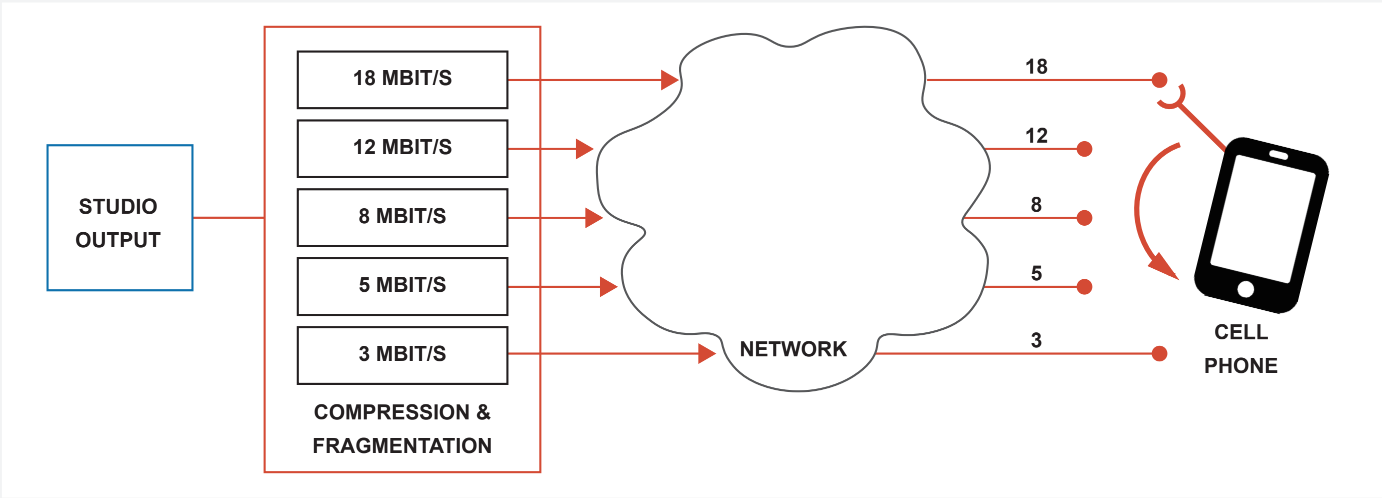


Diagram 3 – The viewers device uses its own algorithms to determine which compression stream to use to provide the best QoE.

DASH to the Rescue

Dynamic Adaptive Streaming over HTTP (DASH) is a relatively new system that provides optimal multi-platform and multi-data-rate viewing. DASH is codec agnostic and relies on the viewers device to adopt strategies to effectively switch between differing data-rates and provide the optimum QoE.

In a typical deployment, a broadcaster might provide five video and audio compression outputs to deliver data-rates of 18, 12, 8, 5, and 3 MBits/sec. These would be continuously made available to the origin or CDN, who would in turn make them available to the viewers device. Software running on the playback device interprets the manifest and programmatically identifies the best data-rate to use given the network conditions.

If the viewer was watching a game on a cellular phone in an empty coffee shop they might achieve WiFi coverage at 8Mbits/sec, but as more customers enter and use the WiFi, it might decrease to 5Mbits/sec. With DASH, their playback software identifies the reduction in data-rate and dynamically switches to the lower compression on the fly. The switch is bound to “key-frames” so the user would be unaware the change had taken place. The opposite happens as customers leave the shop and higher data-rates become available.

Keep New Viewers

Regardless of how a transmission reaches the viewer, and the device they are viewing on, any degradation of QoE will be blamed on the broadcaster. Customers are completely unaware of the route the game they are watching takes, and nor do they care. All they know is that they couldn’t watch their event if something goes wrong in the supply chain.

It’s imperative that broadcasters not only improve their offering through OTT and multi-platform devices, but also invest heavily in their monitoring, at all levels. HTTP systems provide unprecedented access to user viewing habits, network reliability, and QoE. And all this information can be harvested by the broadcaster using appropriate monitoring tools to deliver unified OTT transmissions to keep their existing viewers and gain new audiences.

Part 2 - OTT Technology

In Part 1 of this series, Challenges, we introduced the basic concepts of the technology behind live OTT delivery. In Part 2, we dig deeper to help broadcast engineers and technical managers understand the intricacies of HTTP and IP technology, so they will be able to design and support OTT systems more effectively.

HTTP (Hyper Text Transfer Protocol) has emerged as the dominant technology for internet networks, it scales effectively, and is supported by ISP's and client devices. HTTP messages generally operate on top of TCP/IP (Transmission Control Protocol/Internet Protocol).

IP protocol does not have any form of receipt validation, so the sender has no way of knowing if the receiver has received the packet. However, for the vast majority of internet traffic, TCP is the Transport layer and IP is the Network layer. TCP generally provides reliable transport and maintains "state" in the connection, enabling retransmission if packets have been lost or corrupted during transfer.

Software Ports

The TCP protocol initiates a virtual connection between the playback device and server and identifies the type of connection required using a port number. IANA (Internet Assigned Numbers Authority) governs issuing of port numbers for specific services. Ports are a logical construct and used by the IP-stack software running on a server to efficiently determine the service type contained within the IP datagram.

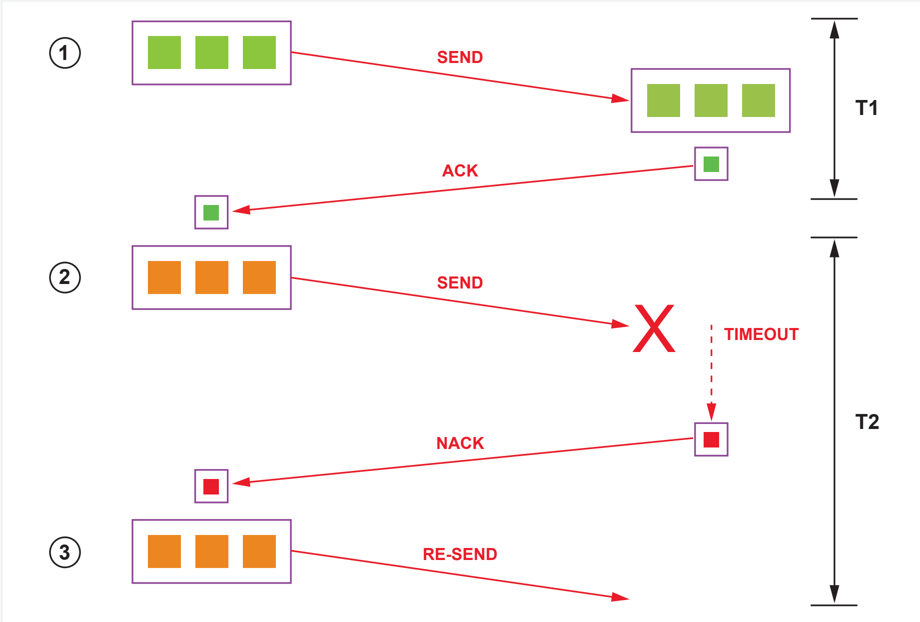


Diagram 1 – This diagram shows a simple TCP exchange. Part 1 sends a window of datagrams from the sender on the left to the receiver on the right, the receiver responds by sending an "ACK" message (for acknowledge). The sender then transmits the next window of datagrams. However, part 2 shows there has been a break in transmission. If only part of the message is received the receiver will time-out and then transmit a "NACK" (for not acknowledged) back to the sender, who responds by re-transmitting the same window of datagrams again. This demonstrates that the time taken for a transfer without error (T1) is significantly less than a transfer with an error (T2) due to the value of the timeout and resend of the packet.

To maintain compatibility, IANA define many port numbers for different services, for example, HTTP uses port 80, Network Time Protocol is port 123, and SNMP is port 199. A server receiving IP traffic will monitor the port number within the TCP or UDP header and send it to its service handler without having to decode the entire TCP payload data. This makes server handling of IP traffic much more efficient as the actual HTTP message is only decoded once in the software service handler.

In the case of an OTT delivery, the web-server streaming the video has a software service that listens for HTTP messages and responds with the data requested by the client player device. Although the web-server will keep track of the TCP status, it does not keep track of where in the stream each client player device is. Even in a live OTT delivery, each player device is asynchronous and will request a slightly different part of the stream from the web-server.

HTTP is very useful for delivering a contiguous feed of video, as out-of-order or missing video presents significant challenges for playback and will degrade the viewer experience. However, there is overhead associated with TCP that can lead to increased network traffic and increased latency between the streaming-server and player device.

Other protocols at the transport layer can eliminate much of the overhead, reducing traffic and latency, but may introduce problems with data integrity or may not be a suitable protocol for transiting the public internet.

HTTP is Ubiquitous

HTTP live streaming formats delivered over TCP/IP make up the vast majority of internet video streaming traffic, but it is important to note that there are a number of important innovations that are currently in progress that may change this, and some other proprietary video deliver formats are being replaced.

Other protocols do exist such as RTMP (Real Time Messaging Protocol) and webRTP (web Real-Time-Protocol). Traditionally, RTMP was used for Flash based viewing experiences, but its use has declined as delivery networks have increasingly looked to leverage a common infrastructure for all internet traffic. Furthermore, Flash has been deprecated from many viewing environments in recent years.

UDP (User Datagram Protocol) based delivery protocols can be leveraged to eliminate some of the latency that can plague OTT applications, but to date, these have not been widely implemented. UDP presents difficult scaling challenges for most traditional broadcast media delivery. And for the foreseeable future, HTTP delivery will likely represent the lions-share of live video streaming.

Natural Limit of Services

IP datagrams are presented to a streaming-server through a limited number of physical connections, usually Ethernet. Consequently, each IP packet will be processed in turn and due to the limitations of the hardware, only a certain number of HTTP messages can be processed within a fixed time. Therefore, there is a natural limit on the number of player-devices that a single streaming-server can service.

One solution is to increase the network speed into the streaming-server and the hardware resources available such as CPU cores, memory and disk space. But at some point, this will also reach a natural limit and is simply not scalable.

Another option is to load-balance the HTTP traffic. A load-balancing device directs HTTP streams to a series of streaming-servers, all containing the same content, but giving the impression to the player that it is only communicating with one server. Although this method makes the servers scalable, it is still limited by the network capacity going into the datacenter servicing the OTT streaming service.

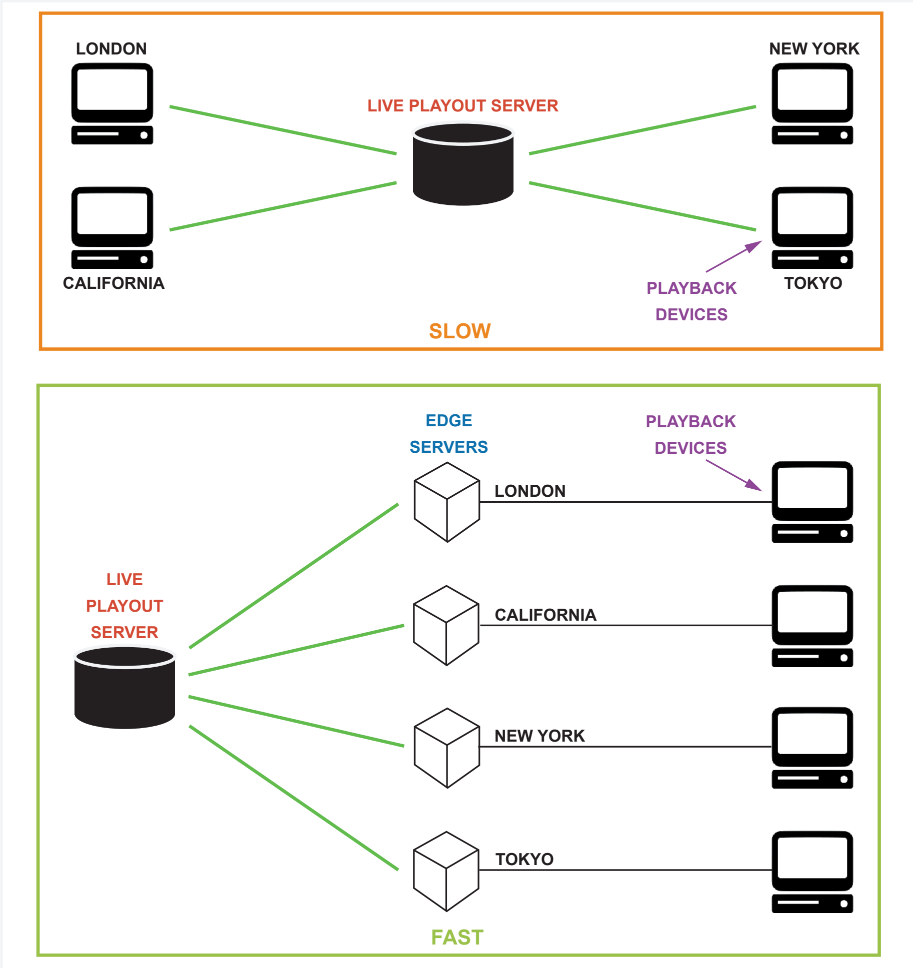


Diagram 2 – The top diagram demonstrates the traditional "slow" method of media streaming. A centralized server will maintain HTTP/TCP/IP connections with many playback devices throughout the world causing the server and network to soon reach the limits of their capacity. The lower diagram demonstrates a CDN system where edge-servers are distributed throughout the world closer to the playback devices. There is less load on the live-playout-server as fewer devices demand data streams from it, TCP latency is lower as ISP backbone networks can be used, and more edge-servers can be added as audiences grow to achieve greater scalability.

Distributed Servers

A third option is to distribute servers containing the same media closer to the location of the player devices. This has two advantages; firstly, the model is completely scalable, and secondly, latencies associated with TCP distribution are greatly reduced.

For on-demand streaming the distributed server model works well as movies can be sent to the distribution servers ahead of publication. But for OTT live delivery, the streaming servers act as relays and will buffer the live stream from the central playout server adding a layer of latency.

This method of distribution, that is one that caches requests and fulfills responses closer to the client, is exactly what content delivery networks (CDN) provide, with extensive edge cache facilities designed to handle internet scale http delivery.

Leveraging CDN's and content optimized HTTP cache strategy is a fundamental component of delivering internet scale video to OTT audiences, and represents one of the biggest factors that can impact streaming costs and the viewer experience.

Many modern architectures now include just-in-time packaging (JITP) on a resilient origin streaming server hosting the live content. This is a multi-CDN approach with optimized caching layers that scale on-demand as audience size grows.

Robust monitoring systems placed throughout the delivery chain quickly identify which service providers are responsible should there be a degradation of service. Multi-CDN's usually rely on multiple service providers collaborating to provide a seamless and enhanced viewing experience for the audience.

Just in Time Optimization

With JITP support, content owners can dynamically structure the package formats that are delivered to the audience, enabling stream optimization, content personalization and protection schemes, and other responsive late-binding workflows to create a more engaging viewer experience.

Hosting the content origin can enable a more cost-efficient way of delivering the same programming through multiple network paths and provides the broadcaster with deep analytical data for content and network performance.

Implementing a multi-CDN strategy ensures that the broadcaster is not reliant on a single delivery path to reach the audience. This enables the broadcaster equipped with the right analytics the ability to dynamically move audience members in to the most cost and performance optimized delivery path.

All these strategies represent a challenge that require deep reporting and real-time analytics. These strategies enable the broadcaster to measure how changes in the production and delivery chain impact audience quality of experience and can be immediately optimized to create the most compelling viewing experience possible.

Adjustable Bit Rates

Viewers on the move experience differing network data rates and quality as they transfer between cells. Therefore, most modern streaming platforms deliver adjustable bitrate (ABR) packages of content. HTTP streaming creates short segments of video and sends them to the player device in bursts. But if these segments are replicated, then they can be transcoded at different bit rates and resolutions.

ABR provides a range of encoding and processing settings to produce packaged content optimized for playback across a wide array of device and network playback conditions. ABR packages are presented to the player using a manifest file that acts as a playlist, telling the player where it can go to retrieve the next segment.

Algorithms running in players detect when their buffers are running low, so the player can opt to an adjacent bitrate variant. Other approaches include the ability to identify only those variants that are ideally presented on a given device, or in each network condition. For example, if my mobile device is on a cellular network, I may choose to omit the highest resolution variants as this would not improve the viewing experience while jeopardizing smooth playback.

HTTP plays an important role in distribution of OTT over the internet. Even though there is currently much research for an alternative, HTTP is going to be the prominent distribution method for the foreseeable future. Multi-CDN builds on HTTP to provide scalability and help broadcasters deliver a better viewer experience. In Part 3, we will look at monitoring and why it's critical to OTT and multi-CDN systems.

Part 3 - Monitoring

Throughout this series of articles, we've been investigating the detail of how OTT internet delivery works. In Part 3, we dig deeper into the operational systems, and investigate the additional benefits and necessity of monitoring.

Live television delivery is the key feature for broadcasters and content developers looking to differentiate themselves from on-demand services. Terrestrial, satellite, and cable operators have all benefited from low latency delivery provided by point-to-point and radio frequency mechanisms.

However, consumer viewing habits have changed, and audiences now want to watch their favorite programs on a host of devices from mobile phones to smart TV's. Although the internet is the most convenient delivery platform for viewers, it presents new challenges for broadcasters.

Viewers will Switch

Audiences now have greater choice and are more discerning. If their live sports event is breaking up or distorting, they are more likely to find an alternative broadcaster or streaming service to switch to. And the viewer may even go straight to social media to vent their anger. Consequently, broadcasters need to monitor the whole program chain from playout to device delivery.

As viewers move from linear television to streamed live events, new methods of monetizing content are being established. Pay-per-view events may seem the obvious solution, but even these have their challenges and customer viewing must be closely monitored.

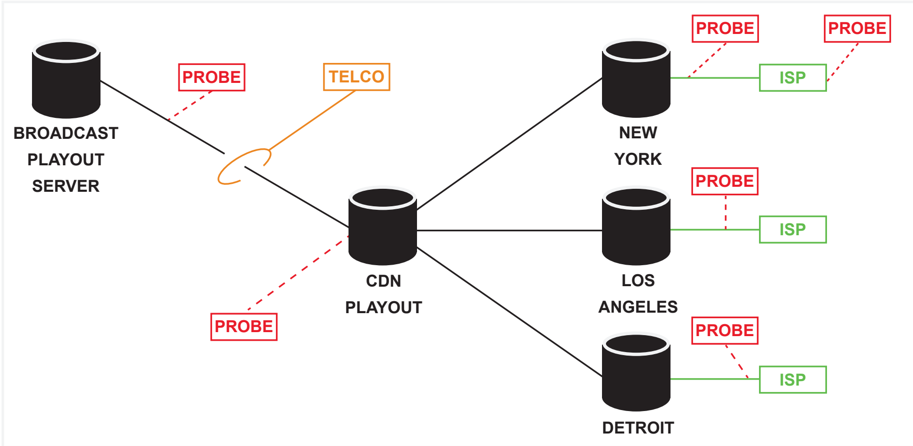


Diagram 1 – Placing monitoring probes throughout the OTT distribution chain will allow broadcasters to gather a mine of information about the integrity of the viewing experience as well as viewer trends and habits for marketing and direct advertising.

QoE is Paramount

OTT monitoring needs to go much further than the standard video and audio level tests to keep viewers engaged with their service. Delivering IP media packets to a mobile device is not enough as broadcasters must be sure the viewer is receiving a good QoE (quality of experience).

Delivery networks usually consist of many different service providers. An ISP for the last mile, a telco to interconnect to the broadcasters' playout center, and a CDN may provide a network to connect the viewers last mile to the broadcaster's playout server. Each of these can be independent companies and have their own interests at heart.

Once a program has left the playout facility and the video and audio has been quality tested and proved to be in-spec, the next two parameters critical to a good QoE are latency and bit rate.

HTTP/TCP provides greater resilience over the internet but at the potential cost of increased latency. If an IP link is dropping packets then it may not affect the integrity of the data as any lost packets will be resent, but it will increase the latency as resent packets need to be buffered prior to transmission.

Such a condition can occur at all three service providers or at any point within the CDN network, resulting in potential differential latency between regions.

Strategic Monitoring

Ideally, monitoring probes should be placed at the three points of demarcation of each service. Monitoring probes on the output of the playout center can measure latency and bit-rate as the packets make their way to the central CDN playout server. And placing probes at the output of each CDN regional playout server will help determine if excessive latency has occurred.

As well as providing monitoring data to help locate errors, probes can be used to provide QoE information to allow broadcasters to fine tune delivery and give audiences the best possible viewing experience.

OTT delivery is a fine balance between streaming quality and playback device buffering size. When moving around, the quality of data service provided varies enormously, especially when travelling.

Mobile devices switch between cell transceivers and even WiFi if the service comes into range. This initially causes a disturbance to the received data stream causing some loss of data. Any dropped segments will need to be resent by the playout server.

Buffer Compromise

The length of the buffer employed by the mobile device is designed to smooth out these transitions so there is no loss in service to the device media playback streamer and therefore a high quality of service is maintained for the viewer.

The disadvantage of buffering is that the viewer must wait until the buffer reaches its playback threshold and this could take many seconds. The effect for the viewer is to see the "downloading" icon on top of a freeze frame of the live event they are trying to watch.

Congestion, buildings, and other mobile device users all conspire to reduce the quality of data streams available and force the mobile device to switch between carriers and cause fluctuation in data rates. But DASH (Dynamic Adaptive Streaming over HTTP) goes a long way to fix this as the playback device can constantly monitor the available bit rates and move seamlessly to the best available rate at that time.

Strategically placed probes, especially at the end of each regional CDN, allows broadcasters to analyze and determine the best media bit rates and segment sizes available to the mobile device. Too many available bit rate streams will cause unnecessary switching in the playback device and is wasteful of bandwidth. Not enough will cause picture disturbance and defeat the concept of DASH.

Segmentation sizes greatly influence the QoE. If they are too large then there will be increased latency in the distribution network and the playback device buffering will be too large, causing long delay times between the live event and the viewer seeing it on their device. If segmentation sizes are too small, then the latency will be lower, but the playback device buffering will be lower too, and if it's too small then picture disturbance will result as the viewer travels.

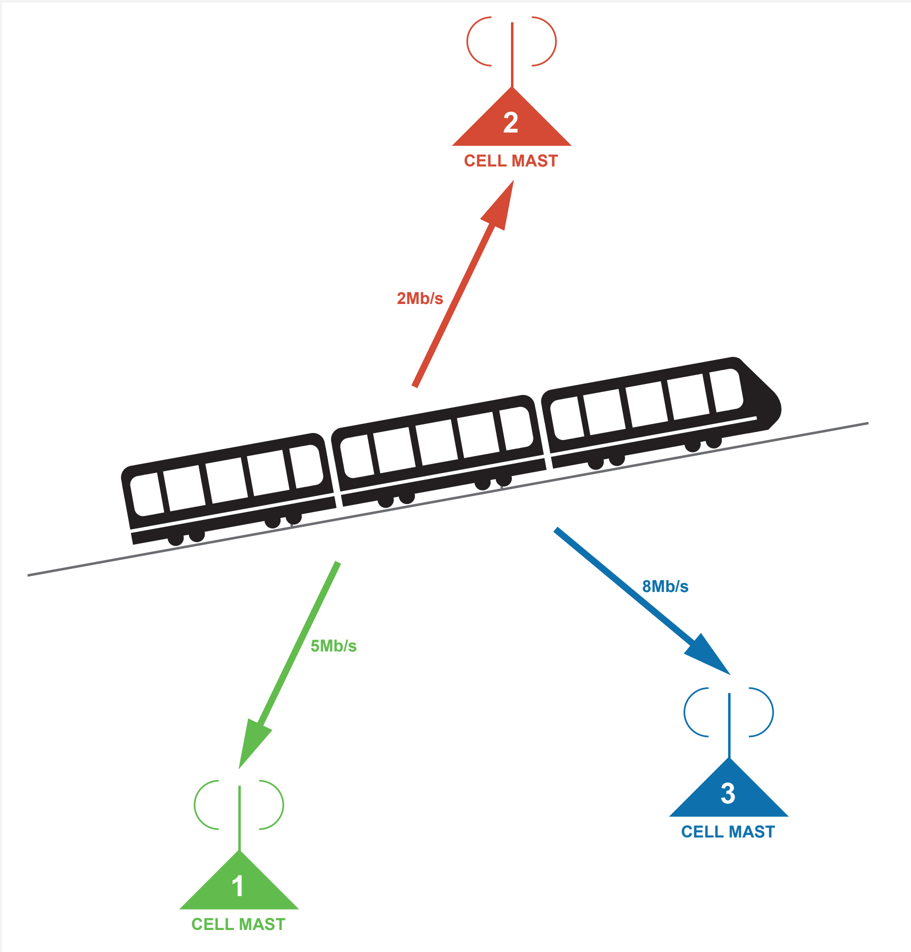


Diagram 2 – Mobile playback devices on a moving train will switch between cell masts 1, 2, and 3, as the train travels. DASH will switch between the best data rates available giving the viewer the best QoE experience possible.

Broadcasters will increasingly make live event special promotions available on pay-per-view OTT. To achieve this, the viewer will log onto a website, enter their credentials, and validate their payment method. The playback device will receive a decryption key allowing its DRM (Digital Rights Management) software to display the content.

Validate Viewers

Audiences tend to start viewing close to the beginning of the event and this could result in thousands, if not millions of viewers, all validating their playback devices at the same time. This would result in significant congestion just at the beginning of the game resulting in many viewers not being able to receive the first few minutes.

Although this is technically not a broadcast issue, it still falls under the remit of QoE and if the viewer is not satisfied with the service as they cannot stream the game, then they will go elsewhere. This is as serious as the video breaking up or there being no audio.

Consequently, QoE is part of the media streaming process and broadcast engineers will need to collaborate with their IT colleagues to understand how to rectify these challenges. Remedies available include allowing viewers access to the first ten minutes of the game and then staggering validation in the background across all playback devices or spinning up more validation servers at the beginning of the game to take on the extra anticipated load.

Unlike traditional broadcasting, OTT quickly provides the necessary data to make these decisions possible empowering broadcasters to dynamically optimize systems to deliver the best QoE possible.

Simplify with Orchestration

Orchestration has been developed to bring this altogether and simplify OTT delivery. It provides a schematic overview of the network (especially at the demarcation points between different network suppliers), consolidation of the monitoring probe data, and control of key parameters in encoding and segmentation.

Broadcasters can even switch between CDN or telco providers on-the-fly if they do not think they are performing well. And they can analyze their data with the service provider to find out where bottlenecks, congestion, and lost packets have occurred to pro-actively keep systems optimized.

As well as providing engineering data, broadcasters now have virtually instant information about the viewing habits of their audiences and can gain a better individual understanding to personalize advertising.

OTT requires more than making sure the media signal has reached the viewer intact. It encompasses the whole quality of experience of the audience to make sure their complete viewing experience is satisfied, they stay watching, and come back again.

Telestream

The Sponsors Perspective

The Paths to Streaming Distribution: A Tale of Two OTT Use Cases

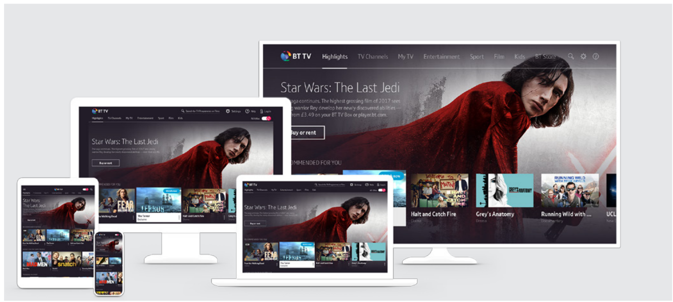
The consumer video market is growing fiercely competitive between content creators and content aggregators and OTT live and OTT VOD formats are growing increasingly fragmented. Consumers are benefiting from the tremendous choice of content and sources for that content while their tolerance for a negative Quality of Experience (QoE) continues to fall. Providers find themselves in the stark reality of having to balance getting to market faster, with greater unit cost efficiencies and with persistent QoE.

The following examples highlight how two leading players have effectively balanced these three business priorities. Each example illustrates the operational challenge, the execution approach, and resultant business impact. Each example furthermore focuses on a different OTT use case: one on multiscreen VOD and the other on OTT live streaming.

Delivering on the Seamless Multiscreen Experience

BT TV, a subscription IPTV service offered by the UK's largest telecom service provider, BT, has adopted Telestream's Vantage Media Processing Platform. This move creates a robust, high-speed multiscreen OTT media processing solution with automated quality assurance to better serve millions of their customers.

The company evaluated all the available transcoding options and found that Vantage offered high quality content in the widest range of multiscreen formats, with media processing times that are significantly faster than any other platform. "Vantage contributes significantly to our business agility and the fast turnaround times of our multiscreen files allowed our new App and Web product to launch on time …." explains Peter Harvey, Head of Content Operations (VOD and Digital Media) at BT Technology.



Employing Vantage has allowed BT to launch enhanced features into the BT TV APP & Web player, enabling customers to access multiscreen versions of any rented or purchased content with the minimum of delay. The project supports the BT TV APP across iOS and Android devices as well as Amazon tablet. This project was fueled by BT TV's need to enhance its business agility and support business development goals focused on its OTT VOD operations. The decision to integrate Vantage into BT TV's video infrastructure was made to enable the shortest possible delivery times for multiscreen formats, which was paramount due to the time constraints of the project and large volume of content to be processed.

Vantage is being used to analyze the technical properties of VOD content, extract and create appropriate metadata which is used to drive Vantage workflow decisions and to transcode VOD content into high quality multiscreen formats for OTT delivery. "The Vantage Workflow Designer has proved popular with our team as an easy to understand, yet powerful and dynamic toolset. Vantage has given greater control of our VOD processing, reducing manual effort and helping the team to work more efficiently. It has reduced multiscreen media transcoding times by up to 30 percent, which was key in selecting Vantage," commented Peter Harvey.

Achieving World Class Results in OTT UHD Live Streaming

During the 2018 FIFA World Cup, owners of certain Hisense 4K HDR connected TVs who downloaded the UHD version of the Fox Sports GO app were able to watch every one of the World Cup games and watch in UHD 50p with HDR color. This was made possible by an exclusive agreement between Fox Sports and Chinese TV manufacturer Hisense.

Prior to the start of the World Cup, Kevin Callahan, Fox Sports vice president of field operations and engineering stated, “For the most popular sporting event in the world, it’s a natural that some of our viewers would jump at the chance to see World Cup matches in the brilliance of UHD/HDR. Fox Sports’ goal is to offer our audiences the most extensive and most engaging coverage of their favorite sports. We were excited to work with Hisense to offer live streaming UHD/HDR content. The real challenge was ensuring that the viewing experience was seamless.”

Connected directly into Telestream Lightspeed Live where it is encoded to the exacting standards required by the Fox Sports Go app and Hisense, a special wide color gamut UHD video feed, via SDI from Host Broadcasting Services (HBS) enabled the unique UHD 50p HDR 10 stream. In this use case, that standard is an HLS package containing a Main 10 profile—level 5.1 HEVC/H.265 compressed bitstream with HDR 10 metadata. As one might expect, the quality was brilliant.

For the UHD/HDR stream, the resultant Lightspeed Live HLS package was delivered direct to Akamai where it was distributed to the Fox Sports Go app on Hisense TVs. “We were confident in Telestream’s ability to take on a production grade UHD/HDR live stream, and the results bear that out,” says Clark Pierce, senior vice president of TV Everywhere and special projects, digital technology and integration, for Fox Sports. “The ability to work in HDR10 makes an enormous difference in both quality and performance.”

Latency is always a challenge when encoding live events. Using the Lightspeed Live, the FIFA World Cup UHD feed had incurred no additional latency, and in fact, was frequently ahead of the standard HD feeds being delivered. Telestream has been able to confirm a smooth and efficiently-delivered playback experience by using Telestream IQ probes to monitor performance across the delivery chain.

More than 4.2 million people visited various Fox Sports digital platforms on day 2 of the 2018 FIFA World Cup, a 77 percent increase from day 1. Spain vs. Portugal, for example, attracted 819,000 unique viewers. This year’s event has seen a confluence of technology coming together to reap new benefits for consumers and broadcasters alike. The event has demonstrated that organizations can deliver UHD with HDR to TV platforms over-the-top more efficiently with HEVC codecs. The number of total viewers that have UHD TV sets is growing, but more importantly, the number of avenues to receive 4K/HDR content OTT via apps is growing. The event has also provided a valuable proof point demonstrating that such high-resolution images can be provided without incurring cost burdens due to excessive data rates. Using the same technology for traditional HD streams, costs are proportionally lower which can further offset UHD production and delivery costs.

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