A protocol for live transmission over the internet

Conventional point-to-point connections for the distribution of live media content via satellites or content delivery networks are expensive and have several disadvantages. The Rohde&Schwarz subsidiary GMIT offers a proven solution for transmission over the public internet that is far more cost-effective and flexible.

If media providers want to achieve long-term success, they must provide their audience with appropriate content as quickly and inexpensively as possible. The transmission paths to customers are a source of savings potential. Thanks to increasingly high online bandwidths between continents, data centers and end points, the public internet can be used to broadcast live events, provide TV content to a wide audience and add new programs to online TV services.

A distinction is generally made between content contribution and content distribution (Fig. 1). In both cases, technical obstacles need to be overcome in order to reliably broadcast live content over the unmanaged internet around the clock with high bit rates and minimal delays. This level of performance has only recently been achieved.

The internet protocol is everywhere

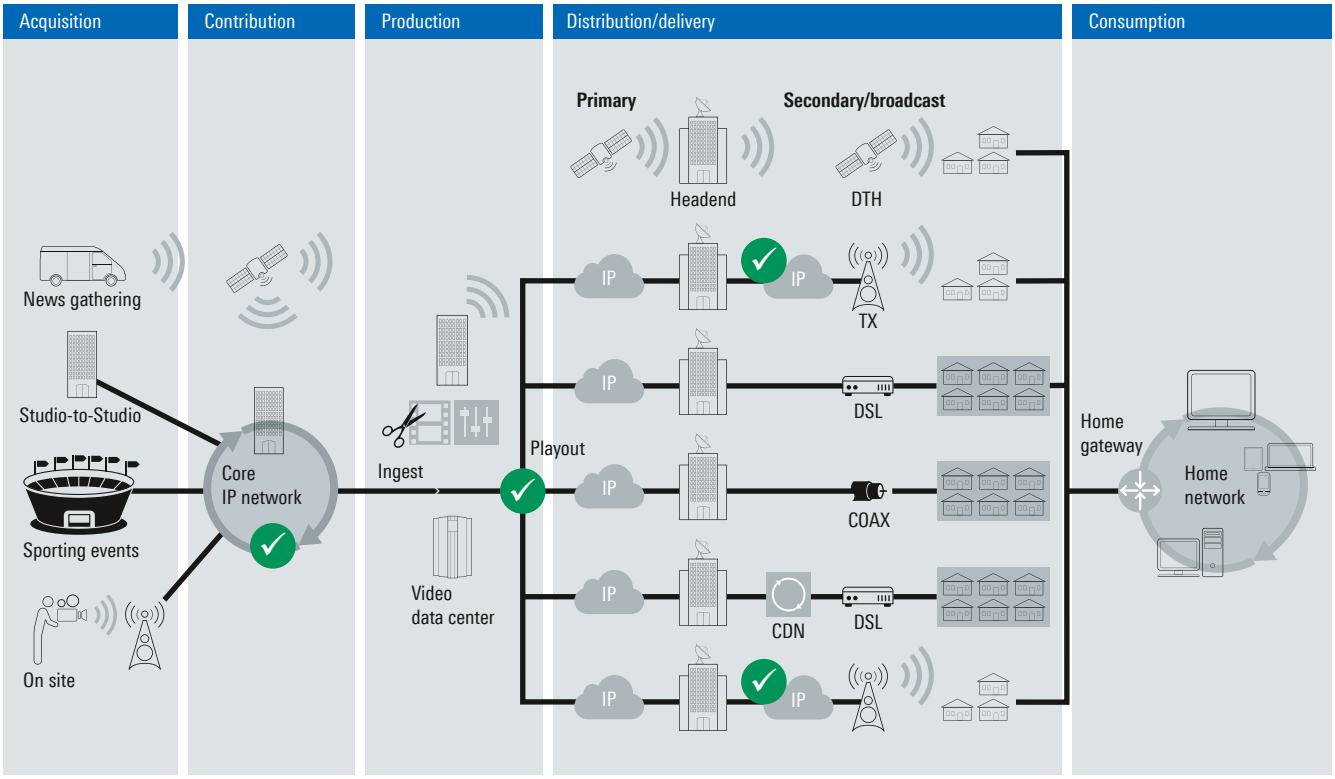

Fig. 1: From acquisition to consumption: IP contribution and IP distribution take place at all media production interfaces.

Conventional point-to-point connections

Teleports

For a long time, satellites were used to feed and distribute live content over long distances. Signals are sent to satellites via teleports and then forwarded to other teleports before being distributed via dedicated, managed optical networks. This process has the disadvantages of very high initial/operating costs and a lack of flexibility when establishing new connections.

Content delivery networks

Content delivery networks (CDN) are another way to distribute content. These networks consist of internet servers in multiple data centers. The size of the network depends on the provider. The servers are connected over the public, unmanaged internet or over managed virtual overlay networks, or in some cases over private optical networks that offer better quality of service. Content is mirrored, i.e. identical copies are stored on these servers. Conventionally, CDNs mirror entire websites by copying their content to servers in different regions. When users access a website via an internet browser, the content (text, graphics, etc.) is delivered by one of the CDN servers closest to the user. Closest refers to the distance on the internet, which is measured in round trip time, for example.

HLS streaming in CDN

In recent years, many established CDN operators have also entered the streaming market. The de facto standard for smartphones and smart TVs is the HTTP live streaming (HLS) protocol developed by Apple, which splits on-demand video content and live streams into "chunks" consisting of MPEG-TS files, each of which is a few seconds long. HLS can handle adaptive streaming. The content is available at various bit rates. The streaming client chooses the most suitable resolution for each chunk in accordance with the bandwidth available at that moment.

Since HLS is based on HTTP, the familiar protocol used to transmit websites to browsers, internet streams can be received in the same way as other web content, even behind firewalls. This is also known as over the top (OTT) because it uses the existing internet infrastructure to distribute audiovisual content – at least over the last mile, i.e. the path to the end user's streaming client.

Disadvantages of CDN

CDNs are currently the most widely used technology, but they have disadvantages. In addition to high cost, one of the main problems is their high latency. Due to content mirroring, many CDN providers incur a delay of 30 and 60 seconds for live streaming with HLS. Very few providers can achieve a latency of only a few seconds, something that was possible with satellite transmissions.

Another disadvantage is that customers themselves are largely responsible for feeding their live streams into the CDN themselves. The content also needs to be reliably transmitted to the nearest network access point. In countries or regions where the CDN cannot offer an access point in a nearby data center, providers will need to revert to a satellite connection or a dedicated optical network.

Purely internet based infrastructures can be problematic

The omnipresent internet with its constantly increasing bandwidth is virtually the only option for transporting media content. The problem is that due to the purely packet based and connectionless nature of this medium, universal quality of service (QoS) cannot be ensured. Packet loss during the handling of internet traffic by routers, and even brief stream dropouts due to routing changes, are common. Every missing data packet causes picture distortion or audio gaps, and every brief dropout causes an interruption in live playback.

The solution: RelayCaster – only available from Rohde&Schwarz

The goal of streaming specialist Motama GmbH was to compensate for the erratic nature of the internet with intelligent processes. The company was a pioneer in this field and presented its RelayCaster product at the 2010 IBC international trade show in Amsterdam. The Rohde&Schwarz Berlin based subsidiary GMIT GmbH acquired the Motama technology in 2017, thereby expanding its product portfolio, particularly in the area of interference-free transmission of audio and video content over IP networks. Rohde&Schwarz has since revamped the product, which is now known as R&S®RelayCaster. The enhanced version allows feeding and encoding of SDI/HDMI data streams. This integrated functionality further improves the cost-efficient platform for transmitting live content.

The idea: a unique protocol

The UDP and TCP protocols dominate online. While UDP may be excellent for transmitting live streams, it is inherently unreliable. This leads to packet losses with intermittent audio dropout or video errors (block artifacts) that can range from minor to serious.

TCP, on the other hand, is 100 % reliable, but that is also why it is unfortunately not suitable for transmitting bandwidth-hungry live streams over large internet distances (long round trip times and many internet hops). Even minor packet losses cause TCP to decrease the bandwidth, causing the live stream to freeze. This problem occurs with all TCP-based protocols such as HTTP, and therefore streaming protocols such

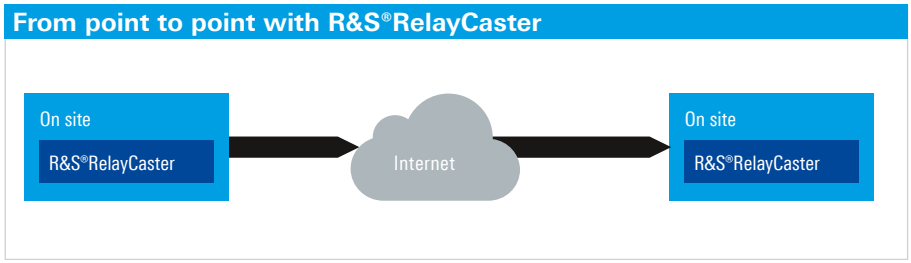

Fig. 2: An R&S®RelayCaster server transmits a live data stream via the public internet to an R&S®RelayCaster receiver. A special protocol ensures reliable transmission and low latency.

as HLS. It causes HLS to switch to a lower bit rate in the case of packet loss, reducing audio and video quality.

The solution from Motama uses two R&S®RelayCaster instances (Fig. 2) for live, error-free transmissions online. The unique R&S®RelayCaster streaming protocol (RCSP) is used between the two instances. One instance sends the live stream from the local network via public internet to another R&S®RelayCaster instance at any internet-accessible location. The receiver forwards the live signal to its own local network, where e.g. a terrestrial broadcast distributes it to the content takers (B2C in Fig. 4).

RCSP solves problems such as packet loss in unmanaged areas of the internet. It can bridge large internet distances, compensate for packet loss and is fully internet-compatible. Optional encryption with the Advanced Encryption Standard (AES) keeps content secure.

RCSP is based on UDP and achieves its quality of service by combining several methods, such as optimized retransmission of lost packets. The protocol combines the good characteristics of the two dominant internet protocols UDP and TCP and operates with a very low latency of about 1 second.

The QUIC protocol developed by Google is also based on UDP, but unlike RCSP where real-time capability takes precedence over reliability, it is designed for absolutely reliable data transmission.

RCSP can transmit a single stream or several streams with any desired bandwidths, provided that the input and output bandwidths are available at the transmitter and receiver, respectively. Based on experience, a margin of 20 % additional bandwidth should be planned in to accommodate strong fluctuations in line quality.

Various versions

Various versions of R&S®RelayCaster are available, which allows almost infinitely scalable networks to be set up. In addition to several server types (Fig. 3), there is also an inexpensive, portable embedded device version and pure software versions for use on leased servers in external data centers and on virtual machines of cloud providers such as Amazon AWS and Microsoft Azure (Fig. 4).

Hybrid solutions that combine existing transmission technologies with new internet based contribution and distribution solutions are a good idea for many projects. Sending

Fig. 3: R&S®RelayCaster is available as a server solution, as a small embedded device and as a pure software solution.

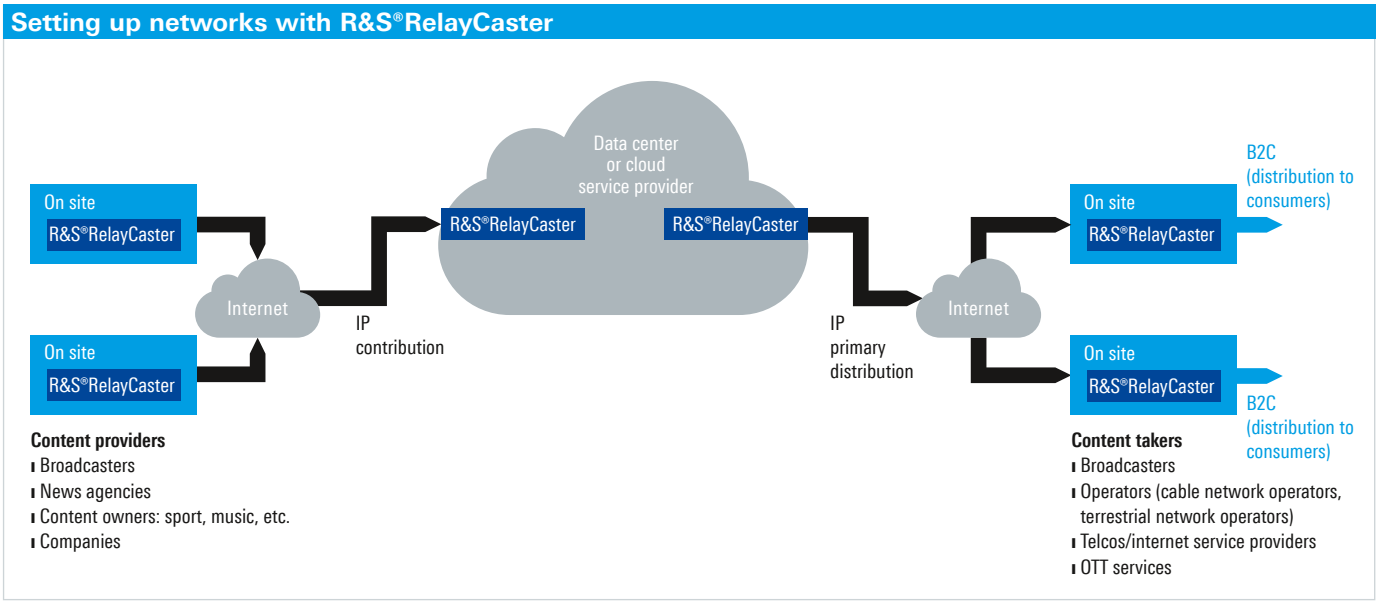

Fig. 4: R&S®RelayCaster can be used to set up extensive contribution and distribution networks that connect content providers to content takers and are almost infinitely scalable thanks to cloud based resources.

live content to satellite teleports and bridging the gap from teleports to CDN feed points are just two of many possible applications.

Summary

Today, the mature internet based IP transmission technology offers a stable and considerably more affordable alternative to conventional dedicated point-to-point connections for content distribution. R&S®RelayCaster fulfils the QoS conditions required for interruption-free transmission via public, unmanaged IP networks. IP-based transmission means that users are no longer reliant on expensive satellite connections, dedicated connections or CDN providers. R&S®RelayCaster is not limited to specific regions or data centers. The solution enables content to be distributed over any distance and to any location that has an internet connection. It is ideal for producers and live content aggregators as well as providers of live content streaming and OTT streaming. This innovative platform allows users to set up their own flexible, scalable content delivery networks and to utilize the internet to reduce their operating costs to an unprecedented extent.

Rohde&Schwarz with its comprehensive broadcasting portfolio, cybersecurity expertise and global presence is a reliable partner in this new, universal IP-based era.

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