
heynen
Heynen works for innovators

How To Deliver a Broadcast-Grade STL over Consumer-Grade DSL

Thinking of Changing to IP?

IP (Internet Protocol) is fast replacing traditional RF and synchronous type links (T1 / E1, ISDN) for both Studio to Transmitter links (STLs) and studio to studio communications. This is chiefly due to the fact that it is significantly less expensive than traditional links and also offers a broadcaster a great deal of flexibility.

However, you cannot simply rip out your T1 and replace it with a DSL tomorrow. IP and mission-critical broadcast audio are not natural bedfellows!

If you send audio content down a standard open internet link, you can expect at least some of the following to occur:

1. A high and variable rate of delay jitter caused by bottlenecks on backbone routers, particularly if they operate very close to their overload condition
2. Loss of single audio packets due to network inconsistencies
3. Loss of a cluster of audio packets, sometimes called a "burst" loss
4. Loss of connection, temporary network failure
5. Packets arrive out of sequence due to variations in network latency

A professional broadcaster just can't tolerate these kinds of risks.

What are the Options?

One way to combat these problems is to use a 'managed' IP network where Quality of Service (QoS) controls can be enforced. On a managed link, the traffic will be regulated and the highest priority given to the audio packets, so that they may arrive at the destination on time and in sequence.

However, managed networks can be expensive to deploy and maintain, particularly when sending audio over some distance.

To truly reap the cost benefits of IP, you need to look at using readily available, easily affordable, consumer-grade IP connections such as DSL, wireless 3G / 4G, and Wi-Fi.

As we saw above, these open internet connections aren't kind to broadcast audio so we need to find a clever way of exploiting the cost advantages of IP without suffering the disadvantages.

Have Your Cake & Eat It!

This 'Clever Way' already exists. APT's SureStream technology is an innovative and multi-award winning approach that is currently enabling broadcasters throughout the globe to transport their broadcast audio content over public IP networks.

SureStream enables you to:

a) Save Money:
You can save up to 90% on the cost of your audio transport bills by replacing your synchronous or managed IP Links with the public internet. With DSL charges typically costing from 5-10% that of a synchronous link, the potential savings are massive and you can generate a return on your investment in under 6 months!

b) Deliver High Quality Audio:
The audio quality of your station should not be sacrificed for the sake of cost savings. SureStream enables you to maintain consistently high audio quality with no drop-outs and no jitter.

c) Keep delay consistent
For professional audio delivery, it is not acceptable for the signal delay to vary or to drift. The ability to maintain the delay at a consistently low level is particularly useful for remote broadcast applications and local content insertion.

d) Relax!
SureStream offers you the same level of uptime and reliability as a 'five nines' Telco service. You are protected not only from drop-outs and glitches but also from a complete loss of connection!

With SureStream, the internet finally constitutes a viable alternative to existing synchronous networks such as T1, E1 and ISDN without any compromise to your station's sound.

Heard it All Before?

This may sound a little familar. There are others who claim to offer the same type of solution but SureStream takes a completely unique approach which combats not just some of the issues of IP Networking but ALL of them.

Other codec manufacturers have implemented various technologies in an attempt to deal with the drop-outs, quality variations or dead air often associated with Audio over IP. They mostly fall into three camps:

1) Bandwidth Scaling
For some, ensuring continuity of service is the key aim so they employ schemes that scale-back the quality of the audio or adjust the delay depending on the availability of bandwidth or the performance of the link. This approach ensures delivery but sacrifices consistent audio quality and consistent delay. SureStream doesn't work like this!

2) Link Switching
A different approach that some offer is called link switching – this is where a codec will monitor two separate links and automatically switch to the one offering the best link quality. However, this assumes that the past performance of the link is a suitable indicator of future performance and leaves the link vulnerable to a Loss of Connection at any point in the delivery. SureStream doesn't work like this!

3) Variable Latency
Sometimes called Elastic Buffering, variable latency causes problems for remote broadcasts, as it will affect natural talk-back with the studio. It also makes it difficult to ensure the timing of local content insertion for studio to transmitter and audio distribution links. SureStream doesn't work like this either!

SureStream is unique in the market and, unlike the approaches mentioned; it does not compromise on any area.

It delivers:

- *Robust and uninterrupted streaming*
- *Low and constant latency*
- *Consistently, High Audio Quality*

All using standard IP links. And not just standard ADSL links but also wireless 3G and 4G, LAN, WAN and Wi-Fi too.

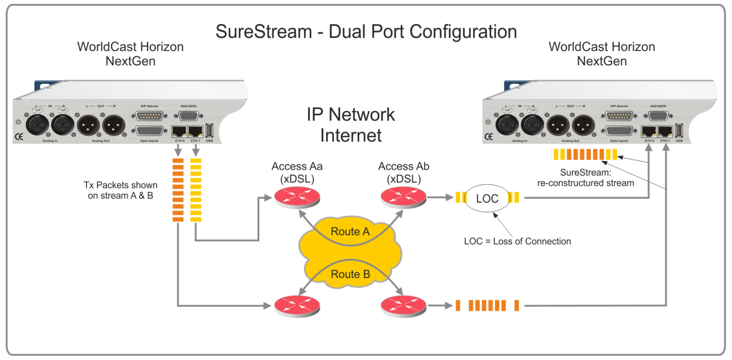
How does SureStream work?

Well, firstly, SureStream capitalizes on the natural behavior of IP packets. The route of an IP packet is unpredictable and will depend on the routers and switches through which it passes. Therefore, if we send two streams from the same source to the same destination, they will travel in very different patterns, increasing the reliability and resistance of the system.

Sending duplicate streams is a good start but not good enough for the kind of perfection we are looking for!

On the encoder side, SureStream employs a number of proprietary techniques that optimize the delivery of all streams throughout the network.

Then, it is on the receiving end where SureStream really works its magic! From the multiple streams received, APT's advanced resequencing engine produces one perfectly seamless, reconstructed stream. Perfect audio from an imperfect network!



In addition, SureStream allows you to configure a buffer level that will compensate for any jitter experienced. Once set, this delay is constant, enabling consistent playout.

Where can it be used?

SureStream works well on a single IP link for impromptu remotes or outside broadcasts but, for mission-critical studio transmitter links and audio contribution or distribution, we recommend utilizing two links from different providers to ensure optimum performance rivalling the a traditional T1 service. With two separate links, you are fully protected not only from network conditions but also any loss of connection.

With SureStream technology, you can save thousands on your audio transport bills and still offer your listeners the highest quality sound.

Want to find out more?

If you would like further information on APT's SureStream technology and our range of IP audio codecs, visit www.surestream.ws or give us a call. We'd love to hear from you:
Miami Office: (305) 249-3110 | ussales@worldcastsystems.com
APT UK Office: +44 28 90 677 200 | info@aptcodecs.com
International Office: +33 557928928 | info@worldcastsystems.com

apt WorldCast Systems | Group

Heynen werkt voor vernieuwers

Hierbij een document van Heynen. Deze producten worden door ons exclusief vertegenwoordigd in de Benelux. Voor een demo of offerte kan u rechtstreeks contact opnemen met onze account managers.

Heynen travaille pour des innovateurs

Voici un document de Heynen. Nous représentons ces produits exclusivement pour le Bénélux. N'hésitez surtout pas de demander votre personne de contacte pour un démo ou une proposition de prix.

Heynen works for Innovators

Here is a document from Heynen. We represent these products exclusively for the Benelux. Please contact our account managers for a quotation or demonstration.

Heynen Leistung für Neuerer

Anliegend eine Dokumentation von Heynen. Wir haben die exklusive Vertretung dieser Produkte für den Benelux-Raum. Bitte nehmen Sie Kontakt auf mit unserem Kundenbetreuer, falls Sie ein Angebot oder eine Vorführung benötigen.

heynen@heynen.com • www.heynen.com •

Heynen bv P.O. Box 10 NL-6590 AA Gennep De Groote Heeze 11 NL-6598 AV Heijen The Netherlands t +31 (0)485 55 09 09 f +31 (0)485 55 09 00
Heynen nv Centrum-Zuid 3047 B-3530 Houthalen Belgium t +32 (0)11 60 09 09 f +32 (0)11 52 57 77
Heynen GmbH Marburgerstrooss 19 L-9764 Marnach Luxembourg t +352 (0)26 91 07 81 f +352 (0)26 91 07 82