NAB Radio TechCheck

The Weekly NAB Newsletter for Radio Broadcast Engineers

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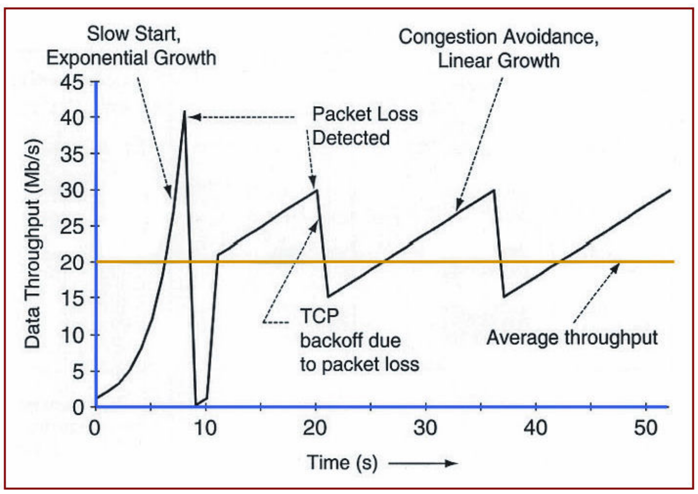
Advanced Technology for IP Remotes

More and more broadcasters are using the Internet as part of their transmission infrastructure. A session at the upcoming NAB Broadcast Engineering Conference (BEC, April 12-17, 2008, Las Vegas, NV – see below for additional information) entitled “Audio over IP” includes a paper by Steve Church of Telos Systems describing the use of IP connectivity for remote operations, which is excerpted here.

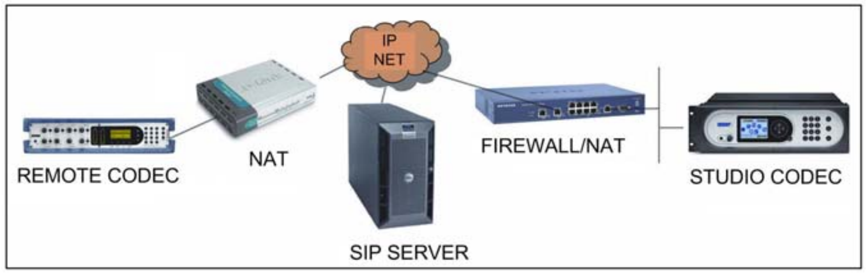
INTRODUCTION – ISDN has served broadcasters well. While ISDN is still a perfectly good technology, it does have some drawbacks. The main one is that usage is billed by the minute. Another is that installation of the line at the remote side usually has a multi-week lead time and has a significant set-up charge. IP networks are becoming the new way to get broadcast audio to here from there. A broadcast codec taking advantage of new technology and optimized for the real-world conditions on IP networks makes this a practical reality.

AN IP-OPTIMIZED CODEC SYSTEM – A broadcast codec intended for IP application needs to be optimized for the purpose. MPEG AAC is an efficient codec, and the Spectral Band Replication (SBR) addition makes it the most efficient within the MPEG family. The downside is it has quite long delay, around 150 ms, meaning 300 ms for a roundtrip, plus yet more for the IP packetization and buffering processes. Too much for interactive two-way conversation. The AAC-ELD (Enhanced Low Delay) codec combines a low delay codec (around 50 ms) with the coding power of plain AAC. AAC-ELD has reasonably good fidelity down to 24 kbps and excellent fidelity when used at 64 kbps and above. At 128 kbps, it is regarded as indistinguishable from the original.



TRANSPORT – Transmission Control Protocol (TCP) solves the lost packet problem via retransmission, but this imposes a delay penalty since the buffer has to be long enough to accommodate the time it takes for the replacement packet to arrive. Streaming audio on the Internet usually uses TCP and there are usually multiple-second-long buffers in the players. That’s why it takes so long for streaming audio to start after you click on the link or play button. When we care about delay, this is not going to do. TCP’s flow control algorithms are also a potential problem, since they can needlessly restrict bandwidth, causing it to vary versus time (see graph). The alternative is User Datagram Protocol (UDP) combined with error concealment. Using UDP, we are as close to the underlying network as we can be, so the delay is as low as possible and the bandwidth as high as possible. Responsibility for dealing with packet loss is moved to the “user,” which is perfectly OK because we can deal with it in a way specialized for our audio application, rather than accepting the compromise of a general approach that was designed mostly for email, Web browsing, and file transfers.

MPLS SERVICE – Multi-Protocol Label Switching (MPLS) service is a telco IP service that is growing in popularity. It is intended for high-quality VoIP telephony, video conferencing, and the like. MPLS networks analyze traffic at the entry point and attach a label to each packet that describes the path the packet should take within the network. Because routers can see the packets as a stream, reserving a specified bandwidth is possible and usual. MPLS services are attractive to broadcasters since they offer a good cost/performance compromise—more expensive than non-guaranteed public Internet service, but less costly than dedicated links or ISDN.



FIREWALLS AND NATS – A typical remote-to-studio setup is shown in the figure and includes one or more Network Address Translators (NATs). NATs are widely used on DSL Internet connections to allow more than one computer on the inside to share a single IP number toward the outside. All connections must originate from a computer on the inside. Since unsolicited incoming traffic can't get through, NATs provide a basic firewall function. This means that any codec inside a NAT would be both invisible and unreachable by another codec on the other side. Firewalls have the same effect.

This paper will be presented on Tuesday, April 15, 2008 starting at 2:00 p.m. in room S228 of the Las Vegas Convention Center. It will also be included in its entirety in the *2008 NAB BEC Proceedings*, on sale at the 2008 NAB Show. For additional conference information visit the NAB Show Web page at www.nabshow.com.

2008 NAB Broadcast Engineering Conference Summary of Presentations

Check out the papers that will be presented at the 2008 NAB Broadcast Engineering Conference in Las Vegas, April 12 -17, 2008.

Mobile TV: Opportunity at 100 MPH!

Monday, April 14 • 7:30 a.m. - 8:30 a.m.
Las Vegas Hilton Ballroom A

The Open Mobile Video Coalition (OMVC) invites engineers from television, telcos, cable and OEMs to learn more about breakthroughs and milestones in engineering, consumer interest and testing, as well as new revenue opportunities in the fast approaching locally broadcast Mobile TV world. Join them for breakfast on Monday, April 14 in Ballroom A.