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DELIVERING PROFESSIONAL-QUALITY AUDIO SERVICES OVER IP NETWORKS

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Introduction

Today's radio broadcasters are dependent upon their contribution and distribution networks to deliver broadcasts to listeners. Contribution networks bring audio sources from multiple locations into one or more studios. Distribution networks are typically point-to-multipoint, and carry program audio to remote transmitters and translators. These networks handle the signals that ultimately feed the transmitters; as such, they must reliably and seamlessly pass high-fidelity audio for transmission. They also allow for remote control, signal monitoring and data transmission.

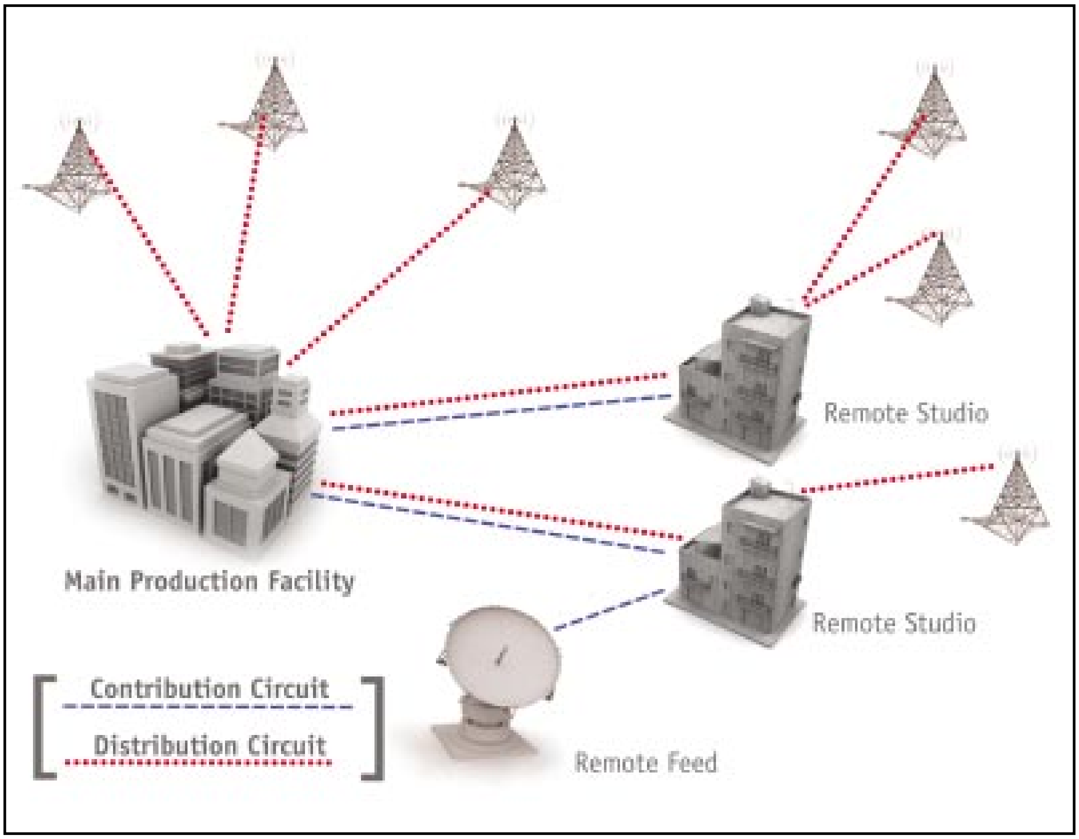

Figure 1: Simplified depiction of a radio broadcast network

To date, fixed-bandwidth digital services such as T1, E1, and X.21 have been the systems of choice, as they provide reliable communications over dedicated circuits. With the advent of packet networking, and in particular high-speed IP (Internet Protocol), the need to use expensive fixed-bandwidth services is being challenged.

IP services offer scaleable bandwidth, are widely accepted and available, and are considerably less expensive than dedicated, fixed-bandwidth lines. However, meeting the mission-critical needs of broadcasters can be a challenge. The reasons have to do with the way packet networks operate.

Fixed-Bandwidth vs. Packet Networks

In a fixed-bandwidth network such as E1 running at 2 Mbps, you get a "nailed-up" circuit dedicated to your use alone, carrying a constant stream of information. Picture a conveyor belt moving non-stop, onto which you can place your user data (audio programs or other); the belt keeps moving at a steady rate, whether you've put anything onto it or not.

The reliability on such a circuit is extremely high, but it comes at a price: you're paying for the full 2 Mbps of transport, 24 hours a day, 365 days a year, regardless of how much or how little you actually use. Also, such circuits are essentially point-to-point, which means you may need multiple E1s into a single location.

Packet networks such as IP, by contrast, are designed to be shared resources. Anyone with access to the network can put data onto it.

IP networks typically operate in one of two modes, TCP or UDP. Essentially, TCP is designed for point-to-point bi-directional communication, while UDP is more suitable for unidirectional links, either point-to-point or point-to-multipoint broadcast. Regardless of which mode, or protocol, is used, the information is broken up into small segments and placed into packets, a process knows as packetization. Along with user data, each packet contains several pieces of housekeeping data such as the destination address, a sequential packet ID, and, when implemented, a mechanism for priority tagging. For example, a packet containing bits of real-time audio could be tagged as urgent, while a packet containing bits from a data file transfer would be tagged as lower priority. Once the data is packetized, the packets are given to the network for delivery.

In the network, the packets are switched through to their destination. Each device looks at each packet, makes a decision on the best route available, and sends the packet on its way. Gateways between networks can easily become sources of short-term congestion, as they must handle all packets moving between the networks involved. It is possible that, based on short-term congestion, packets will be sent along different routes. As the routes vary, so too will arrival times. Differences in arrival time translate into packet jitter. As packet jitter increases, packets begin to arrive out of order Additionally, there is no guarantee that all the packets will arrive at the destination. Dropped packets may result in a loss of data, and increase packet jitter to maximum. If a switch along the way becomes congested, packets can be dropped as needed to reduce the congestion. When utilized, priority tagging allows the switch to differentiate between high- and low-priority packets. As congestion increases, low-priority packets are dropped first.

At the receiver, the packets are buffered and reassembled in sequential order. If TCP is in use, and a packet is missing, a replacement can be requested. Although waiting for the replacement adds delay, it allows the information to be accurately reassembled. This is not generally a problem when the packet in question contains a piece of a file transfer. However, in real-time media transport, there is no tolerance for the system to wait while requesting a packet to be resent. Therefore, real-time transport typically uses UDP. Delay is reduced, but there is no mechanism for the replacement request, and the receiver must use alternate methods to replace the missing data.

One way is through the use of Forward Error Correction (FEC). FEC rearranges the transmitted bits and adds redundant data such that a missing packet can be reconstructed based on nearby packets. The amount of FEC needed depends on the conditions in the network (especially the last mile), and an analysis of traffic over time.

The right amount of FEC can take a reasonable network and make it quite robust. On the other hand, too much FEC is a waste of bandwidth, as increased FEC levels mean increased overhead. Typically, more FEC also means larger buffers, and larger buffers mean larger delays. An ideal system allows the user to adjust the amount of FEC on each stream to best suit the actual conditions.

Finally, there is the issue of unicast vs. multicast. Basically, a unicast IP stream is equivalent to an E1 point-to-point service; packets are sent from one address to another, intended for receipt by that location only. A multicast stream, on the other hand, is made available to the network on a special IP address that allows an unlimited number of users to gain access to it. This is ideal, for example, for distribution of a program from a studio to multiple transmitter sites. It is also highly bandwidth-efficient; the multicast protocol allows the same program to be made available in multiple locations without the need to reduplicate the bandwidth more than once on any given link.

Traditional Approaches vs. the Managed Platform for IP Audio Transport

Thus far there have been two basic approaches to IP audio transport: simple stand-alone audio codecs, and "home-grown" systems solutions, often involving a combination of software and hardware from multiple vendors. Stand-alone codecs do not scale easily as needs and technologies change, while home-grown solutions grow rapidly in complexity and present significant problems in the area of management and control.

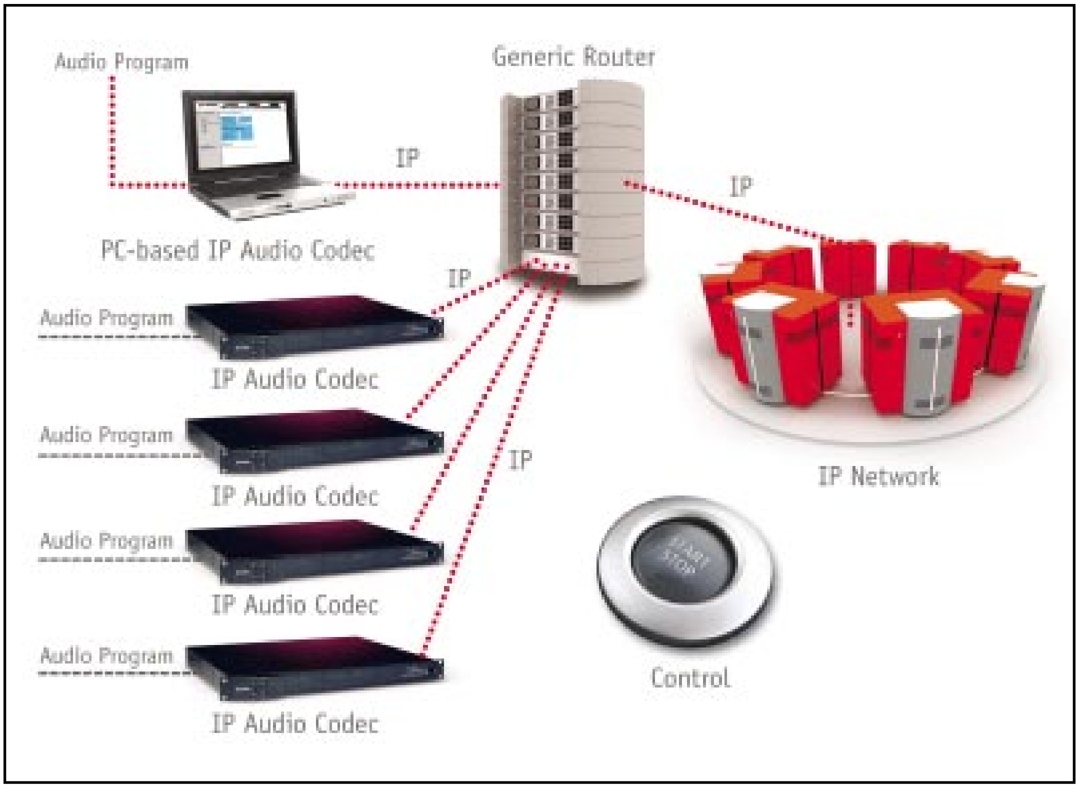

Figure 2: Traditional approach to transporting audio over IP networks

Drawing on our decades of experience in implementing complex T1/E1 networks with the Intraplex® product line, Harris has taken a new approach to IP audio transport: the Intraplex® NetXpress™ managed platform.

What, in this context, is a managed platform? It is an integrated combination of hardware and software that allows broadcasters to handle multiple audio programs, each with the appropriate type of audio compression (or none at all), integrate with existing E1 services, carry legacy data circuits and even connect to PABX telephone systems, all controlled through a single, secure, powerful management interface. It's a platform that provides resiliency and redundancy, that allows the user to set up as many as a hundred simultaneous IP streams, and to optimize each stream for its intended purpose using an extensive suite of tools.

On top of this, the same system connects to legacy T1/E1 networks, allowing program contribution and distribution over both fixed-bandwidth and IP networks simultaneously. It even provides T1/E1 circuit emulation, the ability to carry T1/E1 circuits intact over IP networks, thus allowing users to move in a gradual, controlled manner from legacy fixed-bandwidth services to the new world of IP, all on the same managed platform.

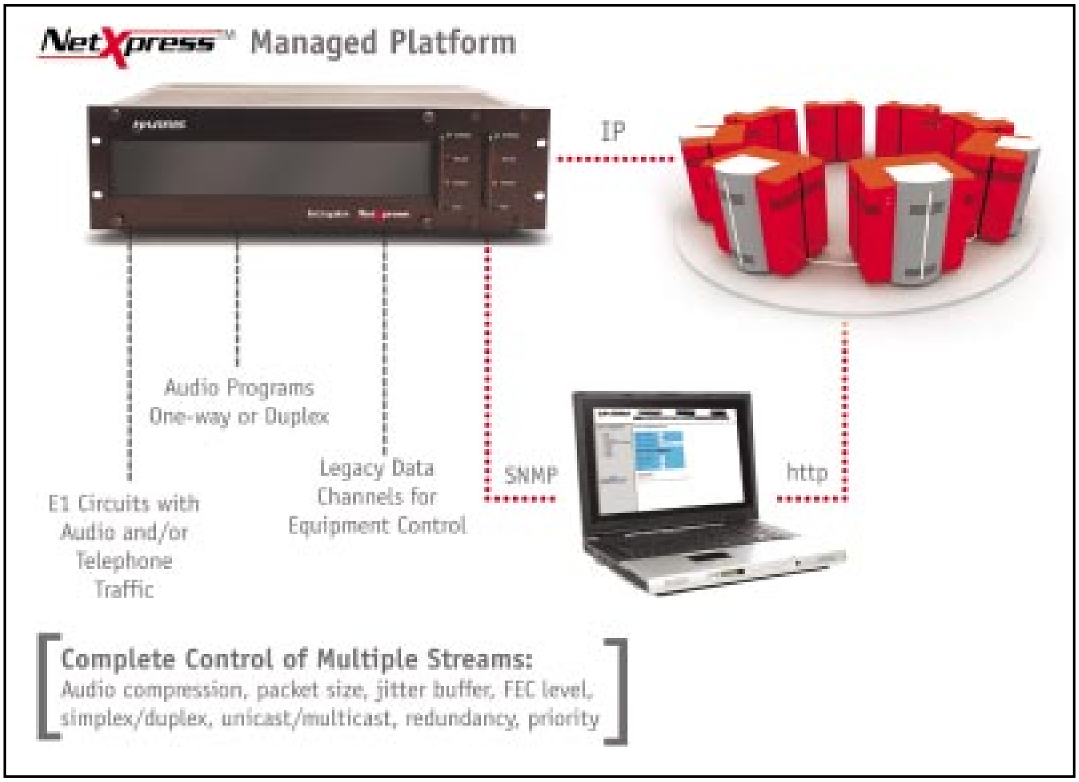

Figure 3: Managed platform approach to transporting audio over IP

Getting IP Right: the Complete Suite of Management Tools

As we discussed earlier, IP circuits present special challenges in transporting real-time program audio reliably. The right tools can help overcome these challenges. Harris provides a complete suite of tools:

- **Redundancy.** The NetXpress™ offers hot-standby power redundant power supplies and network interface modules. It also supports redundant networks, automatically switching traffic to a backup link (where provided) in case of primary network failure.
- **Resiliency.** The system recovers automatically from circuit disruptions and power failures, quickly and quietly restoring all circuits to their previous operating conditions as soon as the disruption ceases.
- **Packet size.** Control of packet size is a key tool in network management. The optimum packet size for any given stream is a tradeoff between latency (delay) and overhead. Where low delay is critical, set the packet size to a minimum, and the packetization delay can be kept below 1 ms – but this will create more overhead. Conversely, where bandwidth is tight, use larger packets. This will incur more delay, but requires less overhead.
- **Jitter buffering and resequencing.** As stated above, packets may not arrive at the receive end in the sequence they were sent. NetXpress™ provides a deep, user-adjustable jitter buffer to accommodate the real-world conditions encountered in IP networks, and has the ability to retain and restore up to 128 out-of-sequence packets
- **FEC.** Forward Error Correction allows the system to rebuild lost or dropped packets, restoring the original audio quality before playout. Again, this is a user adjustable feature – use just the amount you need for each circuit you set up.
- **Network statistics.** The system provides cumulative and current statistics on packets sent, received, lost, and delayed on every stream, allowing you to see what's really going on in your IP network and adapt to changing conditions.
- **Event logging.** Alarms and failures, on both the network and hardware side, are recorded for maintenance and review.
- **Stream type.** Every stream can be set up for one-way or duplex unicast, or for multicast, for maximum network efficiency.
- **Priority tagging.** The NetXpress™ supports multiple

levels of prioritization, ensuring that real-time audio traffic moves through the network ahead of less-critical data.

- **Input policing.** The system can provide transport of LAN traffic along with the audio channels, and can police the input to limit the amount of bandwidth available to the LAN, ensuring it can never create undue congestion levels that might interfere with the audio packets.
- **Legacy telephone support.** The NetXpress™ has the ability to carry traffic from traditional telephone systems, linking PABXs and establishing Off-Premise Extensions without moving to VoIP telephony. To this end, it incorporates built-in echo cancellation and supports traditional telephone signaling.
- **Modularity.** Everything in the system is based on plug-in modules, all of which are hot-swappable, so each location can be equipped with just the functionality required to start, and can be upgraded or expanded as desired without disturbing the existing services.

Management and Control

A managed platform can't be said to be truly managed without an effective means of monitoring and control. Accordingly, the management interface on NetXpress™ is an SNMP-based element manager with access to all aspects of the hardware and network environment. The system contains a built-in HTTP generator, which means it can be viewed and operated via a Web browser (with appropriate SSL security and password control).

SNMP also allows the system to communicate with higher-level network management software, such as Harris NetBoss or HP Openview, making it part of a complete operations management system.

Summary

IP networks offer the possibility of highly flexible, low-cost audio transport, and when properly managed can offer the degree of reliability required for use in contribution and distribution networks. But there are many challenges to be overcome in ensuring this level of reliability. Radio broadcasters armed with a suite of software and hardware tools for redundancy, error mitigation, quality of service, and network performance monitoring can more effectively tackle these challenges.

Broadcasters considering audio over IP transport can rest assured that the tools to make a successful transition exist today, and they will discover new efficiencies for their stations in the IP environment as the technology continues to mature.

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