NAB Radio TechCheck
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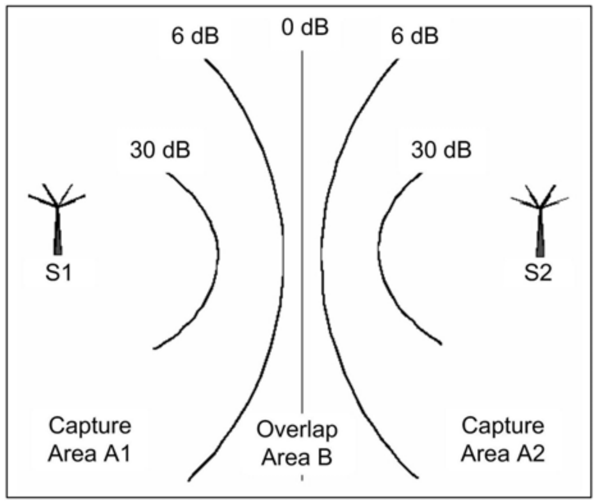
RF Simulcasting over IP Networks

Simulcasting on multiple overlapping transmitters on the same frequency can provide broadcasters with significant advantages in increased coverage and lower operating costs. A session at the upcoming NAB Broadcast Engineering Conference (BEC, April 12-17, 2008, Las Vegas, NV – see below for additional information) entitled "New Technologies for Radio Listening" includes a paper by Junius Kim of Harris Corporation describing how to utilize IP connectivity between a studio and the multiple transmission sites utilized in a simulcasting network.

INTRODUCTION – RF simulcasting uses multiple, geographically diverse RF transmitters operating on the same carrier frequency, modulating the same program material. By using multiple transmitters, geographic RF coverage area is expanded. This paper outlines a system for audio simulcasting over an IP network. The system discussed uses a precision absolute time reference provided by the Global Positioning System (GPS). Using this reference, the system can measure the STL delay between the studio and transmitter sites. The system uses this information to set a programmable digital buffer delay to reach a target delay. The buffer delay changes are smooth and hitless resulting in no noticeable disturbance of the audio program material.

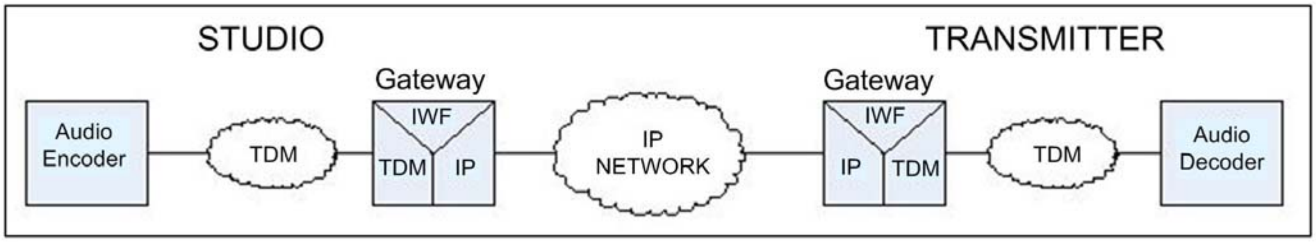


SIMULCASTING – broadcasting from two or more nearby transmitters on the same frequency can lead to reception problems in the overlap areas, i.e., the areas in which the RF signal level from multiple transmitters is similar in strength. The figure at right depicts the contours of relative signal strength from a two-site system. In the overlap area, the relative power levels differ by less than 6 dB. For simulcasting to work effectively, the broadcast signal from each transmitter must arrive at the receiver at a precisely controlled time.

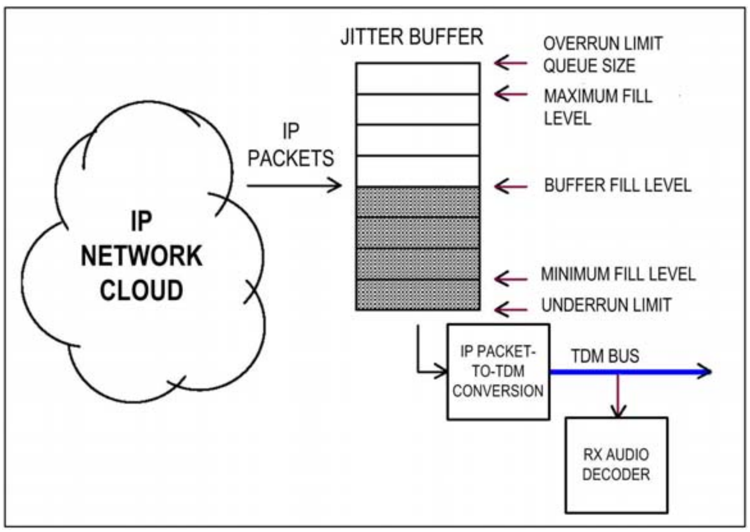
MIGRATION TO IP NETWORKS – previously, FM simulcasting systems were typically limited to usage over Plesiochronous Digital Hierarchy (PDH) network link types like T1 or E1 transmission links. Recently, this technology has been successfully adapted for simulcasting usage over packet switched networks like IP. 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 professional audio contribution and distribution networks. A simulcasting system using an IP network must perform the following functions:

- Encoding and decoding of real-time program audio;
- Transport of program audio over IP via a process of packetization;
- Sending a GPS referenced timing marker from the studio site to the transmitter site;
- Establishment and maintenance of timing across the IP network.

Circuit Emulation Service (CES) technology has emerged as a method to transport Time Division Multiplexing (TDM) trunks containing real-time applications such as audio, across IP networks. This technology is sometimes referred to as pseudo-wire, as it emulates the TDM circuit across a packet network using virtual IP tunnel or path. These emulated services can be implemented using a gateway device that provides for an inter-working function (IWF) between TDM and IP networks (see block diagram below). The primary benefit of this technology is the cost and simplicity of deployment to support all types of existing TDM applications without the need for complex protocol inter-working functions.



NETWORK JITTER – the variation in the inter-packet arrival time at the receiving gateway is caused by network jitter. The paths in an IP network are connectionless and statistically multiplexed with other sessions. The amount of network jitter depends on how the network has been engineered and how many “hops” or routers must be traversed. The figure below shows a diagram of a jitter buffer located at the transmitter site. Each transmitted IP packet in the CES “stream” will have a sequence number in its packet header. Upon reception, the sequence number in the received packet header will be examined to identify early, late, lost or out-of-order packets. If not too early or too late, the packet will be placed into the jitter buffer according to its sequence number. The packet shown at the “bottom” of the jitter buffer is “played out” to a TDM bus. It is processed by a packet to TDM conversion engine which plays out the data in real time. After playout, the packet can be discarded. Once converted to TDM data, an audio decoder processes the data for usage by an RF exciter.



DELAY RESOLUTION – the delay resolution is a function of the fastest clock provided by the GPS receiver, typically 100 ns. Delay changes and measurements can be made to within this granularity. This resolution defines the accuracy of locating the simulcasting overlap region. The geographical point in which the audio from two transmit towers are in phase will move 1 km for every 5.364 μs of delay variance, so 100 ns provides an accuracy of 18 meters.

This paper will be presented on Tuesday, April 15, 2008 starting at 9 a.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.