Audio Over IP Networks for the Radio Broadcast Studio: More than an Interface Technology, it is the Ecosystem

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Abstract

Audio over IP networks is an established technology that has overturned how modern radio broadcast studios are planned, designed, built, and maintained. But is the use of an audio over IP network merely a cost saving implementation, or is it the enabler of a fundamental shift in the capability of a broadcast radio studio? With the emergence of 'non-RF' radio programs that reach the listening public via data networks, how is the integration of audio over IP inside the studio driving the links to audio over the internet outside the studio? With many types of broadcast audio equipment, codecs, mixing boards, STL links, telephone hybrids, audio processors, intercoms, monitors, and even microphones, natively using audio over IP connectivity, the audio network and the ability to interoperate all these pieces of equipment, becomes central to the function of the modern radio studio.

A review of the history of interconnection of broadcast audio equipment follows the evolution of modern audio over IP network technology as it began as a simple interface technology but has become an enabler of the modern radio broadcast studio architecture, and delivers more system function for less cost than possible before.

The importance of interoperation of audio networking equipment is highlighted. Given competing audio network technologies without interoperation, the promise of the modern radio broadcast studio is mired in the need for additional racks of interface gear, with attendant cost and less reliability. The value of an audio over IP network is not just what the network itself does, but what it connects to. The importance of the new AES67 standard for interoperability of audio over IP networks is highlighted.

INTRODUCTION

This paper examines the impact of audio over IP network technology on the design of the radio broadcast studio. The first part is a brief review of what operations and functions are needed to get the radio broadcast studio job done. Then the second part will trace the evolution of the modern broadcast studio facility.

The aim is at the intersection of the IT network professionals who may be a little less aware of the practices in radio broadcast, and the radio broadcast professionals looking at the latest current and future direction trends.

GETTING THE JOB DONE

A brief review to define the needs of the audio technology, by way of what it takes to get the radio broadcast job done. This will set the framework to interpret the progression of facility designs. For instance, consider the contrast between a radio broadcast studio, and a recording studio.

- Radio is a live show. Must react on the spot to events inside and outside the studio.
- News happens. Prepared for the unexpected.
- Call in talk shows are a major format, dynamic, adaptive show flows.
- Patch through unplanned sources, from time to time.
- Avoiding 'dead air' is everyone's hot trigger.
- Work around equipment issues, live, under pressure.
- Many feed sources, not necessarily neatly organized.
- Multiple shows and productions simultaneously.
- Few technical staff responsible for multiple rooms/facilities/locations.
- Updates, rewiring, repurposing of rooms and facilities over time.
- Facility robustness, no central point of failure to affect all operations at once.
- Appeasing the talent
- Appeasing the boss
- Pleasing the listeners!

How IP Audio Fits In

In view of the above needs, let's look at how the features of IP network audio fit in, and fill these needs. For greater in depth detail, see reference [1].

Routing

The fundamental functional feature of IP network audio, is that it allows thousands of independent channels of audio to instantly be available everywhere in the facility. Using a single multicast network, all audio sources are available at any destination location. The IP network IGMP (Internet Group Multicast Protocol) takes care of network bandwidth usage and makes only the desired channels of audio route to each requesting endpoint. Importantly, the change of audio routing, is done locally at the listening destination, no interaction with either the source, or any of the middle connecting network switch equipment is required by the user. This simplicity of operation eliminates coordination hurdles and the opportunity for failures. This flexibility in routing extends to the setup and modification of facilities. Free form plug in, port independence on network switches, reduces central planning. Expansion into open ports is painless, quick and without risk of disturbing existing operations.

An example, the facility of Radio Free Europe in Prague, consists of over 3000 channels of stereo audio.

Control

Control of IP network audio is accomplished over the same network carrying the audio. Using Quality of Service (QOS) the audio and time synchronization packets are delivered with minimal delay caused by the other traffic sharing the network, control and monitoring. Audio routing central control, monitoring, metering and contact closure GPIO (General Purpose Input/Output), as well as HTML web page configuration, control and monitoring, and sharing of audio endpoints, coordination of arbitration and locking for devices in use, of all equipment coexist on the same network.

Being able to reach into any piece of equipment, from any location in the facility, or even from a remote location using a secure VPN (Virtual Private Network), gives the station engineer the ability to respond, diagnose, reconfigure, and resolve issues with unmatched ability. Microphone B not working in the middle of the night show? With a remote login, a look at the meters, reset the incorrectly pressed control, to wit, saves a drive into the studio.

Discovery

Discovery is the process by which you find out what channels of audio are present and available. When you have potentially thousands of channels to manage, that can dynamically change with facility reconfiguration, a live directory of channels by meaningful audio program names is essential. In an IP audio network, this information is communicated by each source from its configuration data and multicast using the same network, to allow the building up of a live distributed database of all channels that can be browsed at will. This feature is profoundly valuable for facility setup and for moment to moment reaction to unexpected situations.

In operation, setting up a new facility consists of configuring each channel of source audio. Give each piece of equipment an IP address, and each audio source channel a meaningful name and a channel number. Then the complete list of all audio channels in the facility is available throughout, and the choice of desired audio at each destination location, is easy to browse and choose. (Preconfiguration apart from a live system is also possible by directly using channel numbers.)

Profiles of setup configurations of mixing consoles, are essentially defining sets of audio channels that appear on the respective faders. Defining which audio appears on the board in what fader position for each show, is fast and easily defined through browsing the live discovery database.

Connectivity

You are what you connect to.

Beside the most immediate live microphone to console to program air chain, the origin of many of the other sources of audio in the modern broadcast studio are essentially software functions, living on computer servers. Editing, storage, playout systems, ad insert systems, all live in the ecosystem of the computer server.

A high channel count of professional quality sound cards would be needed to make all these connections in and out of these computers. Of course the native data connection of a computer is the IP network, so using network audio immediately reduces hardware cost and increases reliability. Physical connections have been found to be the most common electronics failure point, so the fewer the better. Thank Bell Labs for work in the 1970's that made the ubiquitous RJ connectors actually very reliable.

But aside from RJ connectors, network audio enables the connection of audio to something profoundly more fundamental, important, and paradigm changing… *software*. Direct connection to software is the key trend that is powering the evolution of the radio broadcast industry. I hate to say it (I'm a hardware guy myself), but no soldering irons are required.

Interoperation

Being able to connect to other vendor's equipment breaks being on an isolated island, and opens doors to the growth of possibilities in studio design. For this reason, Axia and ALC Networks GmbH partnered in early 2012 to each embrace each other's audio networking protocols, Livewire and RAVENNA. This ability to interoperate is seen as a critical necessity to support the growth of the networked broadcast studio, rather than have the industry being connected by the IP network but hampered by multiple incompatible protocols.

AES67

If some connectivity is good, more is better! This fact drove the development and recent publication of the AES67 standard [2] (formerly called AES X192) for interoperation of high quality audio over IP networks. There were over 100 task group members from the industries of networking, silicon, broadcast, commercial audio, music production, live sound, and others. End users, integrators and equipment manufacturers were represented. The manufacturers with the largest installed bases of proprietary networked audio equipment were some of the founding supporters of this new standard. There is a real commitment to embracing and using AES67. No one vendor can supply the whole broadcast audio ecosystem that we see maturing right now before our eyes. AES67 is the right vendor independent standard at the right time to enable the audio industry to connect together, and evolve explosively.

Robustness and Redundancy

Audio over IP intentionally hitches its wagon to the vast engine of the IT networking world. The IP network rides on IEEE 802.3 Ethernet, a highly developed and robust technology. The many facilities for redundancy, various media types, flexibility, and forces of a vast economy of scale, are brought to bear, to the benefit of getting the broadcast audio job done.

A typical broadcast facility design is to have a so-called 'edge' switch in each studio, to which all audio devices in that studio connect. An uplink is made from the switch in each studio to a facility core switch. This two level hierarchy of network switches provides the important feature that if any one room goes down, other rooms are not affected, and if the core happens to go down, the rooms independently continue to run (with only the loss of studio to studio connections.) The links from the studio to the core can be made redundantly, and redundant cores may be used, all using off-the-shelf enterprise class switch redundancy features.

Power Over Ethernet

One of the unsung heroes of the IP network ecosystem is Power over Ethernet (POE), IEEE 802.3af. While it may seem POE is for desk VOIP PBX phones and web security cameras, POE has some distinct advantages when used for broadcast studio gear.

What can you do with 12 watts? Quite a lot, actually. The Axia product line of 8 channel IP network audio xNodes can all be powered by standard 12 watt POE. Placing the audio I/O in the studio where you need it, with only one Cat5 tether cable simplifies by half the number of cords in the furniture, walls, desks, under the floors, by eliminating the AC power. If 12 watts is not enough, the new POE+ standard IEEE 802.3at can deliver 25 watts.

Aside from the cable convenience factor, POE can be used to achieve the somewhat elusive goal of 1 for N redundancy of internal unit power supplies. A common redundant setup is for all units to have dual power supplies. But given the probability of unit power supply failure, only one of these units is expected to fail at any given time. Because POE contains the mechanism to draw power only on demand, using POE as the backup supply will only draw power from the PSE (Power Source Equipment) when there is a failure of one of the primary power units. And thus the power capacity of the backup PSE only has to be the expected number of failed units (e.g. one or two at most), and not the total number of units. For a large facility with the number of units in the hundreds, for example, this savings can be significant.

EVOLUTION OF THE RADIO BROADCAST STUDIO

Now consider, as I outline the evolution of the modern radio broadcast studio as identified through 8 stages:

1) Classic analog studio. Point to point wiring. Patch bays for contingency routing without pulling up the floor.

2) Digital audio (AES 3). Better audio quality, but still point to point cable routing and manual patch points. Introduced the need to manage synchronization issues.

3) TDM matrix routers: Enabled by digital audio. Instant access to any channel in the router matrix, and remote command controls. This key ability justified the high expense of the matrix switch.

4) IP network audio inside the studio: Packet routing instead of TDM routing. Leveraged IT industry network switches. Enabled direct connect to computers and hard drive based playout systems. Multicast all channels available everywhere. Audio, sync and control on one network.

5) IP network audio connections used to get in and out of the studio. IP over satellite links, IP for program distribution, delivery. IP radio STL (Studio to Transmitter Links). Network connection to webcasting streaming servers.

6) Radio studios at separate geographic locations, yet interconnected as if local. Interoperating studios of large facilities between cities (i.e. XM Radio Nashville to Washington D.C.), NPR Content Depot system. SiriusXM Washington D.C. to NYC.

7) Centralized equipment server model. Only minimum necessary equipment in the physical studio, use audio over IP to geographically centralized server site. Audio, control, sync, coordination, all use common IP WAN. Central managed facility gains economy of scale. Example: ViLOR at the BBC.

8) Software systems running on generalized servers maintained by 3rd party providers (a.k.a. cloud computing). Economy of scale and centralized maintenance. Risk of air chain being in 3rd party hands, but is that completely avoidable?

I freely admit if there is healthy skepticism about stage 8, relying on the cloud for the core part of the operation of the broadcast studio. It seems to defy many of the principles of building a facility for uninterrupted operation of live on air programming. But economic considerations have a way of gaining an irresistible force behind them. And as soon as something becomes *possible,* it is not often long before someone will try to make it work in a real operation, despite any prevailing wisdom to the contrary.

At this point, I myself am very interested to know what the prevailing attitudes are regarding where the industry actually is along this 8 stage path. Progress consists of both technical advances and the *acceptance* of those advances into the common wisdom of best practice.

Looking at these stages of evolution, the first 3 steps are the historic classical non-IP audio. Stages 4, 5 and 6, the beginnings of use of IP network audio are also well established in the past decade. Stage 7 is the structure of a large initiative by a world class broadcaster, over about the last 2 years. Stage 8 is interesting, as it is essentially not so much a technical advance between 7 and 8, as it is an economic advance and shift of responsibility ownership.

If you are willing to concede that a professional organization founded on providing uninterruptable compute server platforms will become better and more effective at doing so than an organization founded on the *content* of the radio broadcast material, then the last stage 8 can be viewed as a kind of technical advance too. An advance in the care and feeding maintenance of compute server farms. Whenever I doubt the 'cloud' of cloud computing, I remind myself that I would be happy to be hitching a ride on the same server facility that banks were based on, for instance.

Fortunately, stage 7 is a kind of 'waiting room' for full blown facilities in the cloud. A broadcaster owning and operating its own geographically centralized server facility reaps a large portion of the technical and economic benefits, while still having their own staff ultimately in control.

The last step of a broadcaster owned central server to a 3rd party cloud server, if and when taken, is essentially only changing the designation of which server facility is in use. This change is facilitated by the network connection itself. In fact it could be trial tested, at a non-peak time, and a switch made back to the known good server facility on demand. This ability reduces the risks of planned server facility migration.

I mention in this context once again, the importance of the AES67 interoperability standard for networked audio. The enabling industry standard lines up the industry players to plug into each other, and lights the fuse for these advanced stages of facility evolution to go 'boom'.

Notice the shift in large scale facility structures of the last 3 stages. These change how and where the physical buildings are located, as the network collapses the limitations of physical space. This shift in the brick-and-mortar of the industry, where staffing is located, where talents and services are located, new patterns of economy of scale, putting all the physical equipment in the hands of those tending to it, is the major shift in the broadcast studio facilities of our time.

Once this physical, geographical, structural shift happens, there is no going back from audio over the IP network. The design of the radio broadcast facility is *defined* by fitting into the IP network ecosystem. Audio over IP is the enabling technology, and it is the ecosystem.

ACKNOWLEDGMENTS

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REFERENCES

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[2] Audio Engineering Society, AES67-2013 "*AES Standard for audio applications of networks – High Performance streaming audio-over-IP Interoperability"* www.aes.org/publications/standards