Migrating Audio Systems to the Network to Improve Workflow and Lower Total Operating Costs

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It's now time to rethink the traditional audio-for-television model, embrace IP, and implement the future today—a future that includes Audio over Internet Protocol (AoIP). Here's what's at stake and how leading broadcasters are making the transition.

INTRODUCTION — EVERYTHING IS DIGITAL

When personal computers first arrived in 1975, heralding the democratization of computing for the masses, few of us had any inkling of the immense scope of change they would eventually bring to the television industry.

Think back: Once we recorded audio to spools of magnetic tape on linear transport machines, edited with razor blades, and mixed through large audio consoles. Now, recordings are made to drives sitting in racks while editing and mixing are now primarily done on significantly evolved versions of those same personal computers.

My, how audio technology has changed.

It's now time to rethink the traditional audio-for-television model, embrace IP, and implement the future today—a future that includes Audio over Internet Protocol (AoIP). This white paper will focus on AoIP technologies that enable sending high-quality audio over Layer 3 packet-switched IT networks—a technology that's proven to be a game-changer for radio station engineers because it offers the ability to deploy systems quickly, upgrade or replace components easily, and integrate audio and operational control of virtually the entire radio production system.

Now it's television's turn.

– Jay Yeary, author

TRENDS IN AUDIO HISTORY

Over the last 40 years, television audio formats have made mighty transitions: from analog mono audio, to digital two-channel stereo to 5.1 surround—and soon to 7.1+4 immersive sound.

Working in digital necessitated the digitization of content libraries for storage, retrieval, and repurposing—and doing this efficiently required connecting everything to the network. In fact, networks are faster and more ubiquitous than ever, with even our telephone and communication systems part of a global digital communications network, making worldwide connections as easy to make as local ones.

With our content creation, delivery, and communication systems all digital and all networked, the time seems ripe for our disparate technologies to converge.

Unsurprisingly, given the large number of devices being delivered with Ethernet ports on them, convergence is already taking place due to readily available equipment from the IT industry. Though broadcasting is often thought of as a large industry, it is dwarfed by IT.[1] The sheer size of the IT industry turns sophisticated equipment into widely available commodities that can be purchased easily—the opposite of the very expensive, purpose-built products we're accustomed to purchasing for television projects.

In addition to widely available and simple-to-connect equipment, the IT industry gives us a robust structure to build

1 IT: *https://www.comptia.org/resources/it-industry-trends-analysis-2017*
Television Broadcast: *https://www.comptia.org/resources/it-industry-trends-analysis-2017*

systems on. Fault tolerance, Quality of Service (QoS), scalability, and security are hallmarks of well-built networks and they also happen to be the very things we look for in broadcast plant infrastructure.

Fault tolerance gives us redundancy, QoS allows us to monitor and prioritize the network for the most important content, and security is in ever-increasing demand for both content and the network. And then there's scalability. Scalability is particularly interesting in an IP-based infrastructure because it enables starting small and adding on as necessary rather than being forced into the wholesale changes necessary every time technology takes a leap. In fact, the positive drivers for IP technology are strong enough that the Audio Engineering Society, the Society of Motion Picture Television Engineers, and virtually every other industry organization are developing standards that ensure future audio and broadcast technology are driven by it.

70 percent of people surveyed use networked audio systems for nearly half of their work, compared to less than 20 percent two years earlier.

Source: 2016 Audio Networking Survey

In the Know: Protocols for AoIP

There are several protocols that are critical to insuring AoIP devices work together.

Real-time Transport Protocol (RTP) over User Datagram Protocol (UDP): All devices on the network are required to use the RTP profile, "Audio and Video Conferences with Minimal Control," with default ports 5004 for RTP and 5005 for RTCP. UDP is preferred over TCP for transport of audio because of lower overhead and minimal delay in delivery. UDP has no reassembly process so the proper MTU (1500 bytes for Ethernet) and RTP payload sizes (1440 bytes max) need to be adhered to so packet fragmentation is avoided. Packet fragmentation occurs when the data medium changes and the Protocol Data Unit (PDU) is larger than the new medium can support. Using just one medium, such as twisted pair copper cable, will help alleviate this issue.

Session Description Protocol (SDP): SDP helps with connection management and includes information on packet time, clocking, payload, and session description for all multicast and unicast sessions.

Session Initiation Protocol (SIP): Likely the most important protocol because it is specified as the method devices use to connect to each other. SIP has a host of connection features such as forwarding, parking, redirecting, and it includes diagnostic features as well. SIP-enabled devices know about each other by way of their User Resource Identifier (URI), which can be managed through a server or as point-to-point connections. Servers manage the connections of their clients but peer-to-peer connections need other assistance, either by manual entry, a discovery protocol, or by receiving the information from an upper-level protocol. AES is not the only organization acknowledging the power of SIP—the European Broadcast Union (EBU) has selected SIP in their Network/Audio Contribution Over IP (N/ACIP) standard for compressed audio IP codec interconnection as is the EBU group on Intercom Interoperability Over IP (I3P).

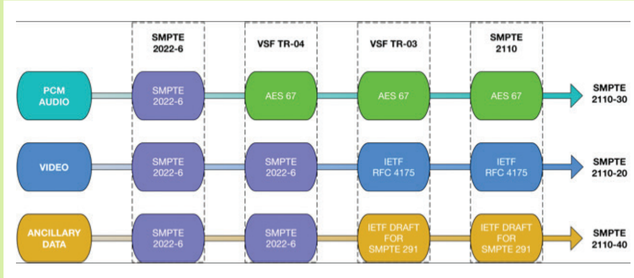
A Look at AES67

While AES67 is the professional AoIP interoperability standard, adherence is voluntary and not all AoIP solutions are 100 percent compliant. The simple answer to incompatibility would seem to be choosing equipment from one AoIP technology provider and sticking with it, but it's almost impossible for a single source to provide all the AoIP solutions needed.

AES67 was specifically created to solve the interoperability dilemma but for interoperability to work, all devices must implement the specifications set forth in the standard. AES67 equipment that is properly referenced is accurate to video frame boundaries, which makes it a natural adjunct to video sources, and is just one reason it was selected as the audio transport for the forthcoming SMPTE 2110 media-over-IP standard.

Additional components to making an AoIP system function in a television plant include the requirement that all switches and intermediary devices be managed devices so protocols can be enabled and controlled.

There is a master reference clock, akin to those in television facilities now, which uses IEEE-1588-2008 Precision Time Protocol (PTP) as the common reference clock to insure all streams remain in sync, are phase-accurate, and can be received without errors. The PTP clock is sourced from a grandmaster network clock generator that should be referenced to the facility clock. Audio devices on the network are timed to a sample accurate media clock operating at the sampling frequency of the audio, currently 48 kHz in US television facilities.



AES67 in the new Video Over IP workflow

Support for both Multicast and Unicast messaging is included in AES67, but Internet Group Management Protocol (IGMP) v2 is required if Multicast is used. Multicast sends each packet to many recipients, typically by sending data to all ports other than the sending port. IGMP makes devices report their multicast group memberships to intermediary devices on the network to avoid flooding all ports with messages. Enabling multicast messaging on a network will require the support of IT. Unicast, being a point to point protocol, sends each packet to a single recipient and is less demanding on the network as whole because messages go to one destination and don't flood the network.

If AoIP devices are being used on a business network, traffic shaping should be done, with assistance from IT, through Quality of Service (QoS) and Differentiated Services (DiffServ) to ensure media traffic is prioritized.

Although discovery is not a part of AES67, it is necessary for AoIP systems of any size, so several protocol suggestions are made in the standard, including; Bonjour, Session Announcement Protocol (SAP) version two, Axia Discovery Protocol, and Wheatstone WheatnetIP Discovery Protocol. AMWA's proposed specification, IS-04 Networked Media Open Standards (NMOS) Discovery and Registration API, is on track for use in SMPTE 2110. When selecting a discovery protocol, the best choice is one that requires static IP addresses for connections so they remain consistent.

Networked audio systems really came into existence with the introduction of CobraNet in 1996, the first audio-over-Ethernet system, and people have been moving audio over the network since. And then came the first Audio over IP solution, which came about in 2003 when Telos' Axia division introduced Livewire as method of networking radio studios together.[2] This game-changing technology gave radio station engineers the opportunity to deploy systems quickly, upgrade components, and integrate audio and operational control of nearly the entire radio production system. Other AoIP solutions have come to market since the release of the first version of Livewire, though virtually all use some sort of proprietary technology.

Realizing that incompatible competing AoIP solutions would cause frustration among users and hinder widespread adoption of the technology, in 2013 the Audio Engineering Society released AES67, their standard that details the

2: *https://www.telosalliance.com/Axia*

"Transport of High-Performance Audio Over Networks Based on the Internet Protocol" with the aim of insuring interoperability between professional AoIP devices.[3] The AES67 standard specifies interoperable mode linear PCM audio with a sampling frequency of 48 kHz , a resolution of 24 bits, and packet latency of 1 millisecond. Other sample frequencies, resolutions and latency choices are permitted. Rather than reinventing the wheel, the standards group decided early in the process to use existing and well-tested IP protocols, some in daily use in Voice over IP systems worldwide, and many in use by AoIP solutions already in the marketplace.

AES67 does not include device discovery or an over-arching control protocol, components that were intentionally left to other AES working groups to focus on the very crucial aspect of audio transport interoperability. Fortunately control and discovery solutions are available now with more on the horizon for standalone AoIP solutions as well as for those integrated into television workflows.

THE CHALLENGES FACING BROADCASTERS

AoIP technology addresses myriad issues for broadcasters, including giving engineers the opportunity to deploy systems quickly, upgrade components easily, and integrate audio and operational control of a production system.

The Telos Way

Founded in 1985 by radio engineer and inventor Steve Church, Telos Systems has a long history in the professional audio world as the provider of industry standard radio call-in solutions, telephone hybrids, and ISDN codecs. In 2003, they became audio networking pioneers with the release of Axia Livewire, the first networked Audio Over IP solution ever created.

Now, as the Telos Alliance, encompassing six companies including Axia, Linear Acoustic, Minnetonka, Omnia, Telos, and 25-Seven, their products serve an ever-expanding customer base. The recently formed Television Solutions Group (TVSG) was formed to use the expertise from each of the Telos Alliance companies to help meet the needs of television broadcasters, post production teams, and live content creators as they transition their current technology to Audio over IP.

As the creators of Livewire and as charter members of the AES67 standards working group, the company has been working with AoIP longer than anyone else in the industry. With more than 70,000 Livewire and Livewire+/AES67 devices in use worldwide, more than 100 technology partners, and many AoIP solutions (including distributed and decentralized routing, mixer hardware and software, site-to-site connectivity, call management, signal distribution, format conversion, real-time audio processing, audio measurement and monitoring, and IP intercom and communications), the TV Solutions Group is able to offer audio technology solutions for television projects of all scopes.

3: http://www.aes.org/publications/standards/search.cfm?docID=96

Typical challenges for broadcasters attempting to set up a seamless, unified audio systems in a television plant include:

• challenge of adding new analog equipment to a network
• difficulty in ensuring that all audio devices operate with similar quality and latency
• lack of control from a single interface
• frustration with a lack of interoperability
• need to avoid packet fragmentation
• need to unravel and understand various system protocols
• ability to easily manage all switches and intermediary devices so protocols can be enabled and controlled
• need to orchestrate a simplified system design

It's now time to rethink the traditional audio-for-television model, embrace IP, and implement the future today—a future that includes Audio over Internet Protocol.

When it comes to AoIP, reasons still abound for slow adoption in television, even though the protocols in use are quite mature and in daily use across the globe. Concerns include how it meshes with legacy technology and infrastructure; lack of confidence in the network and questions regarding its suitability to withstand the rigors of television production; whether network device latency will cause delays in audio feeds that would impact air or distract on-air talent; and the likelihood that it might, in some way, fail during live broadcasts, since every on-air second counts and each one lost to a technical problem equates to lost revenue.

"Despite manufacturers wanting to capitalize on this growing trend [of AES67 and AoIP], there is still a lot of mystery and misconceptions about AES67 and AoIP in both the Radio and TV industries," said Marty Sacks, vice president of sales, support and marketing for the Telos Alliance.[4]

SCOUTING FOR A SOLUTION: THE UNIQUE ADVANTAGE OF AOIP IN TELEVISION

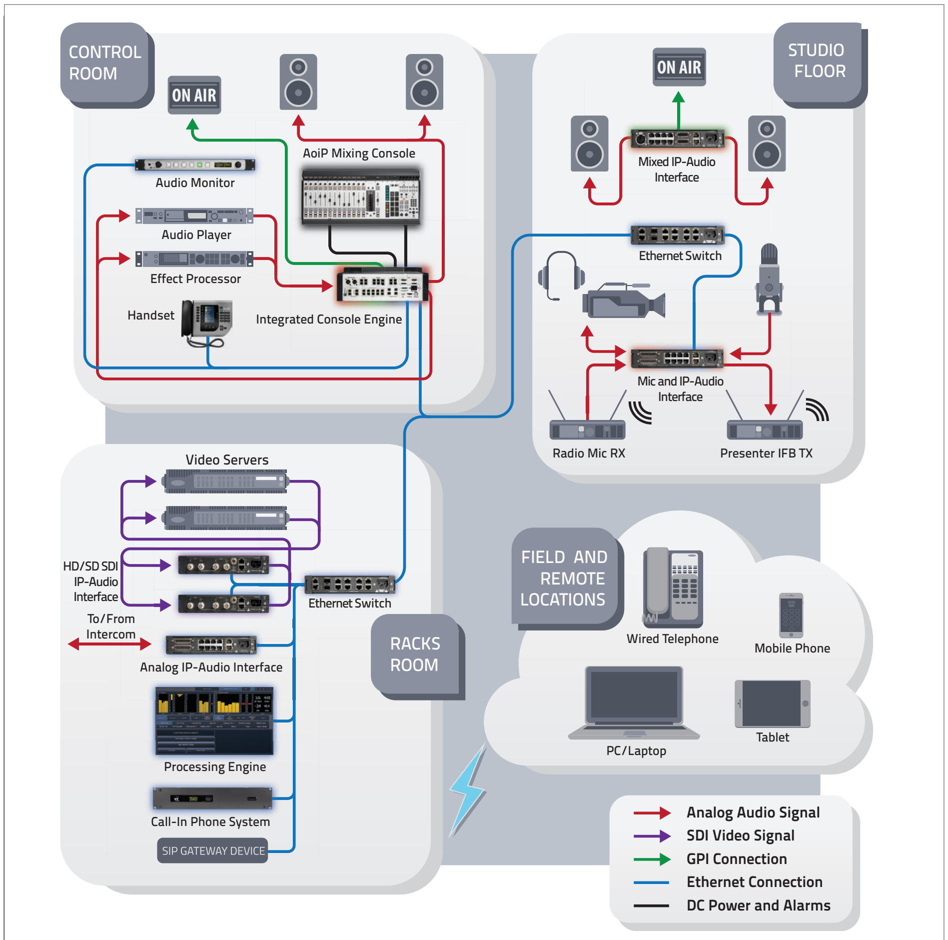
But it's clear that the technology has its advantages.

The obvious advantage of AoIP technology is that it simplifies system design and connections since most are made with Ethernet cables to end and intermediary devices. Those connections don't just carry audio however, they also carry information about the data, including connection management data and control and discovery protocols.

While AoIP networks carry digital audio, adding analog equipment to the network is easy by using conversion devices that also add control and routing flexibility of those analog signals in the process.

Converting analog signals to digital, moving all communications and audio IO to the same system, transitioning from plain old telephone (POTS) lines to Voice Over IP (VoIP), and to IP codecs in place of ISDN codecs means that all audio devices, remote and local, now act as a single system with essentially the same audio quality and the same latency. Legacy intercom systems become part of the network of routable audio signals instead of an entirely different system. IFB, audio for studio productions, remote locations audio: all are simply more audio sources on the network.

4: *https://www.telosalliance.com/News/The-Telos-Alliance-Leads-the-Way-in-AES67-AoIP-NAB*



An example of an AoIP connected broadcast television system

With the proper control software, all these systems and their audio are controllable from a single interface with the ability to save and restore profiles, while displacing other matrixes and control systems.

When considering the technology for television, the integration of AoIP with existing equipment is relatively easily through conversion devices and can be done with minimal impact on day to day operations since the technology is scalable; thus, buildout and conversions can be done only as necessary. Network speeds continue to improve and with media traffic prioritized on the network, reliability improves.

"Once you accept the idea that the entire audio infrastructure that you've come to know can be replaced by a network that more closely resembles an office IT system, you will be able to understand how the transportation of audio as data represents an evolution with such profound benefits that it can no longer be ignored," said Martin Dyster, vice president of business development, Telos Alliance TV Solutions Group.[5]

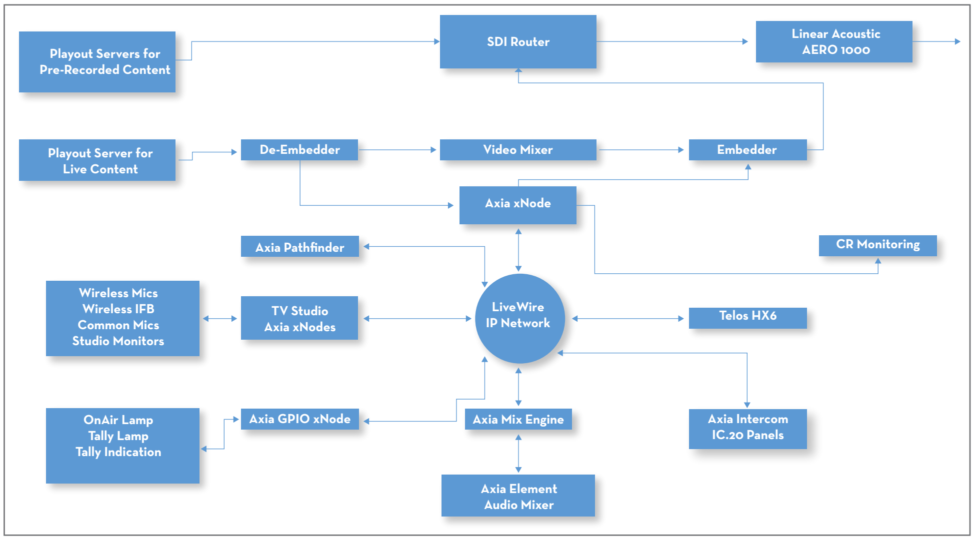
Full network redundancy can be built to support the media network in the event of an equipment failure, from dual power supply devices to redundant switches, and including automatic failover. Should problems arise, AoIP networks can use the same diagnostic and management tools developed for networks and telecom systems. Replacing a failed piece of IT equipment, once it is in hand, is often as simple as bolting it in the rack, powering it up, loading a configuration file, and connecting it to the network. As for network latency, AES67 was designed specifically for live audio, with a default packet latency of 1 millisecond, low enough that talent should never hear delay in their IFB.

According to the 2016 Audio Networking Survey conducted by RH Consulting, 70 percent of people surveyed use networked audio systems for nearly half of their work, compared to less than 20 percent two years earlier.[6]

The same survey listed interoperability as one of the top three reasons for choosing an AoIP technology, while most of those polled prefer to run AoIP devices on AoIP-only networks.

One facility adopted AoIP for television after seeing how well its radio studios successfully and whole-heartedly embraced the technology. The Russian Media Group had all their audio devices, analog and digital,

5: *http://blogs.telosalliance.com/choosing-an-aoip-format-for-a-broadcast-production-facility-with-aes67*
6: *"Audio Networking Survey, Summer 2016", RH Consulting*



Studio microphones, IFB, Studio Talkback and Cues can all be handled on the single network cable after RUTV integrated Livewire AoIP infrastructure into its TV facility.

connected and even controlling their devices over GPI, through one networked system. Once the television team saw the system in action, they were ready to implement AoIP for television. Running just one Cat-5 cable and installing an Axia xNode gave RUTV the ability to bring all their Radio and TV sources together into their Master Control mixer. All studio microphones, IFB, talkback, outside feeds, playback devices, and six-

The 2016 Audio Networking Survey listed interoperability as one of the top three reasons for choosing an AoIP technology.

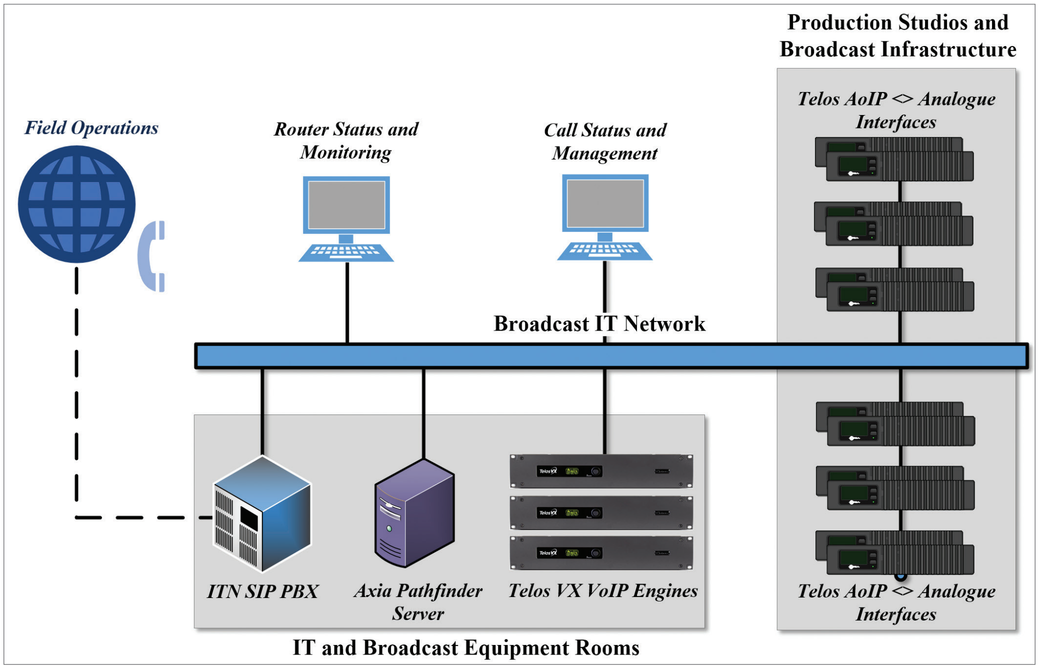
line telephone call-in system are now all on the AoIP system, though control room monitoring, cues, IFBs, and mix minuses require a few additional cables.
"Usage of Livewire offers great flexibility in management of audio signal delivery for any program," said Russian Media Group CTO Andrey Mamontov. "In addition, it greatly speeds up audio workflow for the creative team."

For the London-based news and multimedia company ITN, the credit for moving to AoIP might be laid squarely at the feet of Brexit. ITN needed to upgrade their 25-year-old analog router and associated infrastructure in time to cover the British referendum that eventually led to Brexit. The company needed to replace an aging analog routing system with one what would allow ITN to move its systems into an IP-centric world without loss of service or functionality.

Along with the router replacement, ITN was looking to move to a distributed architecture, gain tighter integration with their internal PBX phone system, and implement VoIP capabilities—while simultaneously integrating everything with their existing routing system. And all of this it without impacting day-to-day operations.

The company selected an AoIP routing system built using commodity IT switches, 30 Axia xNode IP audio interfaces, three Telos VX VoIP interfaces, and Axia Pathfinder servers, all of which are controlled by a custom middleware solution provided by their UK partner Broadcast Bionics.

"Pathfinder discovers, creates and manages a virtual router out of all the discreet AoIP endpoints," said Sim Johnson, product manager for Broadcast Bionics. "In an IP



ITN was looking to upgrade their 25-year-old analog router and associated infrastructure with a solution that would allow it to move its systems into an IP-centric world without any loss of service or functionality.

audio system, there is no central 'brain,' other than the network, so you need a piece of software to tie all the endpoints together and take that distributed system and make it into a router. We had to do some bespoke development to produce customized screens to fit the customers' requirements, but when we went to install the IP audio system, it all was incredibly straightforward."

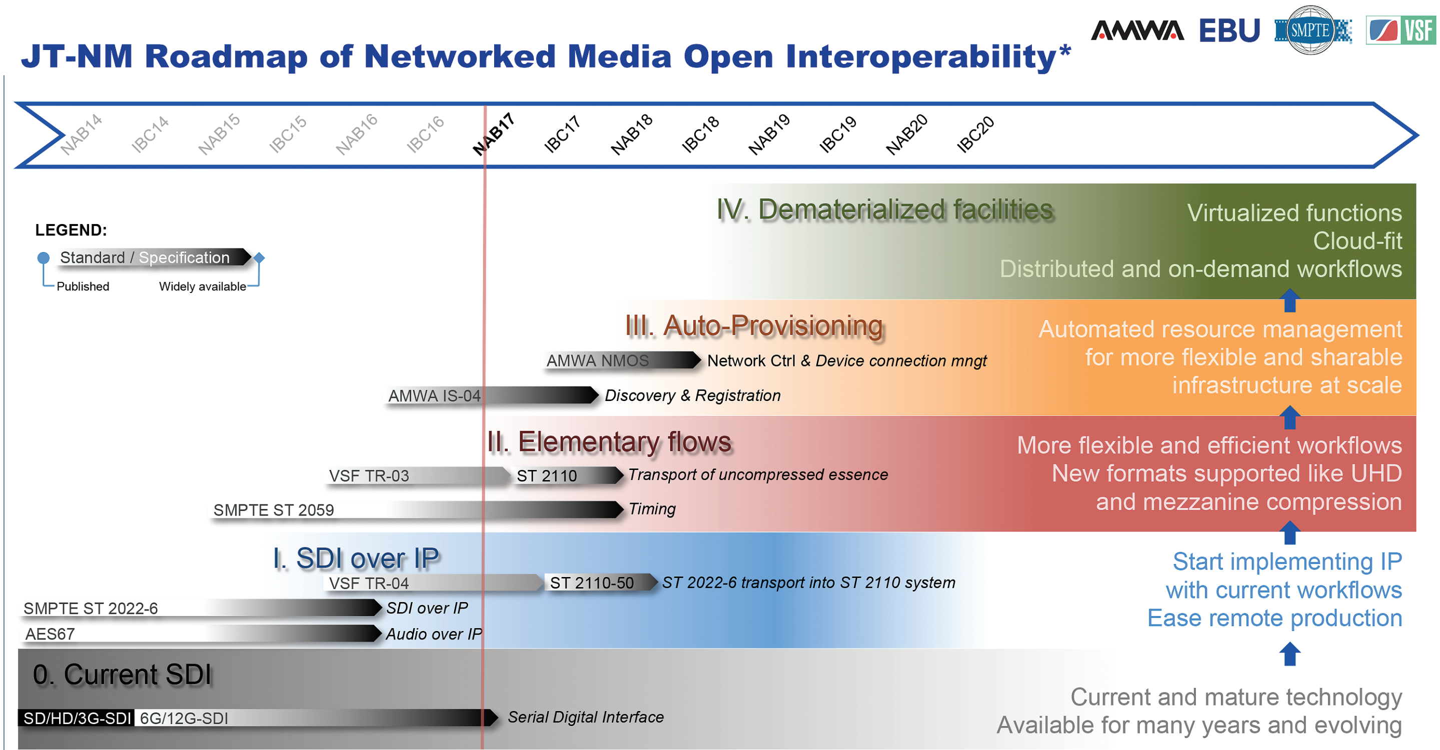
The system has worked so well that ITN has expanded the AoIP system to support distributed monitoring and offline recording of calls.

Though the technology is solid and the convenience of connecting devices with Ethernet cables is obvious, the often-overlooked element of AoIP is that it is actually more cost-efficient to implement, manage, and maintain than traditional television equipment. The ability to merge all audio communication systems gets expensive hardware out of the rack, minimizes software control systems, and eliminates the cost of the communication lines for each device (a single ISDN line averages $100 each per month). "SIP/VoIP has been found to be 50-90 percent less expensive than traditional telephone services," said Telos Alliance CTO Greg Shay.

AoIP also replaces purpose-built expensive equipment with COTS equipment. In fact, as a plant moves more toward IP devices, almost everything becomes commodity equipment. Audio crosspoint routers can be replaced by Ethernet switches and distribution equipment is no longer needed as soon as everything is addressable in the AoIP system.

"Compared to a traditional TDM router backend solution, AoIP is approximately half the cost per channel," Telos' Shay said.

AoIP also reduces power, cooling, and rack space requirements in terminal and operational rooms.
Scalability means no wholesale replacement of technology is needed up front and



The merging of IP technologies across broadcast. [7]

growing the system is easy. Migration to AoIP is easily done in stages due to the scalability of the design, which minimizes downtime and maximizes uptime. Since the network and routing infrastructure are limited by bandwidth and not physical port numbers, adding capacity only requires adding more switches and managing network changes.

Content can be secured, too. Security is an increasing concern on all networked systems and by using best practices from the IT world, users can secure content, not just from unauthorized access by outside users, but also from in-house users to protect from accidental routes and changes in the system.

SUMMARY

With AES67 part of the SMPTE 2110 television standard—and with almost every industry organization working on some aspect of IP implementation—the question of 'when' AoIP will be implemented for television is no longer a question.

While video standards and equipment are still in development, robust AoIP solutions are ready to go now. The next-generation, IP-based broadcast system, ATSC 3.0, is on the horizon, bringing with it immersive and object-based audio, and AoIP makes adding the infrastructure to support additional elementary streams and audio channels manageable without wholesale technology replacement.

Since the new video standards call for audio to remain separate throughout the creation and production process, there are no longer any technical reasons to postpone the planning and implementation of AoIP—it just comes down to deciding when to make the move and choosing the correct AES67-compliant solutions. It's time to rethink the traditional audio-for-television model, embrace IP, and implement the future now.

7: *Joint Taskforce for New Media, April 2017 JT-NM Roadmap of Open Interoperability (April 2017)*

ABOUT THE AUTHOR

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