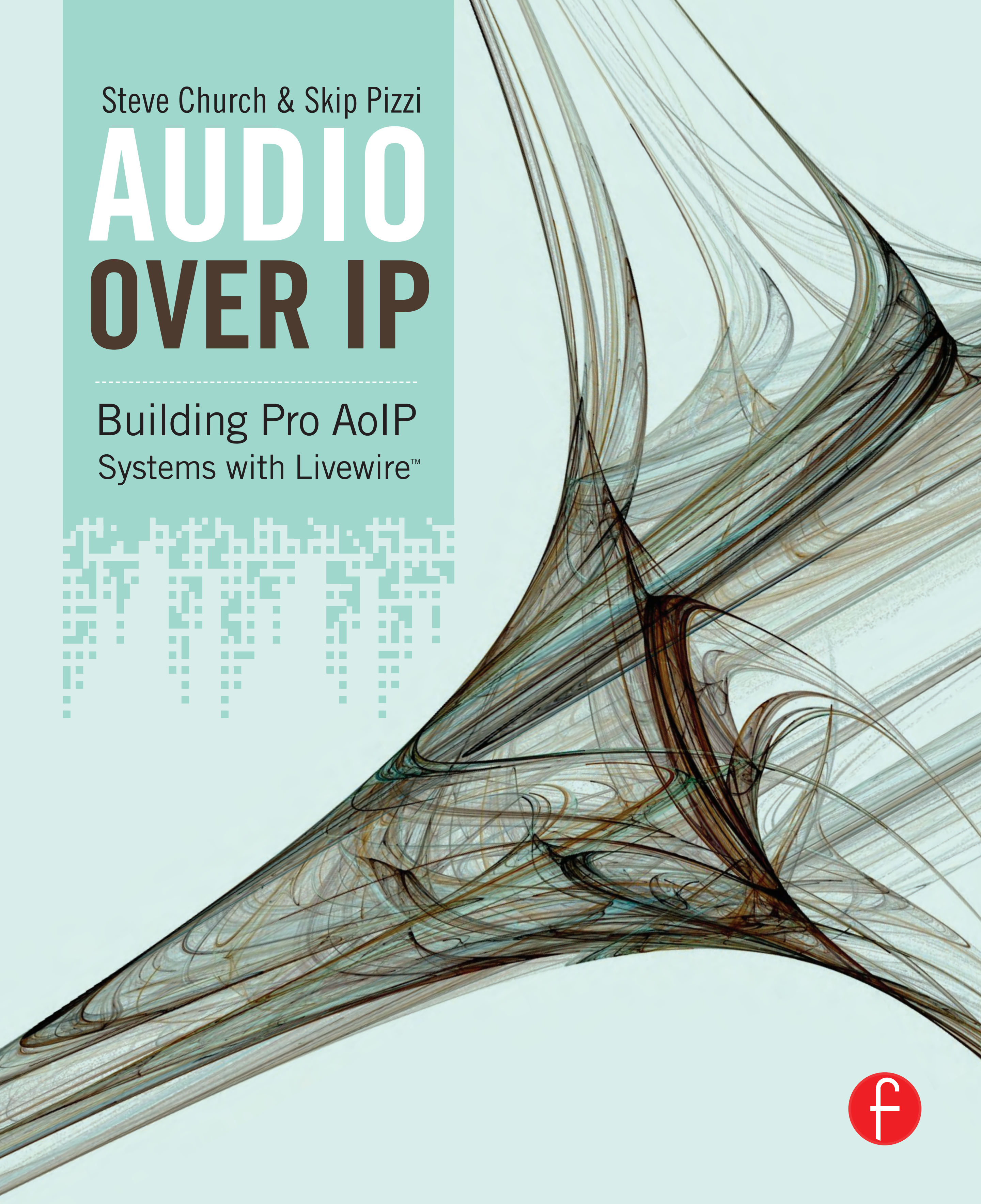
Steve Church & Skip Pizzi

# AUDIO OVER IP

## Building Pro AoIP Systems with Livewire™



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**Steve Church**

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Focal Press
Taylor & Francis Group
NEW YORK AND LONDON

First published 2009 by Focal Press
70 Blanchard Road, Suite 402, Burlington, MA 01803

Simultaneously published in the UK by Focal Press
2 Park Square, Milton Park, Abingdon, Oxon OX14 4RN

*Focal Press is an imprint of the Taylor & Francis Group, an informa business*



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**Library of Congress Cataloging-in-Publication Data**
Pizzi, Skip.
Audio over IP : building pro AoIP systems with Livewire / Skip Pizzi, Steve Church.
p. cm.
ISBN 978-0-240-81244-1
1. Digital audio broadcasting. 2. Netscape Livewire (Computer file) I. Church, Steve. II. Title.
TK6562.D54P59 2010
006.5--dc22 2009029537

ISBN 13: 978-0-240-81244-1 (pbk)

# Dedications

*To my wife and muse, Lana, who, happily, says she missed me while I was writing.*
*–Steve Church*

*To my family, who make all of life's lessons worth learning.*
*–Skip Pizzi*

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# Acknowledgements

Our thanks go to the many amazing people who have contributed to the development of Livewire AoIP technology and its realization, most notably Greg Shay, Michael Dosch, and Maciej Slapka in Cleveland, and Maris Sprancis, Oleg Krylov, Gints Linis, Normunds, Artis, and the rest of the LUMII team in Riga, Latvia. We also are indebted to our editors at Focal Press, Melinda Rankin, Carlin Reagan, Paul Temme, and Angelina Ward, and our colleagues at *Radio World*, Paul McLane and John Casey. To all, we are grateful.

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# Preface

In 1984, the writer Italo Calvino began composing a series of lectures he never delivered. They were entitled "Six Memos for the Next Millennium," although he only completed five of them before his sudden death in 1985. The lectures were later published in a book of the same name.[1]

His lectures—or "memos," as he preferred—were critiques of literature, considering a myriad of works ranging from Lucretius and Ovid to Joyce and Dostoevsky. Yet a quick look at the lectures' titles shows how they serve as apt metaphors to technology, as well. Their subjects are paragons we associate closely with the digital age that now flourishes in the new era that Calvino addressed from some temporal distance. He named his memos simply:

1. Lightness
2. Quickness
3. Exactitude
4. Visibility
5. Multiplicity
6. Consistency

Engineers will observe how these could easily be taken as high-level design requirements for any proper technology. And in fact, as Calvino wrote these lectures,[2] the Internet as we know it today was being born.[3]

From our contemporary perspective, these concepts still apply well, and also fit nicely into audio engineers' narrower worldview of an idealized digital environment. So they particularly pertain to the subject at hand—audio over Internet Protocol (AoIP) for professional applications.

The agility, speed, accuracy, clarity in design, scalability, and reliability that AoIP systems possess closely mirror the six virtues that Calvino set out. In fact, we could dare to add a seventh, "Efficiency," to complete the set of qualities we ascribe to today's AoIP technologies. Of course, returning to the mundane as we ultimately must, this last attribute translates to cost effectiveness, which is likely the most appealing of all to today's implementers. But it is the other, more fundamental characteristics that combine to enable this more pecuniary advantage.

---

[1]Italo Calvino, *Six Memos for the Next Millennium* (Cambridge, MA: Harvard University Press, 1988).
[2]ARPAnet had just fully converted to TCP/IP in 1983, and the term *Internet* was recognized as the network's official new name at that time.
[3]Christos J. P. Moschovitis, ed., *History of the Internet: A Chronology, 1843 to Present* (Santa Barbara, CA: ABC-CLIO, 1999).

Don't be alarmed—we'll not dwell long on Calvino's literate musings. Rooted as we are in the no-nonsense environs of radio studios and audio production facilities, we'll move quickly to our main goal: to illuminate the practical workings of AoIP. To round your learning, we will provide both the necessary theoretical concepts and hands-on examples of AoIP systems for professional audio and broadcast use.

We begin with a treatment of general AoIP principles, then proceed to how these are realized in one particular family of products—the *Livewire*[4] system. The use of real-world reference points are valuable to understanding, aiding in the transference of purely conceptual information to knowledge that can be acted upon.

The motivation for our choice of Livewire for specific description and concrete examples is twofold: First, it is a standards-based system, making it well suited to the task of illustrating the value of the standardized networking approach. Second, the Livewire system is in wide use around the world. (And, third, it doesn't hurt that we're pretty familiar with it.)

We further believe that our coverage of Livewire as a specific instance of AoIP does not reduce the utility of this book for users or potential users of other AoIP systems. On the contrary, having real examples is vastly preferable to sticking purely to theory. We trust that many of the elements of Livewire we discuss will be easily recognized and made applicable to other systems.

Compare this to a book on web design. If the presentation considered only generic source code and did not describe the actual effects on a particular browser, its usefulness would be greatly reduced.

Thus, the chapters at the beginning and end of this book consider generic AoIP technology, while the ones in the middle focus on Livewire's specific implementation of it. Among these central parts, Chapters 6 and 7 play around the boundaries of AoIP, covering Voice over IP (VoIP) telephony, and audio codecs optimized for the IP environment, respectively. These chapters treat their subjects in a largely non-implementation-specific manner, as well, but one in which the professional audio and the broadcast facility are primarily considered.

Finally, this book is for two groups: those who already have installed AoIP systems and those who are considering it. For the first, this book can serve as a manual with a wealth of information on cabling techniques, equipment maintenance, and other real-world topics. For the second, we trust that upon reading this book, you will understand AoIP at a sufficient level to evaluate if it's right for your facility. In either case, we hope you will find this book a helpful guide along professional audio's new frontier.

[4]The *Livewire* format is a standards-based AoIP system developed by the Cleveland, Ohio-based manufacturer Axia Audio, and supported by a growing number of other audio and broadcast equipment manufacturers. (See the References and Resources section for further information on Livewire products and partners.)

CHAPTER 1

# Introduction to AoIP

*"IP is like Pac-Man. Eventually, it will eat everything in its way."*

**—Hossein Eslambolchi, President, AT&T Labs**

*"Rock and roll is the hamburger that ate the world."*

**—Peter York**[1]

*"AoIP eats old-school studio audio technologies for lunch."*

**—Steve and Skip**

The *Internet Protocol*, usually simply called IP, is at the heart of the Internet. IP is the common format used for any kind of data that flows on the Internet and on private extensions of the Internet, such as the local area networks (LANs) employed in enterprise networks and small office/home office (SOHO) networks. Together with Ethernet for transport (cabled or wireless), the rules are set for the entire data networking infrastructure, both hardware and software, which has emerged from a rabble of competitors and has been so broadly embraced over the last quarter century.

IP is now driving a revolution in the field of audio studio design. It promotes a fundamental rethinking of the way signals are distributed and managed throughout the broadcast facility. Since most audio facilities have already been converted to digital, it makes sense to move on to explore the next step in the progression—transitioning to IP—as well.

Given that IP is the *lingua franca* of contemporary data networking, it can provide significant economies of scale for specialized applications such as professional digital audio distribution. This exploits the same process that has made the general-purpose desktop computer an efficient and cost-effective platform for the creation and storage of professional audio content. Audio-over-IP (AoIP) distribution is simply an extension of that thinking and technology, replacing the purpose-built (and relatively expensive) mixers, routers, and switchers that have traditionally been used by

[1]Peter York is a British author, columnist, and broadcaster. It's not clear to us if he was being kind to rock and roll, or hamburgers. It's also not clear if his comment is relevant. But, we are sure that we like it, and that it fits the "eating everything in its path" theme.

audio studios for managing multiple audio signals as they pass through a production or broadcast facility. IP also allows the full and continuing force of Moore's Law (which states that capacity doubles every two years) to be applied to audio distribution, just as the PC has done for recording and editing. (Anyone remember New England Digital's Synclavier? Popular in the 1980s, this audio recorder/editor/synthesizer was an impressive machine that cost its well-heeled owners over a half-million bucks. Today, a $400 PC offers teenagers much more audio production power.)

Beyond cost effectiveness, however, AoIP offers other important benefits, including:

- Scalability (i.e., the ability to easily accommodate growth and other configuration changes).
- Convenience (i.e., easy and fast installation).
- Tight integration with Voice over IP (VoIP) phone systems, IP codecs, and PC-based applications.
- Smooth incorporation of other services such as associated text and visual content.
- "Future-proofing" (i.e., high likelihood of fitting well into any scenario for future facility requirements).

Putting all these elements together creates a value proposition that is hard to ignore when you are considering options for new facility designs or existing studio upgrades.

Studio audio systems using IP-based technology are now sufficiently mature to allow audio producers and broadcasters to confidently make the transition, providing them with substantial savings while simultaneously positioning them well to accommodate future needs.

## 1.1 TWO TO TANGO

The broadcast audio studio has a long legacy relationship with the telecommunications world. The earliest audio facilities and standard practices were developed by Bell System and Western Electric engineers in the early 20th century, and the two worlds have never strayed far from each other since.

In particular, broadcast audio has retained a close connection to the telecom environment, since so much of broadcasting's content comes and/or goes from the studio via telco-provided paths. Broadcast equipment designers also have leveraged (and continue to) the massive research and development (R&D) investment made in telecom/datacom technologies.

AT&T's U-Verse service is instructive. It is a consumer telecommunications offering that bundles TV, voice, and Internet, all of which are IP-based. Meanwhile, Alcatel/Lucent, which now owns AT&T's central office equipment business, shows no circuit-switched products on their web site, instead focusing on IP-based central office solutions. AT&T was, of course, the company that invented the circuit-switched paradigm that powered telephony since the 1970s, and served as the inspiration for traditional broadcast routing gear.

U-Verse is an example of an "IP but not Internet" application. The TV and voice services don't need to use IP, but AT&T has decided to consolidate all the services on a common infrastructure, presumably to both save money by leveraging high-volume hardware and to have maximum flexibility via IP's do-anything capability to adapt to whatever the future might bring.

It is not surprising that the next generation of studio audio technology should once again follow a path blazed by telecommunications technologies. AoIP is also "IP but not Internet," leveraging high-volume standard hardware and offering future-proof flexibility.

## 1.2 ARGUMENTS FOR AoIP

What makes IP so compelling? It's "just a protocol," right? Yes. But a protocol in the data networking context can provide tremendous value to users. At the technology level, it's simply a set of rules: the way data is assembled into packets, how confirmation of reception is communicated, etc. But to users, it means that any conforming equipment is interoperable. And because the IP protocol was designed with generality and extensibility in mind, it enables designers to create novel applications.

Although originally developed for email and file transfers, as the speed of the Internet increased, IP came to be used for media transmission as well, which is now well known as *streaming media*. This development has fundamentally altered the nature of how people use the Internet, and has subsequently had significant impact on all aspects of the media industry as it struggles to cope with the changes it brings and to take advantage of the new opportunities it engenders. Though the Internet's inventors were probably not thinking of streaming when they designed IP, they *were* thinking that keeping the core open and layered would unlock the door to a variety of applications that future creative types might dream up.

Which brings us to AoIP. While they are related, AoIP is not streaming media. Streaming is exemplified by public Internet applications such as YouTube and Pandora. There are no delivery guarantees for these services, and delay can range into tens of seconds.

On the other hand, AoIP is intended to be run exclusively on a controlled local network infrastructure. In some cases, this is just an Ethernet switch. In others, it's a sophisticated system comprised of multiple IP routers and/or Ethernet switches. In all cases, an AoIP system is designed to ensure reliable, low-delay delivery of audio streams suitable for professional applications.

### 1.2.1 Scalability

Perhaps the most fundamental advantage of AoIP systems over other audio technologies—analog or digital—is the ability of its underlying IP and Ethernet architectures to adapt to change and growth.

For example, a traditional audio environment must have its spatial or imaging format (e.g., mono, stereo, or surround) predetermined, along with the number of simultaneous audio channels it requires (e.g., one, two, or more). An AoIP environment has no such requirement, and can easily adapt to any audio channelization format. This applies to accommodation of any other "layers" in the system as well, such as control-data channels. In traditional architectures, a dedicated path had to be specified for these extra channels (such as RS-422 control data). AoIP systems allow such auxiliary components to be easily and flexibly carried alongside the audio payload.

Similarly, a traditional "crosspoint" audio routing switcher must have its input and output (I/O) configuration fixed in its hardware design. In this way, such a device reflects *circuit switching* and parallel design, whereas AoIP systems implement *packet switching* and serial design. The packetized, serial approach allows great flexibility and responsiveness in accommodating changes in I/O configuration.

Just as telcos have moved away from the circuit-switched paths of their earlier years for similar reasons, studio audio systems can now enjoy the same advantages of scalability and flexibility to implement expansion in any dimension. This comes not a moment too soon, given the competitive pressures coming to bear on broadcasters to accommodate increased content production and expanded audience choice.

### 1.2.2 Cost Effectiveness

At almost any reasonable size, an IP-based audio system will compare favorably with the cost of a traditional system—both in terms of its hardware and materials pricing, and its installation costs. The reduction in wire alone provides substantial economy.[2] Maintenance expenses for AoIP systems are generally also lower.

These cost differentials increase with the size of the facility, which is why so many larger installations have already moved to IP-based solutions as their needs have called for new technical plants.

### 1.2.3 Convenience

The small physical footprint, low operating cost, ease of reconfiguration or upgrade, and fast installation of AoIP systems make them extremely convenient for engineering and operations alike at the audio studio facility.

From initial design to implementation to daily operation, IP-based systems make life easier.

[2]Remember that a packet-switched system like AoIP does not require individual wiring paths to each I/O of every device. For example, an audio mixing console or multitrack recording device can have all of its inputs and outputs interfaced to the rest of the facility via a single cable in an AoIP environment.

### 1.2.4 Smooth Integration with Other IP-Based Systems

VoIP phone systems and IP codecs can be tightly interconnected, creating numerous benefits with regard to both ease of installation and feature enhancement.

### 1.2.5 Talking the PC's Native Language

A lot of studio audio these days is either being sourced from a PC or being sent to one. IP/Ethernet is the PC's native language, allowing a powerful low-cost interface. Via a single RJ-45 connector, many channels of bit-accurate, high-resolution, bidirectional audio can be connected. Control comes along for the ride.

### 1.2.6 In the Tech Mainstream

Being in the tech mainstream means that there are a wide variety of learning resources. Books, web sites, and college courses that cover IP and network engineering abound.

Category (Cat) cables, assembly tools, RJ patch cords, jacks, testers, etc. are widely and locally available. Even some Ethernet switches and IP routers are often stocked locally.

### 1.2.7 Future-proofing

Nothing strikes fear in the heart of the engineer or manager more than making a bad decision on a big-ticket purchase. Moving to an IP-based audio architecture takes a lot of the pressure off, since it offers such flexibility and allows broad ability for reconfiguration down the road. Provisioning for unforeseen changes is much less problematic and cheaper with AoIP than with any predecessor architectures.

Note that the above advantages only fully apply to systems that use *standard* IP in their design. Not all audio systems that use computer networking (over Ethernet and/or on RJ-45 connectors) for interconnection are necessarily "true" AoIP systems. Some systems simply use Ethernet as a physical layer with a proprietary data format above it (e.g., Cobranet), while others may use more IP-like formats but with non-standard protocol variations.

Some of these nonstandard approaches may have offered some value in the past (such as reduced overhead and latency over standard IP networking), but given the capacity, speed, and performance of a properly configured, standard IP system today, the penalties paid by working in a nonstandard environment generally far outweigh any advantages that such variations might provide, particularly when considered over the long term.

Therefore, this book confines itself to the consideration of fully standardized IP-based systems only, both in its generic AoIP discussions and its specific references to the *Livewire* system (which is an example of such a standards-based AoIP approach).

**THE GRAYING OF AES3** For digital audio transport today, AES3 is the main alternative to an Ethernet-based system. Invented in the days of 300-baud modems, AES3 was the first practical answer to connecting digital audio signals. But it's now over 20 years old and is showing its age. Compared to AoIP's computer-friendly, two-way, multichannel-plus-high-speed-data capabilities, AES3 looks pretty feeble with its two-channel and unidirectional constraints.

Then there's the 50-year-old soldered XLR connectors and lack of significant data capacity. AES3 is a low-volume backwater, with no computer or telephone industry R&D driving costs down and technology forward. Your 300-baud modem has been long retired; it's time to progress to the modern world for studio audio connections, too.

## 1.3 IP-ANYTHING

As the world transitions almost everything to IP, we will likely discover even greater synergies as time goes on. The leveraging of IP as a mechanism to use *generalized* systems and transport paths for various *specific* tasks has undeniable appeal. We've seen U-Verse as a prime example, but this argument is also finding favor in a wide range of other industries, from hotel TV systems to health care. Emerging digital TV transmission systems including the new mobile variants are also favoring an IP distribution model.

For broadcast-industry engineers, familiarity with digital networking technologies, including IP, has become a near-requirement of the job anyway (e.g., it's needed in implementing the online services of a radio station), so why not apply this knowledge to studio audio, too?

It's becoming clear that IP is truly the way of the digital media world, particularly for any industry that values connectedness, agility, and cost effectiveness. In the radio environment, it's not an overstatement to say that AoIP is the future of studio audio signal flow. Arguing otherwise is difficult: There is and will continue to be so much development within the IP environment that it only makes sense to harness the power of that effort, while also allowing Moore's Law to have its ongoing effect on hardware cost reductions. The effects of these very forces are being enjoyed by so many other industries today; why not in professional audio as well?

## 1.4 WHAT'S THE CATCH?

This is not to say that there aren't some challenges. Primary among these is the latency that the encapsulation process of audio data into IP packets can cause. As you will see in the chapters ahead, on a controlled local network, this can be made sufficiently small to satisfy pro-audio requirements.

Another issue is a simple one of connector standards. Since AoIP generally travels on copper Ethernet cables, the RJ-45 connector is used for all terminations. Some AoIP system implementers, including Livewire, also use RJ-45 for analog and AES3 digital audio I/O with adapter cables converting to XLRs, phone, RCA, etc. While this minimizes the number of different connector types used in a facility and reduces the physical space required for connector panels, some engineers might not be comfortable with this approach.

The need to accommodate and retain compatibility with analog and AES3 digital audio will remain for some time at any AoIP facility. At the very least, live microphone signals will need to be converted from their native audio format. So until microphones and other audio sources come with native AoIP outputs, interface "nodes" will be needed.

Also note that, at least for the time being, AoIP equipment is not yet fully compatible among various vendors. Thus, settling on a single vendor is going to be necessary for each installation.

Engineers installing and maintaining AoIP systems will have to learn enough IP network engineering to have a basic understanding of the technology (or more, if they are so inclined, which will surely be career-enhancing in these times). This book covers most of what is needed for those basics, and suggests other resources to help you go further.

## 1.5 IMPLEMENTATION AND INTEGRATION

Given the advantages of scale provided by AoIP systems, it makes sense to make the AoIP domain as large as possible within a given facility. This implies that audio signals in other forms should be converted to IP packets as close to the source as possible.

The best place to do this in most studio configurations is at the studio mixing console(s) and/or the central patch bay (i.e., technical operations center, or TOC). Microphone outputs and signals from other "legacy" audio sources can be immediately converted to digital audio form (if they aren't already) and packetized as IP. Once in the IP domain, these signals can be addressed and routed to any other location on the network. This can include destinations within the confines of a facility via LAN, or anywhere in the world via a gateway to the wide area network (WAN).

Another advantage of this approach is that a mixing console can act as a router. In other words, because any input on the console can have a unique IP address, it can be connected to any AoIP source on the network. (Even more amazing to veteran audio engineers is that this can be accomplished even though the entire console is connected to the network via a single Ethernet cable.) A central switching control unit (typically a PC) can assign these I/O connections, or the mixing console itself can have a control interface for this purpose. In addition, standalone hardware switch controllers can be distributed around the facility, essentially duplicating the appearance and function of traditional router-control panels.

Certainly, the studio mixing console setup can also be equipped with traditional analog (mic/line) or AES3 inputs as well. Because these sources are converted to IP and placed on the network, they are available to any location in the facility that needs them. (See Figure 1.1.)

Consider also how PC-based audio playout/automation systems can be interfaced to such a system. Rather than their audio outputs being directed through PC sound cards to traditional audio inputs, the automation system can be fitted with an IP driver that provides a software interface between the PC audio and the IP network directly in the AoIP domain. This not only maintains high audio quality, but cuts costs in the automation system since no (or at least fewer) sound cards are required. The IP interface can also carry control data and content metadata as well, eliminating the need for separate data links between devices.

Moreover, a *single* IP driver interface between an automation system and an IP routing architecture can carry many independent audio channels (up to 24 stereo for Livewire), whereas a traditional switching system would require a crosspoint (plus wiring) for each sound card input and output. The combined hardware savings (sound cards + crosspoints + wire + installation) accruing in a large facility is likely to be substantial.

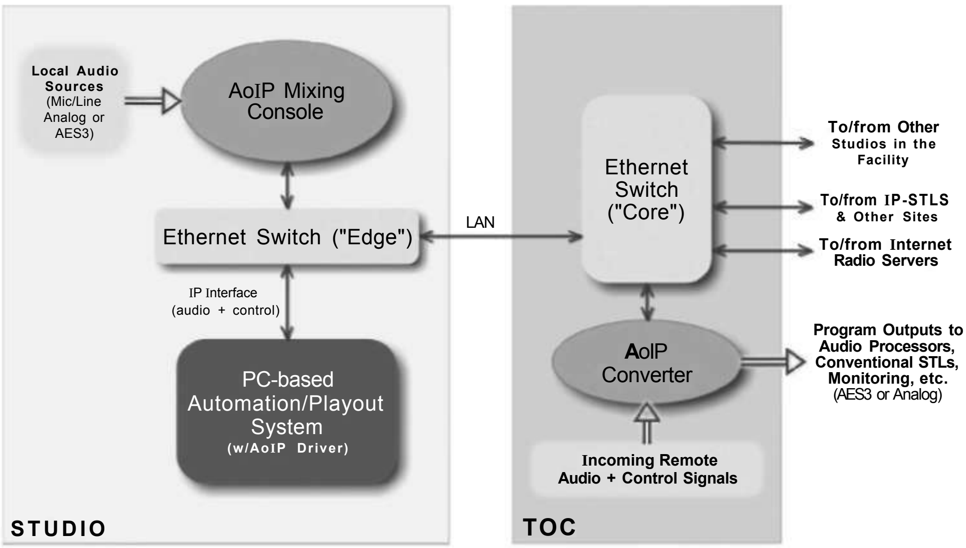


**FIGURE 1.1**

Conceptual block diagram of a typical AoIP-based broadcast studio facility, showing one studio and a TOC.

As Figure 1.1 indicates, a typical AoIP facility includes multiple Ethernet switches, usually arranged with one large ("core") switch in a central room, and smaller ("edge") switches placed as needed in other rooms around the facility. Such distributed routing intelligence improves performance and also provides redundancy in case of switch failure.

The proliferation of VoIP and other real-time applications via IP have spawned broad implementation of *nonblocking* architecture in Ethernet switches. This approach eliminates data collisions within a switch by ensuring adequate capacity for $n \times n$ connectivity—that is, any input on the switch can always be connected to any output on the switch, under any usage—through the switching fabric. Mission-critical performance is thereby maintained by using Ethernet switches that implement a nonblocking design, and when properly implemented within an AoIP system, switch capacity will never be exceeded.

In some AoIP facilities, the functions of the Ethernet switch can be replaced by an IP *router*. Simply stated, both the Ethernet switch and the IP router perform the same function of getting payload packets to and from their proper locations, but in different ways. Truly standard AoIP systems won't care which is used, however. We discuss the nature and differences of switches and routers in detail in the upcoming chapters, including applications where one or the other may be preferred.

The use of Ethernet switches and IP routers by mission-critical and other high-reliability telecom operations has driven major manufacturers to provide excellent around-the-clock and overnight-replacement support. Note also that as a facility grows, it may need to replace older switches with newer models; the fact that IP and Ethernet are ubiquitous standards means that all upgrades will remain backward compatible. Meanwhile, Moore's Law ensures that as such new hardware becomes available, price/performance ratios will continually improve. It's all good.

The AoIP domain is also extending beyond the studio. Figure 1.1 shows how AoIP is converted back to AES3 (or even analog) for program outputs' connection to conventional studio-to-transmitter links (STLs), but the diagram also indicates that an STL could carry AoIP to the transmitter site (via a WAN or other dedicated link). Whether leased from telco or using a station-operated radio frequency (RF) path, if adequate bandwidth is available, multiple audio channels, control, and metadata can all be carried via IP on the link—bidirectionally, if desired—with minimal latency.

**WHITHER THE ETHER?** Ethernet is a surprisingly congruent name for a technology initially intended purely for the IT world, but now serving AoIP in broadcast studios. How did that come to be?

Ethernet was named by its inventor, Robert Metcalfe. He had been involved in a radio data network project in Hawaii called ALOHA. The first Ethernet was a bused coax that carried data packets similar to the way ALOHA had sent them over the "ether."

Metcalfe was using the word jokingly. For many years after James Clerk Maxwell's discovery that a wave equation could describe electromagnetic radiation, the aluminiferous

*(continued)*