Modular, intuitive control.

element

AXIA
A TELOS COMPANY

Meet Element, the modular broadcast control surface from Axia Audio. With configurations as small as 4 faders and as large as 40, Element is versatile enough for all your air, production, news and edit applications. Element is designed to fully exploit the benefits of the Axia IP-Audio network. Share live audio between studios, eliminate PC sound cards, and instantly reconfigure shows and setups, all using standard Ethernet.

Element provides flexibility and features not found in control surfaces costing three times as much. Like advanced integration with Telos phone and codec products. Four auxiliary sends and two returns. Three-band SmartQ™ equalization on every channel, and Omnia dynamics processing on every microphone source. Powerful yet easy-to-use talkback and monitoring capabilities. Automatic mix-minus outputs on every channel. Opto-isolated GPIO logic ports for machine and remote control.

Because of its advanced DSP engine architecture, Element's four program outputs can be configured to mix in either stereo or surround (5.1+2). Element's on-screen status display is the most comprehensive anywhere, with all meter, timing and options information presented in a friendly, instantly accessible format using a standard VGA LCD monitor. Configuration and management can be done remotely from any PC browser — even from off-site.

But all of this functionality only scratches the surface (ahem). Element allows you to network all of your facilities' audio peripherals for greater efficiency. You'll realize the benefits of reduced studio cost and complexity, eliminating expensive cable bundles, patch bays, line switchers, DA's and even those huge mainframe-type routing switchers. Replace all this and more with a standard Ethernet switch and a handful of CAT-6 cables.
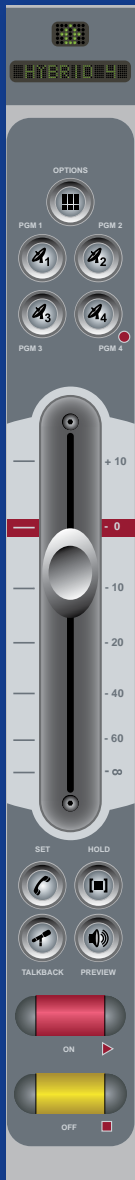
Premium components are used throughout, like these silky-smooth, long-life conductive plastic faders.

A variety of module choices lets you build exactly the right console to suit your needs.

No wimpy displays here: Element's full-screen readout provides comprehensive status at a glance.

You've probably noticed the recent explosion in the number of consoles on the market. Seems like everyone and his brother now makes a control surface! There's a bewildering amount of choices. So, what sets Element apart from all the rest?

Element isn't just another pretty surface. It's an integral part of the Axia IP-Audio networking system. This means you're not just buying a console — you're getting a **complete audio and data routing infrastructure**, with a sophisticated switching network thrown in (for less than the price of some others' standalone consoles).

Axia's IP-Audio networking system doesn't just switch audio. It allows real-time audio transport, machine logic controls, program associated data and standard network traffic (like messaging and file transfers) to be combined on a single network powered by robust, cost-effective Switched Ethernet.

Thanks to this flexible architecture, **Element has no dedicated faders.** Each fader can accommodate any source available on the network. In very large facilities, there might be *hundreds* of sources — and your operators can access any of them with the push of a button.

Element also saves fader-source assignments and other configuration data as "profiles" that can be recalled for **instant reconfiguration**. This ability enables a single Element to be used throughout the day to perform different, specialized air or production tasks without any downtime. Break-away segments, in-studio performances or interviews requiring different sources can be loaded instantly.

The morning team likes the mics on the left and the phones on the right? Let them lay out the board just they way they want it. Mid-day talent wants a completely different layout? No problem.

And here's another thing traditional consoles could never do: because Axia networks are intelligent computer based routing systems, **machine logic always follows source audio.** This means that as sources are moved from one fader to another – or one studio to another – no logic reconfiguration is necessary. Simply load a source on any fader, and that fader's controls are immediately communicating with the source.

What else can Element do? Read on.

Simply powerful.

Power lurks just below the surface. Touch the Options button above any fader to call up additional functions. On-screen navigation makes option selection a breeze. Power users will love those extra features, but everyone will love the friendly, simple controls.

All options can be saved for instant recall when the operator loads a Show Profile. Source selections, EQ and mic dynamics, back-feed settings, Aux sends and returns and more are all accessible with a single keypress. Easy and powerful: you can have it both ways.

The Overview (literally).

1   Element's long-life conductive plastic faders were chosen for their ability to deliver years of worry-free service. Any Element fader channel can accommodate any input source. Each fader's Options button grants access instantly to expanded channel features and frequently used functions.

2   Assignment buttons for the four main output buses (Program-1, 2, 3 and 4) are located at the top of each fader for quick access. Program-4 doubles as a Record/Phone bus to make short work of recording phone bits and setting up conferences.

3   Each channel has its own 10-character alpha display plus a channel number/Status Symbol icon. For static sources like microphones and CD players, the text is fixed, but when computer-based audio sources are connected, the text displays can show the names of currently active program elements — songs, artists, *et cetera*.

4   Channel Options keys provide quick access to a channel's Source assignment, EQ, Dynamics, Pan, Mix-Minus and Aux Sends settings. These keys become active whenever a channel's Options key is pressed. Below the Channel Options keys, a set of keys are provided for convenient in-console control of external profanity-delay units.

5   Unique context-sensitive SoftKnobs let operators easily modify on-screen options and settings, adjust EQ curves, scroll lists of available sources, and trim Aux Send and Aux Return gain. Element's on-screen display shows the action assigned to each knob.

6   Global options & timer control keys let operators recall Show Profiles, select monitor and metering preferences and customize the way Element works. Timer functions help users keep track of timed events with on-screen elapsed and remaining time displays.

7   Talk shows can get hot and heavy in seconds flat. That's why Element's talkback & preview keys are conveniently placed next to each fader's on/off controls. An extremely flexible and full-featured talkback system lets board ops and producers quickly communicate with hosts, remote talent and guest callers.

8   We've seen what board operators can do when they get rolling. So we designed every control for heavy use and error-free operation. Like these On and Off buttons with integrated guards. Who says "heavy duty" has to look like the controls of a forklift?

9   Good-looking, scuff-and-scar resistant coating is featured on end pieces and wrist rests to help guard Element against the rigors of studio life (and heavy-handed jocks).

10 When using a Telos TWOx12 or Series 2101 broadcast telephone system, the optional Telos Phone Control module pictured here provides enhanced call handling features. Two dedicated faders flank a line selector which includes exclusive Status Symbol readouts that mirror those on the phone controller for at-a-glance visual caller management.

11 Rich, flexible monitor section with individual volume controls for Control Room and Studio monitors, headphones, and Preview speakers. Dedicated selector keys let talent quickly listen to any Program or Aux bus, or a host of external location feeds. Monitor options include "split" headphone operation, Preview in Headphone override and more. These options are saved along with the other settings so talent always gets the right preferences when loading a show.

12 Control wheel and direction keys help talent navigate quickly through on-screen options.

13 A built-in numeric keypad (with # and * keys) lets operators place calls with phone systems or codecs attached to the Axia IP-Audio network without leaving the control surface. Also provides numeric input during surface setup.

14 No more scrambling to start recorders, reassign phones and mics to record buses, or construct custom mix-minuses to capture on-the-fly bits or interviews — a simple push of the Record Mode button performs these tasks for you.

Large meters with peak indicators help operators maintain optimal program levels.

Day/date and clock displays are front and center. Can be set for 12- or 24-hour operations.

Large analog clock also contains a "last minute" visual indicator ring for the programmable countdown timer.

Far right meter can be set to measure any source assigned to the Preview channel.

Programmable count-down time ticks off the seconds to break-aways and network joins. Or use it for ultra-precise timing of live productions.

Event timer keeps track of elapsed time. You decide which events trigger automatic starts and resets, or take control manually.

Status panel provides quick appraisal of operation modes. Tallies light to advise operator of Talkback activity, Record Mode operation, when external delay is active, and more. No more forgetting that the mic is open!

Planning for surround broadcasts or production? Element can handle it. Flexible software architecture lets you mix audio in 5.1+stereo format.
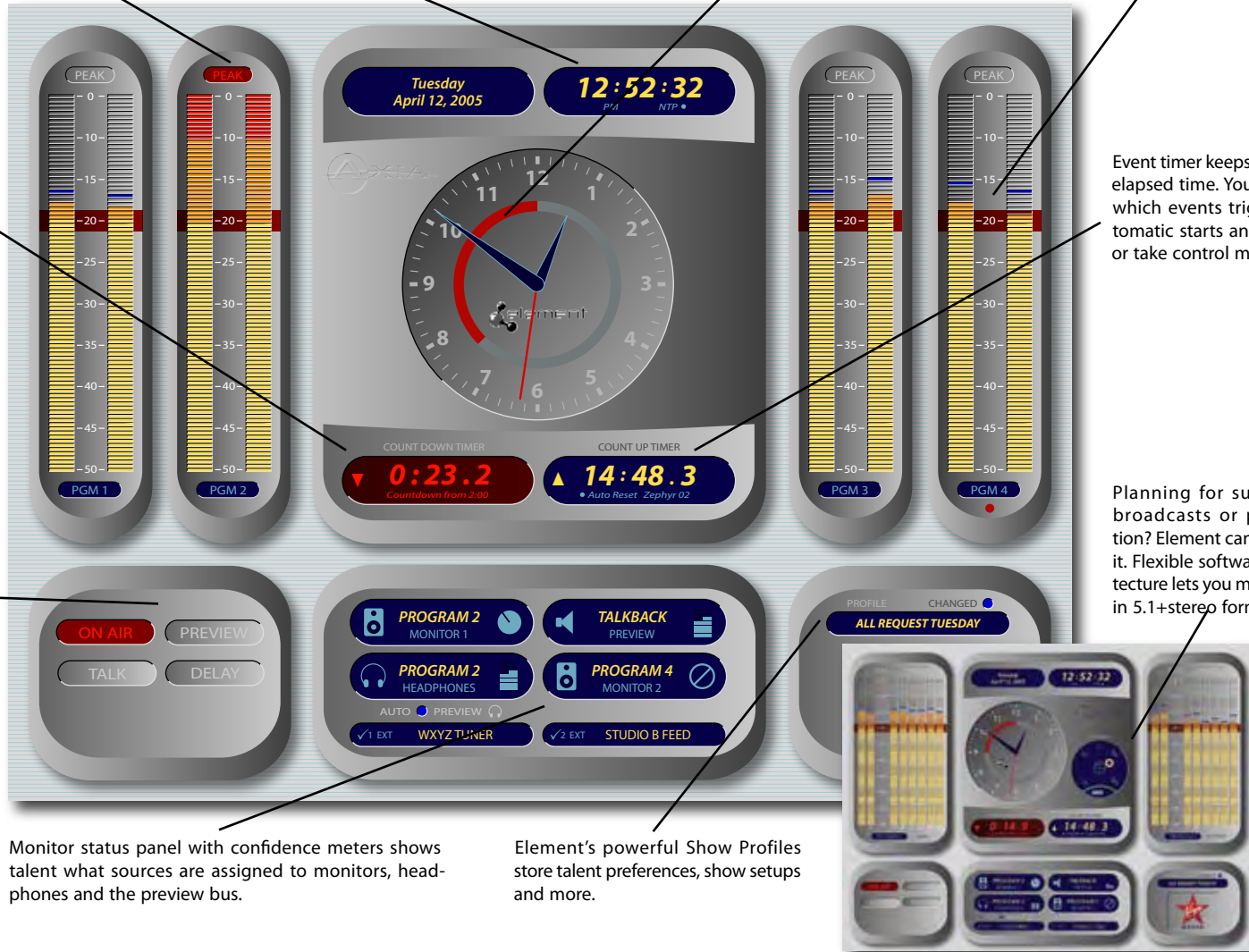
Monitor status panel with confidence meters shows talent what sources are assigned to monitors, head-phones and the preview bus.
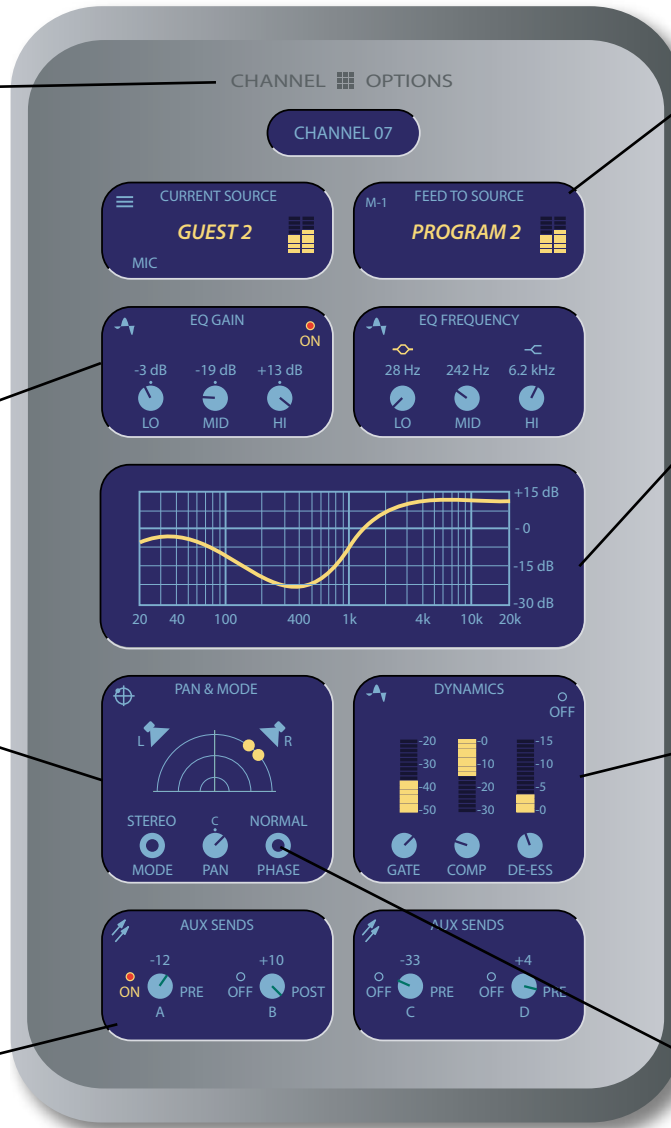
Element's powerful Show Profiles store talent preferences, show setups and more.

Tuesday April 12, 2005

12 : 52 : 32   PM   NTP

11 12 1
10 2
9 element 3
8 4
7 6 5

COUNT DOWN TIMER ▼ **0 : 23 . 2** Countdown from 2:00

COUNT UP TIMER ▲ **14 : 48 . 3** Auto Reset Zephyr 02

PEAK — 0 —10 —15 —20 —25 —30 —35 —40 —45 —50 **PGM 1**

PEAK — 0 —10 —15 —20 —25 —30 —35 —40 —45 —50 **PGM 2**

PEAK — 0 —10 —15 —20 —25 —30 —35 —40 —45 —50 **PGM 3**

PEAK — 0 —10 —15 —20 —25 —30 —35 —40 —45 —50 **PGM 4**

ON AIR   PREVIEW
TALK   DELAY

PROGRAM 2 MONITOR 1
TALKBACK PREVIEW
PROGRAM 2 HEADPHONES
PROGRAM 4 MONITOR 2
AUTO ● PREVIEW
✓1 EXT   WXYZ TUNER
✓2 EXT   STUDIO B FEED

PROFILE   CHANGED ●
ALL REQUEST TUESDAY

Press the Options key above any fader and you're treated to a variety of settings.

Need a little more treble on that mic? See and adjust EQ settings instantly.

Set the stage the way you like it, with pan and stereo balance controls.

Four Aux Send buses give you plenty of space to apply outboard effects or construct custom IFBs.

CHANNEL ☷ OPTIONS

CHANNEL 07

CURRENT SOURCE
GUEST 2
MIC

FEED TO SOURCE
M-1 *PROGRAM 2*

EQ GAIN ON
-3 dB -19 dB +13 dB
LO MID HI

EQ FREQUENCY
28 Hz 242 Hz 6.2 kHz
LO MID HI

+15 dB
- 0
-15 dB
-30 dB
20 40 100 400 1k 4k 10k 20k

PAN & MODE
L R
STEREO C NORMAL
MODE PAN PHASE

DYNAMICS OFF
-20 -0 -15
-30 -10 -10
-40 -20 -5
-50 -30 -0
GATE COMP DE-ESS

AUX SENDS
-12 +10
ON PRE OFF POST
A B

AUX SENDS
-33 +4
OFF PRE OFF PRE
C D

If a backfeed is active for the chosen source, this window indicates its content.

No need to guess at EQ curves: this graphical display shows board ops the curve they've dialed in.

Built-in dynamics processing by Omnia for all mic sources lets you compress, gate and de-ess to your heart's content. Customized settings for each talent load with their personal profiles.

Got an out-of-phase remote line? Set it right with built-in phase controls.

GLOBAL ☷ OPTIONS
H/P PROCESS
HEADPHONE EQ ACTIVE
+15 dB
- 0
-15 dB
-30 dB
20 40 100 400 1k 4k 10k 20k
LO MID HI
-3.5 dB 43 Hz -21.2 dB 417 Hz +14.5 dB 2.2 kHz
GAIN FREQ GAIN FREQ GAIN FREQ

OMNIA PRESET IN H/P ACTIVE
NONE
ROCK & ROLL
JAZZ
NEWS / TALK
NUCLEAR

Dedicated EQ and dynamics processing for talent's headphone channel lets them hear "air sound", even when delay makes off-air monitoring impossible.

GLOBAL ☷ OPTIONS
PROFILE
LOAD PROFILE *ALL REQUEST TUESDAY*
(NONE)
ALL REQUEST TUESDAY
OVERNIGHT
AUTOMATION
BOB'S PRODUCTION PROFILE
NETWORK NEWS
WEEKEND SHOW
THE SAM & KATY SHOW
LIVE SPORTS REMOTE
THE BLUES SHOW
BOB'S LIVE SHOW

Reconfiguring Element for different talent or show setups is as easy as navigating this list. Push a button and changes are loaded instantly.

Freedom of Choice.

Who likes limitations? Not us. We bet you don't, either. Which is why Element lets you design your control surface exactly the way you like, to suit your unique operational needs.

You can choose from a wide variety of frame sizes to accommodate any number of faders from four to forty. Element even allows you to construct "split" control surfaces — two or more independent frames linked together to control a single mixing engine. Just imagine the possibilities: put an eight-fader mixing panel in an adjacent performance studio, or give the news announcer his own four-fader surface.

Talent takes control of 20-position, 12-fader Element at Triangle FM, Paris, France.

24-position, 18-fader split-frame Element at Algoa FM, Port Elizabeth, South Africa.

Each Element fader module includes an over-bridge-mounted informa-tion panel.

Many Element modules occupy four frame spaces. You're looking at the 4-Fader Line Module.

Need to talk to a host or guest? Press the Talkback key. Press more than one to talk to several locations simultaneously.

Light bulbs are strictly old-school. Element fea-tures LEDs in all lighted buttons.

Status Symbols give in-formation at a glance about mix-minus, phone lines, talkback activity and more.
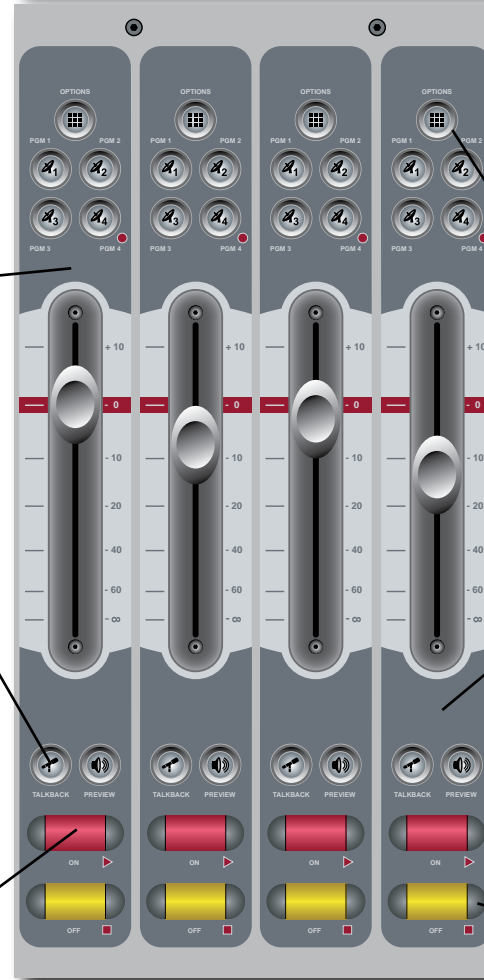
Need a different source? Press the Options key to choose a new source in seconds.

Mar-resistant Lexan® over-lays stand up to even the toughest daily abuse.

Stylish, integrated finger guards help ensure error-free operation.

Telephone Interface Modules

Telos Phone Controller module lets you control TWOx12 or Series 2101 phone systems right from the board.

Say goodbye to mix-minus hassles. Element automatically generates backfeeds for any source that needs one. Status Symbols tell you when they're active.

4-Fader Phone module has special keys for phone systems, but will accommodate any source from the IP-Audio network.

Special keys let you transfer calls to a handset or "busy out" all lines in preparation for contests.

More Status Symbol icons inform operators of line and caller status with just a glance.

Finished with the bit but not the caller? Convenient Hold key keeps them warm until you're ready.

No need to remember which caller is next in queue - Element does it for you.

Ready to dump a call? Just hit the Drop key and it's gone.

Set key transfers phone calls from the control surface to phone system's handset.*

SmartSwitch Modules are programmable, and are available in 10- and 5-button sizes, includes overbridge display.
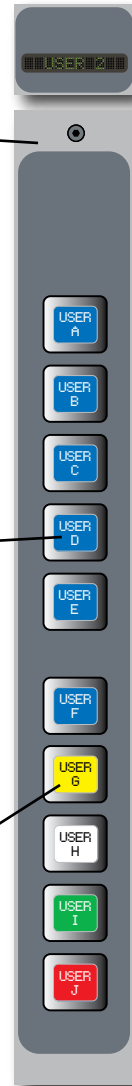
Use the Panel Designer feature built into Axia PathfinderPC™ to design complex, conditional multi-salvo router events that launch at the touch of a Smart-Switch button. Button text can change with route condition.

Button assignments and backlit LCD button text can change dynamically with Show Profiles. Change colors, too: eight different colors are selectable.

Film-Legendable Switch Module with fixed-function buttons is also available in 5- or 10-button sizes. Backlight colors are changeable, too.
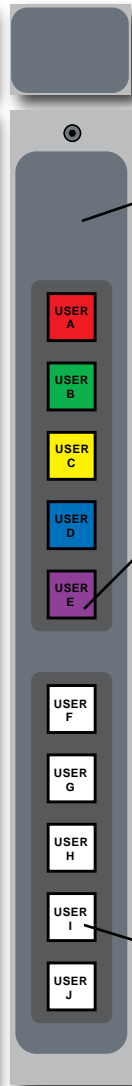
Axia PathfinderPC software lets you create custom router salvos that you can launch at the touch of a button.

Assign buttons with custom routing functions, audio device controls and more.

Speed keys let board ops quickly choose new sources, trim EQ, adjust mic dynamics and more.

Built-in controls for external profanity delay unit are much appreciated by talk show hosts.

Instantly communicate with talk studio hosts and guests.

Headphone controls allow operator to preview sources in one or both sides of the headphone channel.

Independent source selection for headphone, control room and studio monitors.

Two speed keys let you quickly monitor external audio feeds. Press and hold to select new feeds to listen to.

SoftKnobs work with on-screen display to help you fine-tune Element options.

Need to conduct an interview instead of playing music? Load a saved Show Profile in seconds.

Full-featured controls for event and count-down timers.

Record Mode automates setup of last-minute phone bits or interviews.
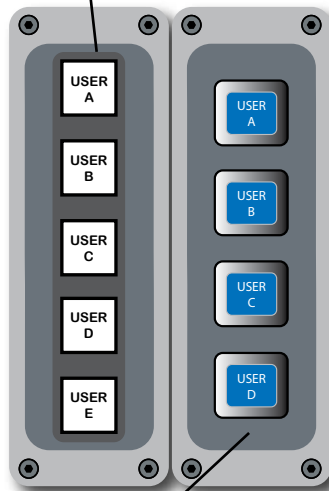
Numeric keypad lets you dial phones and codecs without ever leaving the board.

Control wheel and nav keys assist with making quick changes in options screens.

There's an Element module tailored for just about every operational need a board op might have. But what about studio hosts and guests? Don't worry, we haven't forgotten them. There's a variety of optional user panels designed for convenient in-studio use by guests, producers and hosts.

Film-legendable 5-button switch panel can be used at producer or interview positions to trigger recorders, playout devices or software functions.

This mic control/headphone panel with dedicated Mute and Talkback buttons gives talent full control of their position. Volume/selection knob allows users to select their headphone monitor source; alpha display confirms their choice.

Basic mic control panel provides remote control of mic on/off functions, and includes dedicated Mute and Talkback buttons.

4-Button backlit LCD SmartSwitch Panel gives producers and talent one-touch remote access to often-used machine-control or software functions.

Headphone Selector Panel with alpha display allows users to select their own headphone monitor source. Two preset buttons let talent set and quickly recall frequently-listened-to sources.

Special Producer's accessory panel provides two Talkback keys for communication with control room board op, studio guests, or both. Remote control of mic on/off functions, too.

Mix-minus Made Easy. Finally.

One of Element's most popular features is the way it handles mix-minus for phones and codecs — every channel can provide a mix-minus output without talent intervention. Select a phone or codec source and the backfeed is automatically generated, eliminating a source of confusion and error that's troubled board ops for decades.
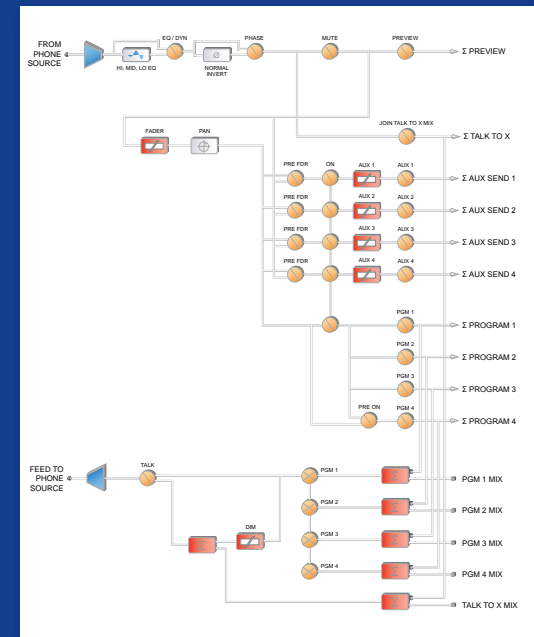
Phone interviews used to mean reassigning channels and setting up an audition bus to feed the hybrid. With Element, assigning any fader to the "Phone" bus generates channel-specific "N-1" and routes it back to the phone hybrid or remote codec. This function is available simultaneously on __all__ of Element's fader positions.

Audio Routing made simple.
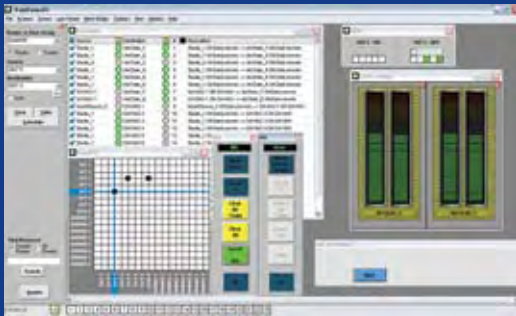
Optional PathfinderPC routing control software gives you the tools you need to build complex audio routing configurations.

PathfinderPC runs on a Windows server platform. Client applications can be customized so that each operator sees a simple, intuitive interface with <u>only</u> the controls they need. Prefer to use tangible buttons rather than a PC interface? PathfinderPC's PanelBuilder integrates with Element's switch modules; push a button to load a salvo, send a conditional logic message, insert talk into a path, etc. Control is a beautiful thing!

It's also easy to construct advanced, custom features — even patch around problems to keep you on the air. Want to monitor a particular output for silence? Tell PathfinderPC to monitor that output's network stream and switch to an alternate path if its level falls below your specified threshold. Then have PathfinderPC send a contact closure to trigger a device, and send an email to your Blackberry letting you know that Pathfinder took action and all is well.

- ▲ From 4 to 40 fader channels, each with instant, unlimited access to any source. You can assign any type of source to any channel.

- ▲ Four main stereo outputs (Program-1 through Program-4), plus four stereo Aux sends and two Aux returns.

- ▲ 10-character alpha-numeric displays above each fader channel always show the current selected source.

- ▲ Each channel is also equipped with a Status Symbol™ display which can provide talkback, mix-minus, and other source-related communication information.

- ▲ Every channel has a stereo Preview ("cue") function, with a unique interlock system for fast, intuitive operation.

- ▲ Reconfigurable monitor section with reassignable controls that let operators instantly change monitored sources "on-the-fly."

- ▲ Flexible talkback system lets board op talk to hosts, studio guests, external feeds — any source with an associated backfeed.

- ▲ Software control of options such as EQ, aux sends and returns, pan and balance and other features delivers maximum flexibility without panel clutter or intimidating controls.

- ▲ Built-in Omnia dynamics processing lets operators combine compression, de-essing and expansion with EQ to "sweeten" microphone sources.

- ▲ A unique Record Mode enables one-button setup of record mixes for phone bits or off-air interviews.

- ▲ Consolidated user display conveys meter, clock, timer and monitor source information at a single glance. Use any external VGA monitor you choose, from a 12" LCD to a DLP wall projector!

- ▲ Precision timer and clock functions, including an event timer that can be triggered by pre-defined sources, a countdown timer with last-minute alerting and a time-of-day clock that can be synchronized to network time using NTP.

- ▲ Instant reconfiguration capabilities allow each user to set and save a customized personal profile that can be instantly recalled. Custom show profiles can be saved for interview shows, music-intensive programming, call-in talk shows, etc.

- ▲ Element functions can be accessed remotely for configuration, management and diagnostic purposes using any standard Web browser.

- ▲ Built-in 5.1 mixing capabilities for production or digital Surround broadcasting.

- ▲ Optional Telos Phone Control module provides direct, on-the-console access to Telos TWOx12 or Series 2101 advanced functions.

- ▲ Control keys for external profanity delay unit can be slaved to any user-supplied PDU using GPIO closures.

- ▲ No audio passes directly through Element — all mixing and processing is performed by the Axia StudioEngine. Element connects to the Axia network using just one cable.

The Mixing StudioEngine.

As we've said elsewhere, Element resembles a traditional broadcast console, but it isn't one; no audio is actually mixed by or even passes through its faders. Instead, think of Element as a "remote control" for the Axia StudioEngine.

Because the Axia system is a network at heart, all connected sources and destinations are accessible from anywhere in the networked facility, making cross-point switching possible from any source to any destination. This networked architecture opens the door to a new way of mixing and processing audio.

The Axia StudioEngine is an extremely powerful mixing and processing device, based on a blazingly-fast Intel processor that can out-perform even the largest dedicated-DSP embedded designs. The StudioEngine accesses audio streams, modifies them, and then presents the resulting streams back to the network as program output (or monitor output, or mix-minus output, *et cetera*). This approach is ideally suited to a network-based audio architecture since all input and output streams are routed through a Gigabit Ethernet port (more on this in the next few pages).

To deliver the reliability and ultra-low latency required, we equipped the StudioEngine with a fast, robust version of the Linux real-time operating system. Then we optimized our engine processing program so that total input to output latency is just a few hundred microseconds. In fact, each StudioEngine has so much CPU power, it can outperform the very largest digital or router-based consoles, with multiple simultaneous inputs, outputs, mix-minus feeds, monitor signals, etc. It can even provide EQ for multiple channels, and has the headroom to support future features. One StudioEngine is used for each surface, so a typical facility will have several Engines.

Along with all that power and flexibility, the Axia Element and StudioEngine also help you reduce studio cost and complexity. You can network all of your facilities' audio peripherals for greater efficiency. And you can finally enable computer-based studio applications – playout and traffic systems, logging applications and more – to exchange data, machine control and even Program-Associated Data over a common network.

Snapshot: Radio Free Asia

"RFA is a private, non-profit corporation that broadcasts news and information in nine native Asian languages.

"The Axia installation is in our Bangkok office, where we handle production of hour-length program blocks. The consoles we had been using were approximately ten years old and rapidly approaching obsolescence. We now have Axia Element surface controllers, each with four faders in an eight-position frame.

"Our on-air talent has been extremely impressed with the Element controllers — due in large part to the fact that the user interface is far more intuitive compared to the previous consoles. The Elements are about as easy to use as one could hope for. Because they're so easily configured, the staff has started to take advantage of the ability to set up the consoles to their liking.

"In reality, there was no learning curve. It took us about 45 minutes to train each broadcaster on the use of the system — and that included powering the system up, loading their individual profiles, and actually using the Element controllers."

— David Baden, Chief Technology Officer, *and* John Penovich, Senior Technician

By now, you probably have a few questions, like "How can I use Ethernet for audio?" and "How do I interface conventional audio equipment with an Axia audio network?"

All Axia products are based on Livewire™, a pioneering technology invented at Telos Systems to convey low-delay and high-reliability audio over switched Ethernet. With Livewire, a single Ethernet cable carries real-time uncompressed digital audio, device control messages, Program-Associated Data, and even routine network traffic. An entire facility can be wired in hours, instead of weeks. Expanding or modifying your system is simple thanks to Axia's inherent scalability and modularity. (For in-depth information about how Livewire works, visit www.AxiaAudio.com.)

The Axia family of Audio Adapter Nodes are used to access and control your sources. There are now five versions: an Analog Node, an AES/EBU node, a Microphone node, a Router Selector node, and a GPIO node.

Each of these 1RU units is equipped with a 100Base-T Ethernet connection. Simply connect your audio devices to an Axia audio node; when a node is connected to the Livewire network, it advertises that its audio sources are available for use, allowing any users access to them. Installing Axia audio nodes is easy — just place them near your audio source and destination devices, and distribute them throughout your facility wherever it's convenient.

For example, a microphone node placed in a studio can collect audio from microphones and also provide outputs to associated studio monitors and headphones. Another node in the central equipment room can collect audio from network feeds, codecs and other shared sources for system-wide use while providing convenient outputs for audio processors and other terminal-room gear.

The Microphone, Analog Line and AES/EBU nodes feature a multi-character LED front panel with confidence metering that shows the audio activity on each of the inputs and outputs, and can also display text labels to ease configuration and identification of audio sources during installation.

To ensure ultra-reliable network operations and extremely low delay, Axia audio nodes run a version of Linux on an embedded processor, and a built-in web server in each node gives you remote configuration and control — in an intuitive, easy-to-understand manner — using any standard Web browser interface.

The Microphone Node has eight professional-grade microphone preamps with selectable Phantom power and software-adjustable gain. There are also eight balanced analog line outputs to conveniently deliver headphone and studio monitor feeds back to the talent. Inputs use XLR connectors; outputs are on easy-to-install RJ-45's that are compatible with Radio Systems' StudioHub+® pre-wired audio connectors.

Microphone Node, 8 preamp inputs and 8 stereo outs.

The Analog Line Node has eight balanced stereo inputs and eight balanced stereo outputs, all on RJ-45 connectors. Each input is switchable to accommodate either consumer-level -10 dBv or professional level +4 dBu. The short-circuit protected outputs can deliver up to +24 dBu before clipping. We make use of the very best quality A/D/A converters and low-noise components, so that each Analog node provides superior audio performance for high-end studio use.

Analog Line Node, 8 stereo ins and 8 stereo outs.

The AES/EBU Node provides eight stereo AES3 inputs and eight AES3 outputs. Sample-rate conversion is available on all inputs; the unit can also sync the Axia network to a house clock. Like the Mic and Analog Line Nodes, the AES Node displays confidence metering for each of its inputs and outputs on the front panel, which doubles as a system configuration display.

AES/EBU Node, 8 stereo AES3 inputs and 8 AES3 outs.

The Router Selector Node resembles the Source Selectors used with expensive cross-point audio switchers. The LCD screen lists available sources, which can be browsed and selected with the scroll wheel; eight "radio buttons" provide instant access for frequently-used sources. Unlike a Source Selector, however, the Router Selector node has audio output direct to headphones, and analog and AES3 outputs. It even provides a convenient analog and AES3 input, making it ideal for production or news studios where operators typically both create and play audio streams.

Router Selector Node, 1 input 1 output, mixed signal.

Finally, the networked GPIO Node provides eight logic ports for machine control, each with five opto-isolated inputs and five isolated outputs. A logic port can be associated with any input or output and routed along with the audio. You can use this node to add more GPIO ports wherever needed when Element's standard eight ports are not enough.

GPIO Node, 8 logic ports, each with 5 ins and 5 outs.
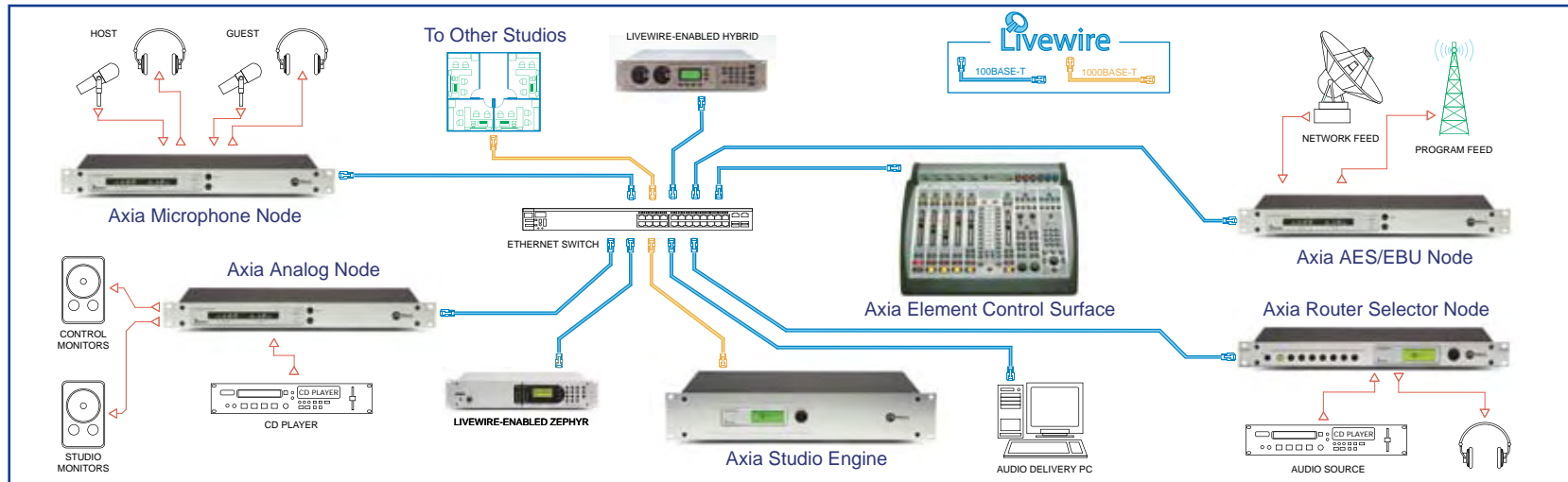
Axia components are "building blocks" that you can use to build networked broadcast audio systems. In fact, a complete studio is amazingly easy to build; once you've identified all of your audio sources, all you need are some Axia audio nodes and an Element to connect and control them. And an increasing number of professional audio devices are being designed with Livewire interfaces that connect directly to IP-Audio networks. Axia eliminates expensive, labor-intensive discrete cable runs because Axia components connect using standard CAT-6 Ethernet cable.

Axia networks are based on switching Ethernet hubs of the same type used to power the high-speed backbones of mission-critical enterprises like airlines, financial institutions, medical data retrieval systems and military communication networks. We require this high standard because broadcasting, too, is mission-critical; this bulletproof reliability ensures that Axia IP-Audio networks meet the demands of 24/7 operation.

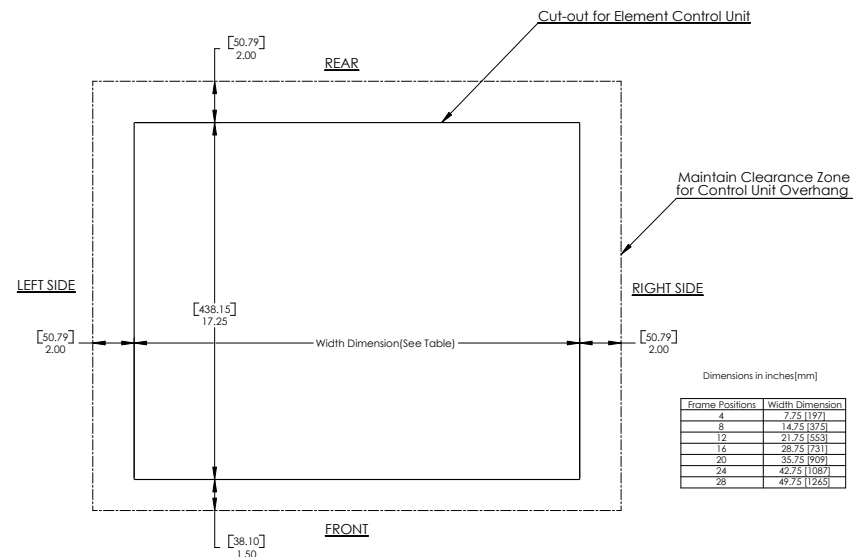The block diagram above illustrates a typical studio. A 100Base-T segment (shown in blue) can carry 25 stereo channels of 48 kHz, 24-bit linear PCM audio in both directions; each Gigabit link (in yellow) can carry 250 channels in both directions, with a system limit in excess of **10,000 stereo channels.** Gigabit fiber links can be used to connect studios in separate buildings, or to connect all of your studios to a central core switch.

Axia networks can also be configured with redundant components to create "self healing" networks and eliminate single points of failure. A second, fully redundant core Ethernet switch, backup power supplies on core and edge switches, even double data connections between edge and core switches can be implemented.

And to customize control of this sophisticated audio routing network, we offer PathfinderPC®, a powerful software program which controls every Axia Node in your plant, allowing you to define automated switching events, construct custom software control panels, and change between presets manually via PC, on a daypart schedule, or via an external trigger.

▲ Single-cable CANBus (Control Area Network) power/data connection between control surface and power supply/CPU.

▲ All switches, faders, displays and encoders selected for extreme reliability.

▲ Rack-mount 48 VDC power supply unit contains Element CPU plus eight GPIO logic ports and four CANBus ports for split-frame console configurations and powered studio accessory panels.

▲ Each of Element's eight GPIO logic ports has five fully opto-isolated inputs and five outputs, for a total of 80 logic signals per GPIO unit. More GPIO ports can be added using more GPIO Nodes, if desired.

▲ Works with your favorite external VGA display to present meter, information and configuration screens.

▲ Time-of-day clock synchronizes to network clock (NTP) if desired.

▲ Modular design allows for easy maintenance and expansion. Modulesh can be replaced during operation (hot swapping) with no effect on any existing audio assignments.

▲ Built-in HTML server allows secure remote access from anywhere (for control and configuration) via standard web browsers.

Cut-out for Element Control Unit

REAR

[50.79] 2.00

Maintain Clearance Zone for Control Unit Overhang

LEFT SIDE

RIGHT SIDE

[438.15] 17.25

[50.79] 2.00

Width Dimension(See Table)

[50.79] 2.00

Dimensions in inches[mm]

Frame Positions	Width Dimension
4	7.75 [197]
8	14.75 [375]
12	21.75 [553]
16	28.75 [731]
20	35.75 [909]
24	42.75 [1087]
28	49.75 [1265]

FRONT

[38.10] 1.50

Element has a sleek, low-profile design. When a cutout is provided in the furniture, Element stands only 8 cm above the counter top at its tallest point— allowing unobstructed views of monitors and guests. All dimensions are given in both inches and millimeters.

Rack-mount power supply contains eight GPIO logic ports for equipment control

Hinged bridge gives easy access to Element's connection panel. One-cable connection to power, CPU.

Module installation and removal couldn't be simpler. Single Ethernet cable connects modules to frame.

Microphone Preamplifiers
- Source Impedance: 150 ohms
- Input Impedance: 4 k ohms minimum, balanced
- Nominal Level Range: Adjustable, -75 dBu to -20 dBu
- Input Headroom: >20 dB above nominal input
- Phantom power: +48 VDC, switchable
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling

Analog Line Inputs
- Input Impedance: >40 k ohms, balanced
- Nominal Input Range: Selectable, +4 dBu or -10 dBv
- Input Headroom: 20 dB above nominal input
- A/D Conversions: 24-bit, Delta-Sigma, 256x oversampling

Analog Line Outputs
- Output Source Impedance: <50 ohms balanced
- Output Load Impedance: 600 ohms, minimum
- Nominal Output Level: +4 dBu
- Maximum Output Level: +24 dBu
- D/A Conversions: 24-bit, Delta-Sigma, 256x oversampling

Digital Audio Inputs and Outputs
- Reference Level: +4 dBu (-20 dB FSD)
- Impedance: 110 Ohm, balanced (XLR)
- Signal Format: AES3 (AES/EBU)
- AES3 Input Compliance: 24-bit with selectable SRC, 32 kHz to 96 kHz input sample rate capable.
- AES3 Output Compliance: 24-bit
- Digital Reference: Internal (network timebase) or external reference 48 kHz, +/- 2 ppm
- Internal Sampling Rate: 48 kHz
- Output Sample Rate: 44.1 kHz or 48 kHz

Frequency Response
- Any input to any output: +/- 0.5 dB, 20 Hz to 20 kHz

Latency
- Analog Input to Analog Output, 2.75 ms. including network, converters, and mixing process
- Digital Input to Digital Output, 1.75 ms. including network mixing engine (ASRC off)

Dynamic Range

- Analog Input to Analog Output: 102 dB referenced to 0 dBFS, 105 dB "A" weighted to 0 dBFS
- Analog Input to Digital Output: 105 dB referenced to 0 dBFS
- Digital Input to Analog Output: 103 dB referenced to 0 dBFS, 106 dB "A" weighted
- Digital Input to Digital Output: 138 dB

Equivalent Input Noise

- Microphone Preamp: -128 dBu, 150 ohm source, reference -50 dBu input level

Total Harmonic Distortion + Noise

- Mic Pre Input to Analog Line Output: <0.005%, 1 kHz, -38 dBu input, +18 dBu output
- Analog Input to Analog Output: <0.008%, 1 kHz, +18 dBu input, +18 dBu output
- Digital Input to Digital Output: <0.0003%, 1 kHz, -20 dBFS
- Digital Input to Analog Output: <0.005%, 1 kHz, -6 dBFS input, +18 dBu output

Crosstalk Isolation and Stereo Separation and CMRR

- Analog Line channel to channel isolation: 90 dB isolation minimum, 20 Hz to 20 kHz
- Microphone channel to channel isolation: 80 dB isolation minimum, 20 Hz to 20 kHz
- Analog Line Stereo separation: 85 dB isolation minimum, 20 Hz to 20 kHz
- Analog Line Input CMRR: >60 dB, 20 Hz to 20 kHz
- Microphone Input CMRR: >55 dB, 20 Hz to 20 kHz

Power Supply AC Input

- Auto-sensing supply, 90 VAC to 240 VAC, 50 Hz to 60 Hz, IEC receptacle, internal fuse
- Power consumption: 90-130 VAC, 50/60 Hz, 3.2A (typical); 300 watts
  180-264 VAC, 50/60Hz, 1.6A (typical); 300 watts

Operating Temperatures

- -10 degree C to +50 degree C, <90% humidity, no condensation

Dimensions and Weight

- Element: Shipping weight varies. See Page 17 for detailed dimensions
- Element Power Supply/GPIO node: 3.5 inches x 19 inches x 13 inches, 10 pounds
- Studio Mix Engine 3.5 inches x 19 inches x 15 inches, 10 pounds
- Audio nodes (all): 1.75 inches x 19 inches x 10 inches, 6 pounds
- Router Selector node: 1.75 inches x 19 inches x 10 inches, 6 pounds
- GPIO node: 1.75 inches x 19 inches x 13 inches, 8 pounds

Snapshot: WOR

"Buckley Radio decided to move WOR to a new location, leaving behind studios we'd called home for over 50 years.

"WOR needed a state-of-the-art facility that was digitally based. I looked at the systems available and settled on Axia. I'd heard that the Axia IP-Audio system could give us the high-end features we needed. The more I learned about Axia, the more impressed I became with their routing switcher and consoles, and how well their network topology was designed.

"So I decided to break new ground and order the Axia consoles and routing setup, nine studios worth. It's been on the air for over a year now, and we love it.

"The decision to install Axia has proven to be a good one. The system worked out of the box. Installation time was cut way down. Connections are simple. The system, coupled with PathfinderPC routing software, is powerful. All the data switches used are top-line off-the-shelf items. Our operators keep raving about how easy things are to operate. Even our listeners tell us how good WOR sounds!"

— Thomas R. Ray III,
Vice President/Corporate Director of Engineering

With a little help from our friends.

A networked audio system doesn't just replace a traditional console and router — it improves upon them, by providing complete integration with PC-based audio delivery systems. Leading companies in our industry have realized the advantages of tightly integrated systems, and are making new products that reap those benefits.

Check AxiaAudio.com/partners/ to find out who else is partnering with Axia. Don't see your system listed? Ask your favorite supplier about becoming an Axia partner.

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