

Axia Training Manual



Revision 1.5 — December, 2006

Customer Service

We support you...

By Phone/Fax in the USA.

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- The address is Support@AxiaAudio.com.

Via World Wide Web.

- The Axia Web site has a variety of information which may be useful for product selection and support. The URL is <http://www.AxiaAudio.com>.

Feedback

We welcome feedback on any aspect of the Livewire products or this manual. In the past, many good ideas from users have made their way into software revisions or new products. Please contact us with your comments.

Updates

The operations of Element and StudioEngine are determined largely by software. Periodic updates may become available - to determine if this is the case, visit our web site periodically, or contact us for advice concerning whether a newer release is more suitable to your needs.

Our electronic newsletter has announcements of major software updates for existing products, as well as keeping you up to date on the latest Axia, Telos, and Omnia product releases.

To subscribe go to <http://www.axiaaudio.com/news/eNews.htm>.

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About This Manual

This manual is meant to help you, a new user of Axia IP-Audio equipment, become familiar with the operational capabilities of your system.

Included you'll find details about the ideas that motivated Livewire, the technology that powers Axia, and how you can use and benefit from it, as well as nitty-gritty details about wiring, connectors, and the like that will help you understand how your system works. You'll also find out how to operate the various pieces of on-air equipment in your system.

You'll also find technical documentation that explains and illuminates the actual construction of your studio systems.

Feedback is welcome

At Axia, we love to hear your feedback. If you find anything in this manual that you feel needs clarification or correction, please let me know: e-mail cnovak@AxiaAudio.com or write:

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A Note From The President of Telos

It's been a tradition since Telos' very first product, the Telos 10 digital phone system, that I share a few words with you at the beginning of each manual. So here goes.

In radio broadcast studios we're still picking up the pieces that have fallen out from the digital audio revolution. We're not using cart machines anymore because PCs are so clearly a better way to store and play audio. We're replacing our analog mixing consoles with digital ones and routing audio digitally. But we're still using decades-old analog or primitive digital methods to connect our gear. Livewire has been developed by Telos to provide a modern PC and computer network-oriented way to connect and distribute professional audio around a broadcast studio facility.

Your question may be, "Why Telos? Don't you guys make phone stuff?" Yes, we certainly do. But we've always been attracted to new and better ways to make things happen in radio facilities. And we've always looked for opportunities to make networks of all kinds work for broadcasters. When DSP was first possible, we used it to fix the ages-old phone hybrid problem. It was the first use of DSP in radio broadcasting. When ISDN and MP3 first happened, we saw the possibility to make a truly useful codec. We were the first to license and use MP3 and the first to incorporate ISDN into a codec. We were active in the early days of internet audio, and the first to use MP3 on the internet. Inventing and adapting new technologies for broadcast is what we've always been about. And we've always been marrying audio with networks. It's been our passion right from the start. In our genes, if you will. As a pioneer in broadcast digital audio and DSP, we've grown an R&D team with a lot of creative guys who are open-eyed to new ideas. So it's actually quite natural that we would be playing marriage broker to computer networks and studio audio.

What you get from this is nearly as hot as a couple on their wedding night: On one RJ-45, two-way multiple audio channels, sophisticated control and data capability, and built-in computer compatibility. You can use Livewire as a simple sound card replacement – an audio



interface connecting to a PC with an RJ-45 cable. But add an Ethernet switch and more interfaces to build a system with as many inputs and outputs as you want.

Audio may be routed directly from interface to interface or to other PCs, so you now have an audio routing system that does everything a traditional "mainframe" audio router does – but at a lot lower cost and with a lot more capability. Add real-time mixing/processing engines and control surfaces and you have a modern studio facility with many advantages over the old ways of doing things. OK, maybe this is not as thrilling as a wedding night – perhaps kissing your first lover is a better analogy. (By the way, and way off-topic, did you know that the person you were kissing was 72.8% water?)

While were on the subject of history... you've probably been soldering XLRs for a long time, so you feel a bit, shall we say, "attached" to them. We understand. But no problem – you'll be needing them for microphones for a long while, so your withdrawal symptoms won't be serious. But your facility already has plenty of Ethernet and plenty of computers, so you probably already know your way around an RJ-45 as well. It's really not that strange to imagine live audio flowing over computer networks, and there's little question that you are going to be seeing a lot of it in the coming years.

The 20th century was remarkable for its tremendous innovation in machines of all kinds: power generators, heating and air conditioning, cars, airplanes, factory automation, radio, TV, computers. At the dawn of the 21st, it's clear that the ongoing digitization and networking of text, audio, and images will be a main technology story for decades to come, and an exciting ride for those of us fortunate to be in the thick of it.

Speaking of years, it has been a lot of them since I wrote the Zephyr manual intro, and even more since the Telos 10 – almost 20 years now. Amazing thing is, with all the change around us, I'm still here and Telos is still growing in new ways. As, no doubt, are you and your stations.

Steve Church,
January 2004

A Note From The President of Axia

Nearly 20 years ago, I designed my first broadcast console for PR&E. I look back on that time with great fondness; we were building bullet-proof boards for the world's most prestigious broadcasters, making each new console design bigger and fancier to accommodate a wider variety of source equipment and programming styles. The console was the core of the studio; all other equipment was on the periphery.

Then things changed: the PC found its way into broadcast audio delivery and production. At first, PC audio applications were simple, used only by budget stations to reduce operating expenses. But soon the applications evolved and were embraced by larger stations. Slowly, the PC was taking center stage in the radio studio.

Like many, I was captivated by the PC. Stations retired carts, phonographs, open-reel decks, cassettes — even more modern digital equipment such as DAT and CD players, replacing all with PC apps. Client/server systems emerged and entire facilities began using PCs to provide most — or all — of their recorded audio. Yet consoles continued to treat PCs as nothing more than audio peripherals. I knew that we console designers were going to have to rethink our designs to deal with computer-centric studios.

During this time, traditional broadcast console companies began producing digital versions. But early digital consoles were nearly identical in form and function to their analog predecessors. It took a fresh look from a European company outside broadcasting to merge two products — audio routing switchers and broadcast consoles — into a central processing engine and attached control surface. Eventually nearly every console and routing switcher company followed suit, and a wide variety of digital “engines” and control surfaces flooded the market.

But, advanced as these integrated systems were, they still handled computer-based audio sources like their analog ancestors. Sure, the router and console engine were now integrated, but the most important studio element — the PC — was stuck in the past, interfaced with 100-year-old analog technology. The PC and console

couldn't communicate in a meaningful way — strange, considering that PCs everywhere were being networked, fast becoming the world's most popular and powerful communication tool.

Then a group of Telos engineers developed a method of using Ethernet to interconnect audio devices, allowing computers and consoles, controllers and peripherals to interact smoothly and intelligently. Powerful, flexible networks had finally come to our studios. As with the transition from carts to computers, the benefits are many and impressive. A few networked components can replace routing switchers, consoles, processing peripherals, sound cards, distribution amps, selector switches and myriad related devices.

This deceptively simple networked system costs a fraction of other approaches, yet has capabilities surpassing anything else. The system is modular and can be used to perform discrete functions in a traditional environment. Concurrently, it easily scales to serve both the humblest and the very largest of facilities. Console, router, and computer work in harmony.

So, equipped with this new technology and countless ideas, we launch *Axia*, the newest division of Telos. *Axia* is all about delivering innovative networked audio products to future-minded broadcasters. On behalf of our entire team, I welcome you as a charter client. *Axia* is the culmination of nearly 40 man-years of some of the most ambitious R&D ever applied to the radio industry. And this is only the beginning. We have more products, innovations, and partnerships in the pipeline.

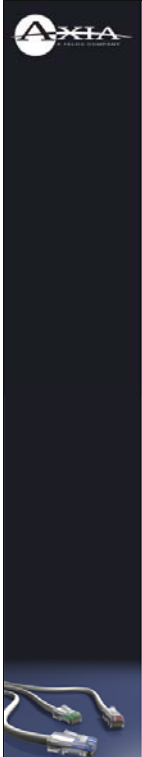
You already know your *Axia* system is unlike anything else. So it shouldn't be surprising that your new system is loaded with new thinking, new approaches, and new ideas in virtually every conceivable area. Some concepts will challenge your traditional ideas of studio audio systems, but we're certain that once you have experienced the pleasures of the networked studio, you'll never want to go back. And now, for something completely different...

Michael “Catfish” Dosch
February 2004





This is a walk-through of the Axia Element modular control surface. Since its introduction at NAB 2005, Element has become one of the fastest-growing console lines in the industry. Congratulations on your purchase!

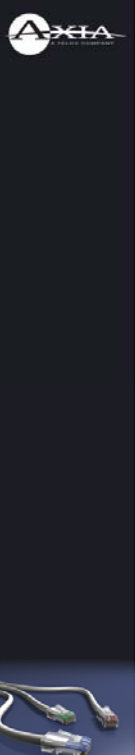


Element is...



Modular.

There are several reasons why Element has become so popular. First, it's modular. You can build Element any size you want, from 4 to 40 faders, so it's always the right size for your needs.

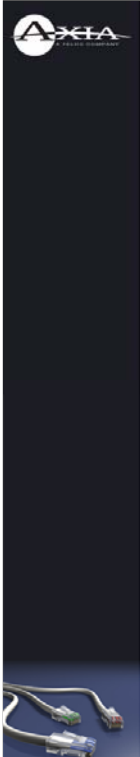


Element is...



Flexible.

Element is flexible, with the ability to set, save and recall individual operators' favorite settings with just one button-touch.



Element is...

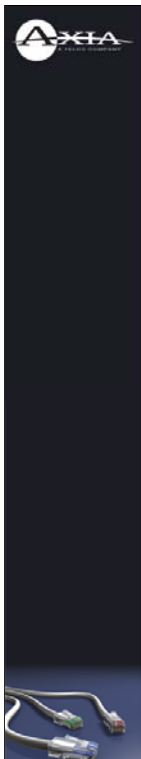


Easy to use.

Element is very powerful, but also extremely intuitive. Board-ops can be trained and on-the-air with very little training.



Element is intelligent. Many operations that used to be complicated and confusing – like setting up mix-minuses or recording off-air bits – are now completely automatic. Board ops can concentrate on doing their show instead of running the board.



Specs at a glance:

- 4 Main Program buses
- 4 Aux Send buses
- 2 Aux Return buses
- Mixes Stereo or Surround
- Automatic mix-minus on every channel
- Selectable source EQ and mic dynamics
- Headphone dynamics
- Sophisticated talkback controls
- Sizes from 4 to 40 faders

But before we look at Element under the hood, let's look at what's on the surface (pardon the pun).

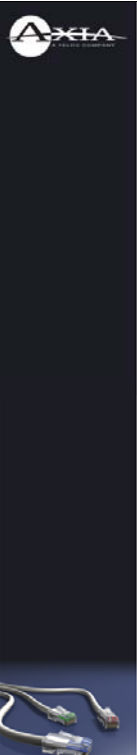
Element has 4 program buses, 4 Aux Send buses, 2 Aux Return buses, and can mix in Surround or Stereo – selectable in software.

Every source can have its own custom mix-minus – one for every fader on the board. Got 16 faders? You can have 16 mix-minuses!

Onboard voice processing from Omnia is standard, along with per-source EQ.

You can even EQ and compress the headphone feed to simulate “air sound” when off-air monitoring is not practical.

Since we made mix-minus easy, we did the same thing for talkback. Just press the button on any fader to talk back to that fader's source. Press more than one to talk back to multiple destinations.



What is IP-Audio Routing?

- It's not Internet audio!
- Uses switched Ethernet
- 48 kHz / 24-bit uncompressed audio
- Delivers real-time audio with guaranteed QoS
- Routes machine logic, PAD, custom backfeeds along with audio source

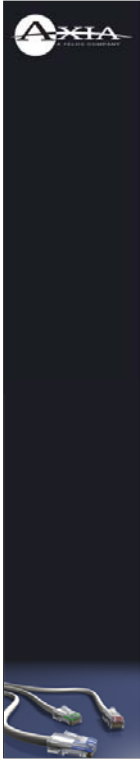
Element works with Axia IP-Audio networks. What is IP-Audio? Well, it's not Internet audio! Internet audio has dropouts, pauses and stuttering because the Internet is an uncontrolled, open network.

An Axia IP-Audio network is a private, controlled, switched Ethernet network where live audio is Priority A-1 – no collisions, no dropped packets, no audio loss.

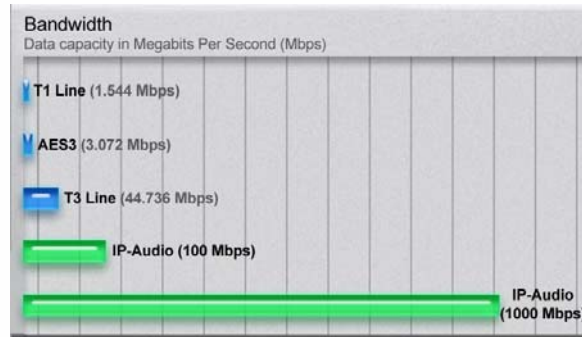
Audio on Axia networks is studio-grade 48 kiloHertz, 24-bit uncompressed audio.

Because they use switched Ethernet, with guaranteed QoS, audio is delivered in real-time with extremely low latency – less than half a millisecond per network hop.

And because IP-Audio uses standard Ethernet for transport, it transports other things along with audio: like machine logic, so that logic always follows audio; program-associated data; mix-minus and custom backfeeds, and more.



Enormous Capacity!



How is it possible to get all of this on one Ethernet connection?

Because of Ethernet's enormous bandwidth. T1 and T3 are pretty speedy, but Gigabit Ethernet has more than 20 times more bandwidth than a T3! With this kind of capacity, there's no chance of running out of headroom.



With Element...

- Any network audio source is available on any fader
- Logic follows audio automatically
- Users can set, save and recall favorite configurations
- Mix-minus is automatic
- You're HD-ready.



Here are some of the things Element can do.

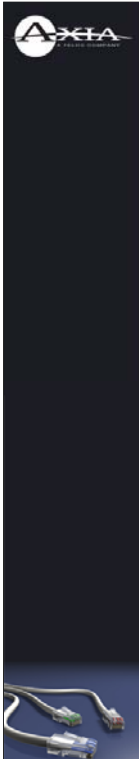
Any source on your network can be assigned to any fader in any studio and it's machine control logic comes along too.

Element has a unique Show Profiles feature that lets talent set up the board any way they like – mics over here, delivery system over there. Users can customize their own board layouts with their own tailored voice processing, monitor preferences and more, and call those settings back up whenever they come in for a shift.

Automatic mix-minuses. When you add a new phone hybrid or codec to the system, you simply tell the software that it needs a mix-minus backfeed, and whenever that source is used, in any studio, a mix-minus is automatically generated.

Element future-proofs your studio. Multiple audio channels? 5.1 Surround? Element can handle it, without any additional cost outlays – the functionality is already built-in.

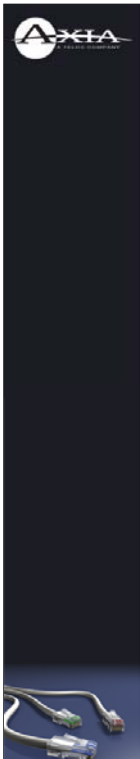
Let's look at the actual Element controls now.



The overview.



Element faders are long-life, conductive plastic. They'll stand up to a whole lot of use. When Board ops touch the Channel Options button at the top of each fader, they can quickly assign new sources, examine source characteristics and adjust backfeed, dynamics and EQ (if available).



Bus assignment keys



These four keys are the assignment keys for the main mixing buses, Program-1 through Program-4. Program-1 is usually the “air feed”. Any fader assigned to Program-1 is going to be on the air.

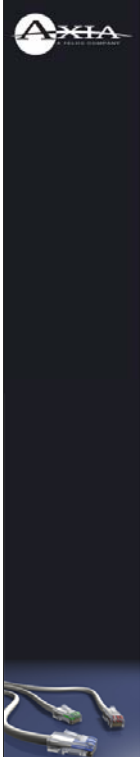
Program-4 is the “record feed”. Assign things to Program-4 when you wish to record them off-the-air for later use.



These 10-character displays tell operators exactly what's assigned to each fader. A Status Symbol icon above the display shows when telephone backfeed or channel talkback is active.



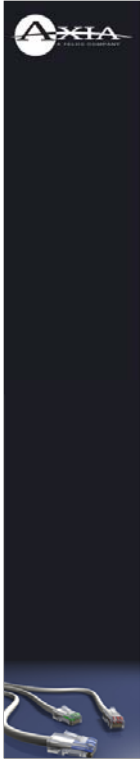
Here you can see how the displays normally look. The top window usually shows the Fader Number (left to right position on the console), and the bottom window shows the source assigned to the fader.



Channel options keys



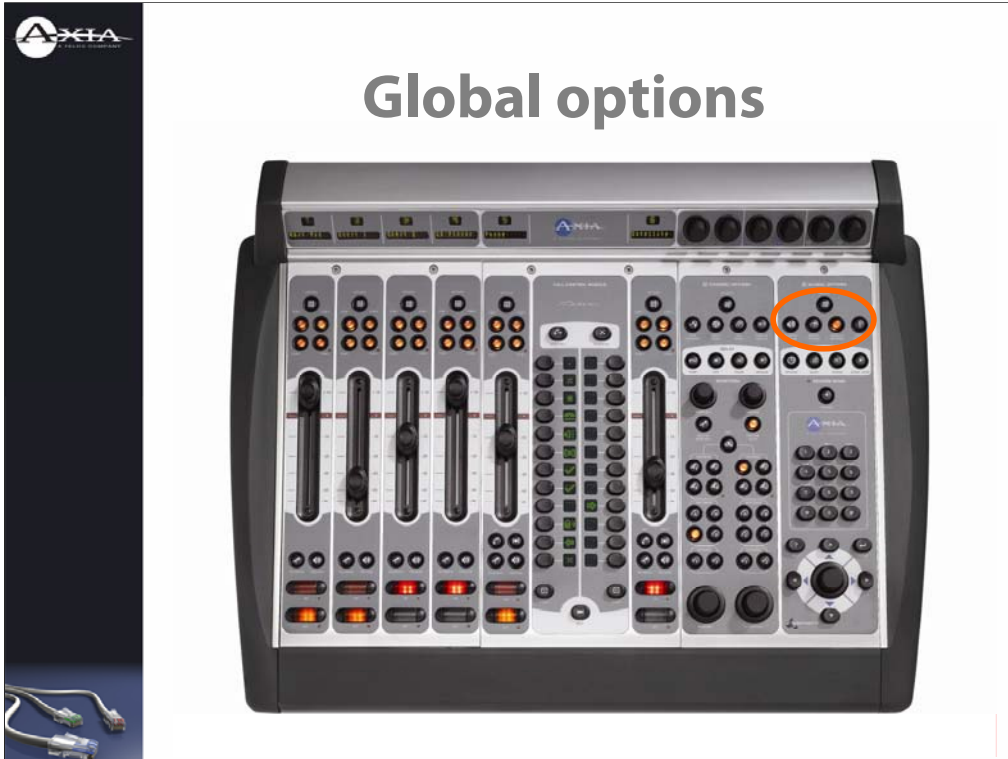
The Channel Options keys give access to source selection, each channel's aux send settings, dynamics and EQ and mix-minus options. A set to control an external profanity delay, if needed, is located here as well.



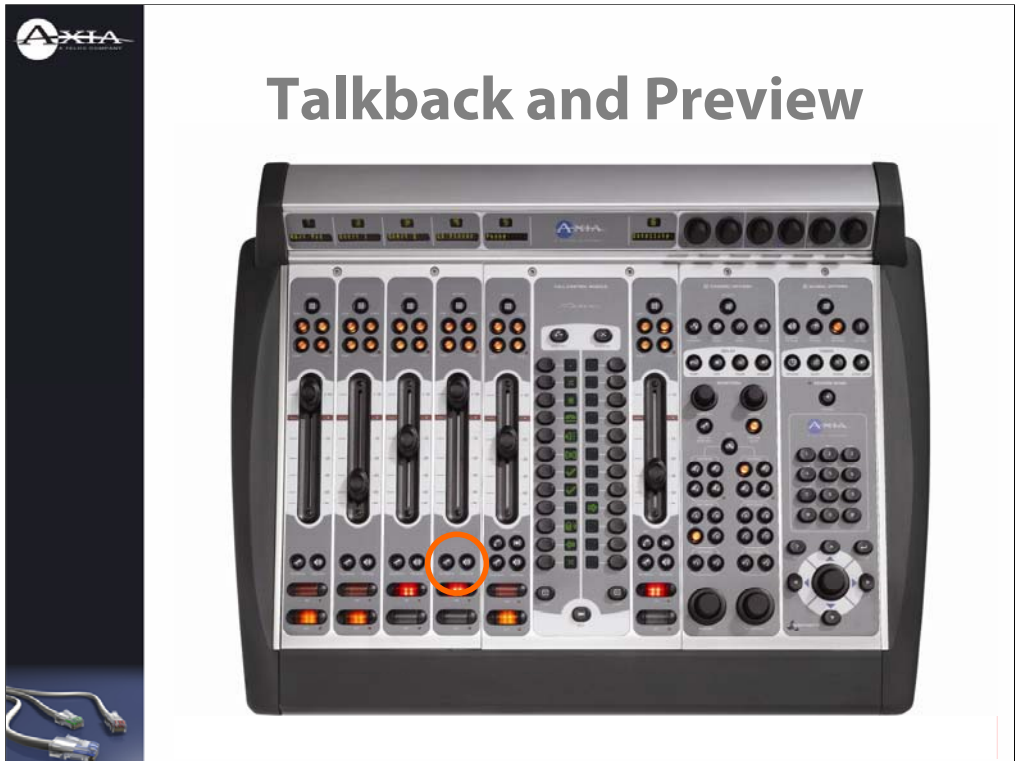
SoftKnobs



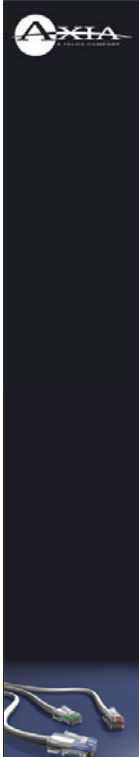
These soft knobs work with Element's on-screen display to help users select and modify custom options.



Global options buttons let you specify settings for event and countdown timers, aux send and return master settings, monitor options and more. The Show Profiles button is here, too – that’s the button users press to load their own custom settings.



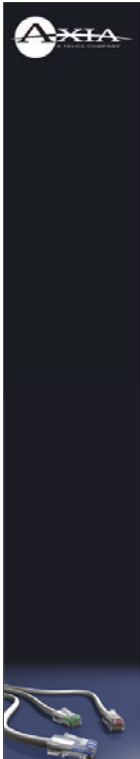
Talkback and Preview (that's Cue) keys are located right next to the on-off buttons. Talkback lets you literally "talk back" to any source that has a backfeed, like a Zephyr remote codec, a phone caller or a studio mic with a dedicated headphone feed. Preview lets you listen to audio from a source off-the-air, in a dedicated "preview" speaker, to see what it is before taking it to air.



Integrated phone controls



The optional Call Controller module works with Telos phone systems and has hybrid controls, two faders and Status Symbols icons that tell operators exactly what each line's status is.



Phone Controller Icons



The Phone Controller can host up to 12 incoming lines on two hybrids (we'll call them "left" and "right"). The icons shown here tell you exactly what each line is doing at any time.

An unused line is shown as a single small dot. This is a line you can dial out on, or which an incoming call may ring to.

If you pick up a line and dial out, the 2nd icon, an "x", will appear on the line in use.

When a caller rings an incoming line, the series of expanding squares shown in the 3rd row above will display on the active line. Press the button adjacent to that line to pick up the call.

In addition to the Call Controller mounted in your Element, there is usually a phone set in your studio. The "phone" handset shown in the 4th row above displays when a call is being taken on the handset.

Likewise, if you're taking a call on the speakerphone, the "speaker" icon will display on the phoneset.

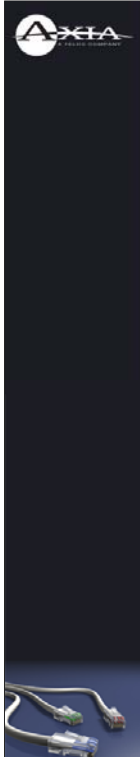
When you're talking to a caller, pressing the "HOLD" button on the hybrid control places the call on Hold. The square with brackets shown on the 6th line displays to let you know the call's on hold. Press the button next to this icon to take the caller off Hold and talk to him again.

If you have a call screener working with you on the air, he or she can pre-screen calls and ready them to go on the air. Calls that are screened and ready for air have the check mark shown in line 7 above display next to them.

Screened calls are placed in a queue. The first screened call in the queue displays the flashing check mark shown in line 8. Pressing the "Next" key at the bottom center of the Call Controller steps through queued callers in order.

When you place a caller on the air by pushing the button next to his line display, the icon shows an arrow pointing left or right, depending upon which hybrid you've chosen to use. Push the button a second time to "lock" the caller on the air – this way, the call cannot be accidentally disconnected. An arrow pointing left or right, accompanied by a "lock" symbol, shows you that the caller is "locked" on the air. To "unlock" the caller, press his button again. The arrow will display by itself, and the call can then be dropped (disconnected).

Pressing the "BUSY" key at the bottom of the Call Controller makes all lines appear "busy" to outside callers. You can use this to prepare for contests or other call-in events. When "busied out", a large "X" will appear next to all lines not currently in use. If a line is being used, it will be "busied" as soon as it is no longer active.



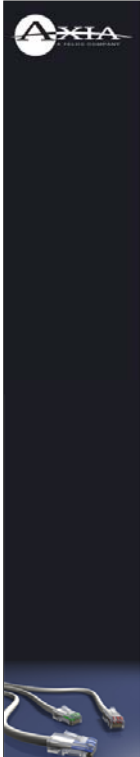
Monitor section



A very flexible monitor section lets board ops specify monitor and headphone sources for themselves and studio guests. You can choose to monitor any of the Program buses or Aux Send buses, or any external source, such as an air monitor or network feed.



Record Mode takes the hassle out of recording off-air bits or interviews. One touch of the Record Mode button starts a pre-determined record device and begins recording all audio assigned to Program-4. This is useful for quickly recording phone calls or remote feeds to be used later.



Numeric keypad and nav keys



The built-in keypad can be used to dial phone calls right from the board. To do this, just push any hybrid key on the Call Controller next to an unused line, and dial.

The navigation (arrow) keys and scroll wheel assist in navigating through on-screen options. They can be used in conjunction with the Soft Knobs at the top of the Monitor Section to scroll through and select options on-screen.



This is the main display screen. Operators will see this screen 99% of the time. Let's look at some of the onscreen elements.

The orange box highlights the program meters. They have visual peak indicators to help keep you from overmodulating. If the "PEAK" indicator is lit, decrease the volume of your on-air source(s).

The far right meter can be set to display the volume of sources in Preview, or just display Program-4.



Day and date displays are on the top along with a digital time of day readout.



The big analog clock in the middle shows the time of day, and also has a visual indicator – the red ring – that’s displays during the last minute of countdown timer activity, so you can see at a glance how long you have left before the timer reaches zero.



The operator can set the countdown timer for any interval. Useful for timing network joins or breakaways.



There's also an event timer that can be triggered manually or automatically when an audio source starts.



This multi-function pod lets you know when your board mic is on so you don't accidentally talk on the air without knowing it. Other tallies illuminate when talkback is active, when a phone hybrid is in use, when a source is in preview.



Monitor section shows what sources are being auditioned on main monitors, operator's headphones, in the adjacent studio (if you have control of one), and in the Preview speakers.



The Profile pod shows what Show Profile is currently loaded, and whether changes have been made to that profile's initial settings. If a user gets into trouble – assigned something somewhere they didn't intend, perhaps – pressing and holding the show profile key resets the loaded profile to default settings.



The display shows other important data, too. When a channel's OPTIONS key is pressed, the center of the display shows information about that fader's source: the name of the source, what mix-minus it's receiving (if any), its EQ settings (if active), aux send info, and source-specific information, such as what transmission mode a Zephyr codec is using.



In Surround mode, board ops see meters with 5.1 information and accompanying stereo levels. There's also a Surround panner display that's useful for production.



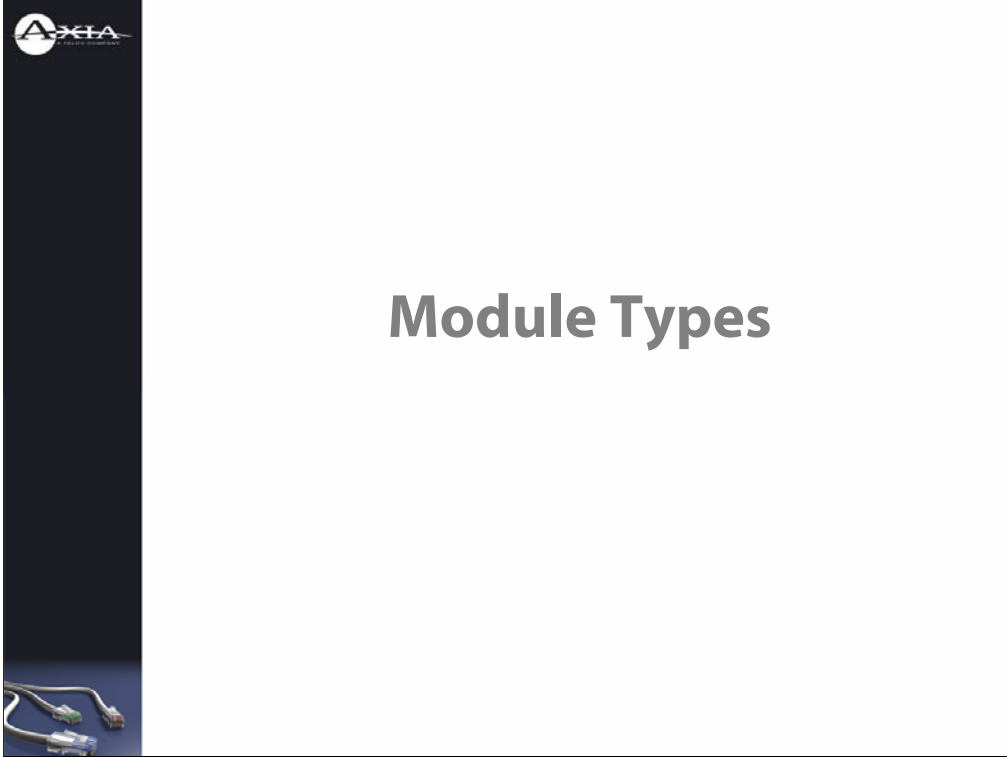
When you're adjusting EQ, dynamics or other options, the center screen gives a graphical representation of your adjustments. Notice the on-screen legend for the Soft Knobs that indicates their functions.



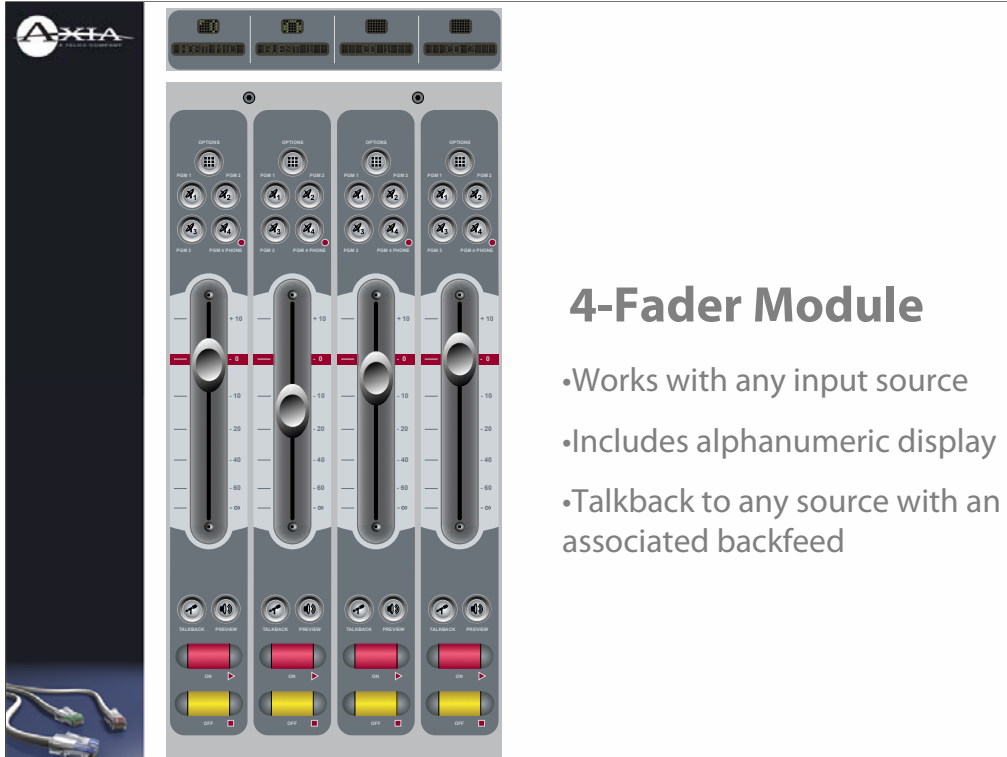
The screen is context sensitive and changes with each adjustment mode. This is the Voice Processing screen. (By the way, these features can only be adjusted if your engineer allows them to be – otherwise, they are inactive.)



Loading a new show profile is easy – users press one button, then choose from an onscreen list. When you choose one, changes are immediate, but they don't disrupt on-air operations. For instance, if a newly-loaded Show Profile assigns a new source to Fader Two, and Fader Two is on-air at the time, the new source won't "take" until Fader Two has first been turned off.



There are many different Element modules that your console may be equipped with.



4-Fader Module

- Works with any input source
- Includes alphanumeric display
- Talkback to any source with an associated backfeed

Most Element modules occupy 4 fader spaces. The 4-Fader module is the cornerstone of the board. Axia networks let you load any source to any fader, so the 4-Fader module will work with any type of input source - mic, line, codec, hybrid, or computer. An overbridge panel provides a 10-character alpha display, and each fader is equipped with a Talkback key to let your operator speak to sources with backfeeds - like studio talent with individual headphone feeds, remote talent with IFB, or phone callers.

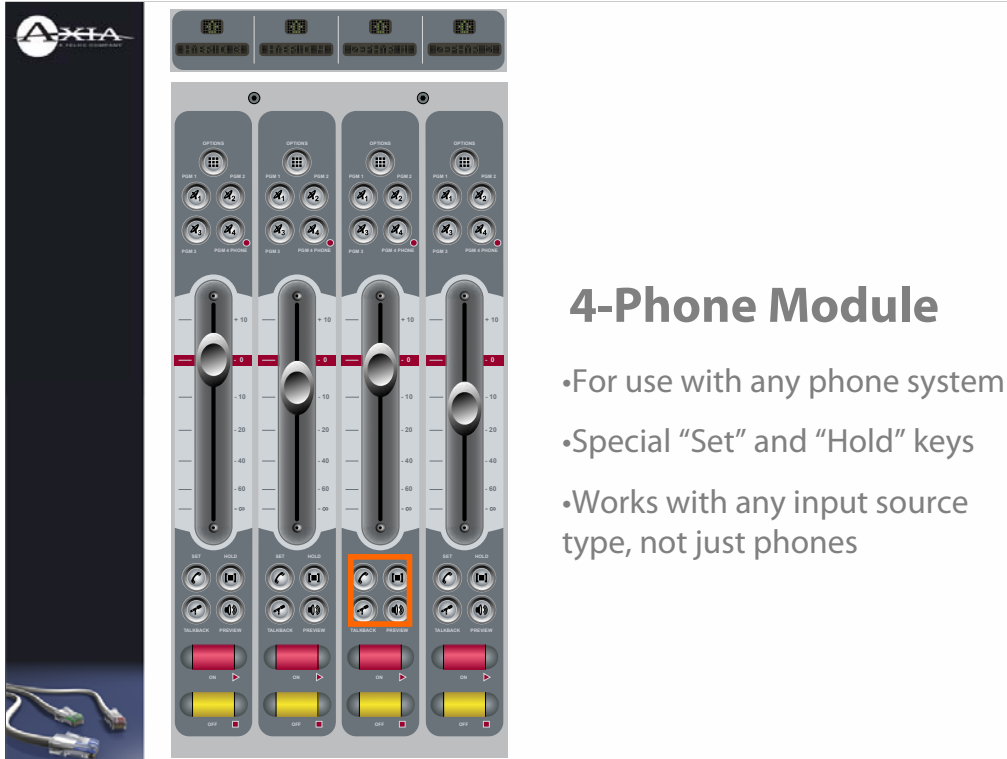
Push the "Options" key at the top of a fader to activate the Options screen. Then use the "Source" key on the Navigation Module to choose a new source for that fader.



Call Controller

- For use with Telos smart phone systems
- Dual hybrid controls with Status Symbols™
- Special function keys for hold, next-in-queue, drop, busy all

The Call Controller module works with Telos “smart” phone systems, like the Series 2101, TWOx12 and the new NX12. Status Symbol icons, discussed earlier, give detailed line information.



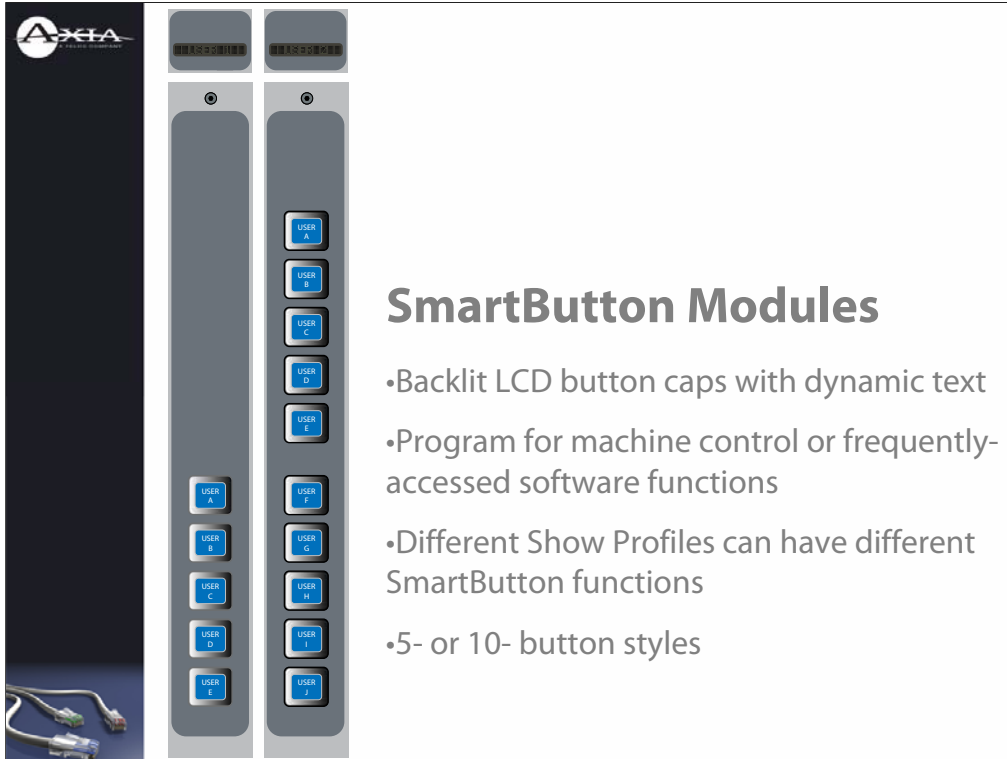
4-Phone Module

- For use with any phone system
- Special “Set” and “Hold” keys
- Works with any input source type, not just phones

The 4-Phone module can be used with ANY source, but has two extra keys for use specifically with phone systems. The “hold” key puts a hybrid’s caller on hold, and the “Set” key transfers the call from the console to the hybrid’s handset for off-air conversation.



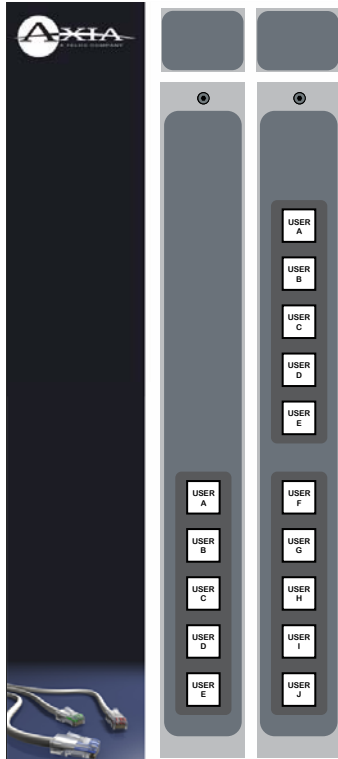
The Production Module gives production room users fast access to source EQ settings, aux send and return gain, and contains pan controls for both stereo and Surround mixing. If your console is equipped with a Production Module, activating the “Options” key at the top of any fader activates the Production Module (its lights will illuminate). You can then make changes that will affect only the sound of the chosen source.



SmartButton Modules

- Backlit LCD button caps with dynamic text
- Program for machine control or frequently-accessed software functions
- Different Show Profiles can have different SmartButton functions
- 5- or 10- button styles

SmartButton modules come in 5 or 10-button versions, with backlit LCD keycaps. Your engineers program their functions, giving you access to frequently-used recording controls or other software functions, or for one-button activation of complex switching operations like control-room transfers. The buttons' functions can be specifically programmed for each Show Profile.



Film-Cap Modules

- Backlit film-legentable button caps
- Program for machine control or frequently-accessed software functions
- User-definable backlight colors
- 5- or 10- button styles

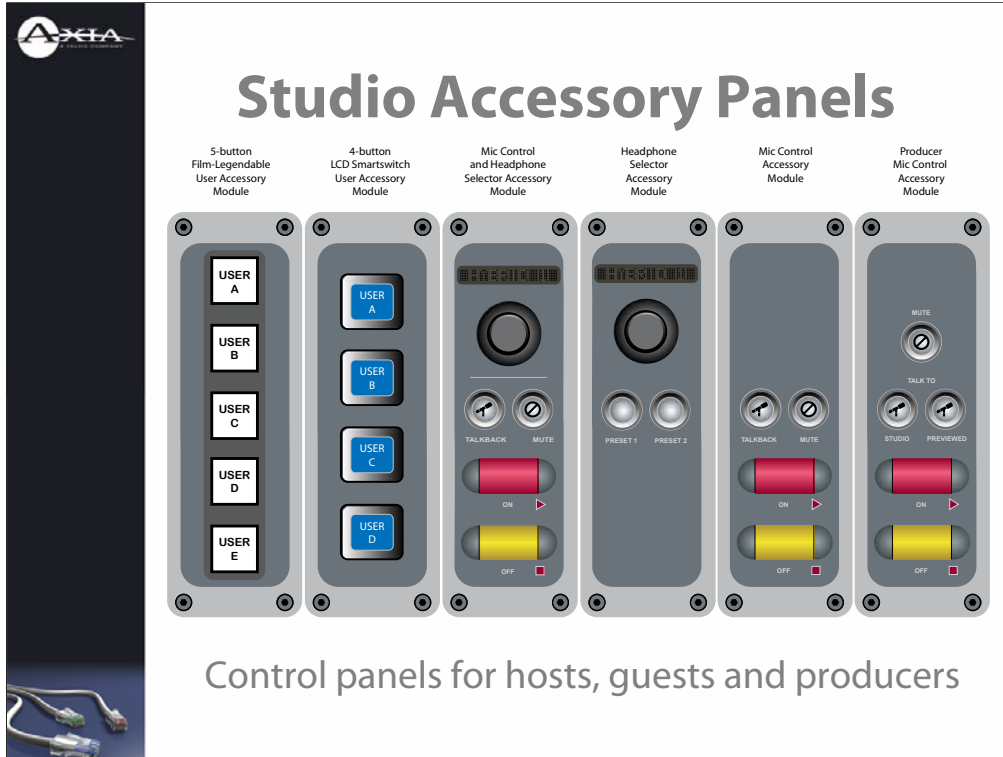
These film-legentable button modules work much like the SmartButton modules, except that their functions do not change when profiles are loaded. Their functions are always the same, all the time.



Nav/Monitor Module

- Master control module
- Comprehensive, flexible Monitor section
- Countdown and Event timer controls
- Numeric keypad and navigation keys + scroll wheel

Finally, the Navigation and Monitor Control Module contains all the controls for monitor and headphone sources, keys to control an external delay unit, an integrated dialing and numeric entry keypad, and a set of six soft knobs for adjusting onscreen options.




A range of accessory panels may be installed in your studios to give hosts, guests and producers control of headphone and microphone options. There are five-button Film Legendable button panels and 4-button LCD SmartButton modules, a mic control/headphone selector panel with an alpha readout and integrated talkback and mute keys, standalone headphone and mic control panels, and an Producer’s Mic Control panel with talkback keys for studio guests or any previewed source.

What's Next

Now that you're familiar with the controls of your Element control surface and where they're located, spend some time exploring them — touch everything! You can't hurt your Element by exploring.

(If you want to reset your Element to its default configuration, just press the “Profile” key and choose “Default Show Profile.” It'll return your console to normal.)

In the next chapter, we'll examine the basic theory of broadcast console operation. 

Chapter Two:

Mixing & Monitoring Basics

Introduction

To the uninitiated, an audio control board looks pretty terrifying with all the knobs, switches, faders, and meters packed into a compact space. Although a control board may appear intimidating at first, it is a lot easier to learn and operate than you think. Generally, a station staff member trains new operators and takes them through the station's control board operation. With some hands-on experience you will get over your fear quickly. As one of my mentors in broadcasting said as I stood looking at my first 24-channel broadcast console, "This isn't rocket science; it's radio!" Still, operating a console requires hand-eye coordination, pre-planning, and some creativity and skill.

The reality is, running an on-air control board is the simplest task a show host has. The difficult work is planning your show, following the station format, and creating a fresh, innovative, relationship with listeners. The same applies to production. The operation of the hardware and software at your disposal is relatively easy. The tough part is creating the script, selecting music and sound effects, and learning how to communicate with your audience. Remember, despite all of the hardware you have to work with, the primary goal is to communicate messages to listeners. The control board is simply one tool used in the creative process.

Broken down into their modular components, audio control consoles are easily understood since they all do basically the same thing. In fact, once the basic theory of operation is understood, you will have the skills necessary to sit behind just about any control board and figure it out operationally.

Operational Theory

Understanding the Different Sections of a Console

An audio control surface performs three basic functions—amplification, routing, and mixing. Audio first enters the input section of the control board from microphones, recorders, or other sources, at either a mic or line level. The first stage of the control board amplifies the various audio inputs to a standardized level for use within the control board.

Using routing, or assignment switches, the operator determines the audio's path through the control board. The audio is eventually routed to the output section, where it is sent to recorders, audio processors, or broadcast transmitters. At the same time, the audio is routed to the monitor section of the control board so that operators and talent in the control room or studio can hear the project on monitor speakers or headphones.

Mixing is the process of using the various volume controls to balance or blend the audio from the numerous sources and achieve the sound the production person or producer desires. Mixing occurs at several stages in the control board and is finalized in the master section with the master output volume controls. Mixing also involves adjusting playback levels in the monitor section that feeds the monitor speakers, headphones, and intercoms, in both the control room and the studio.

Breaking down a console's functions into amplification, routing, and mixing makes it much easier to understand. These three functions occur within the input, master, output, and monitoring sections of the console.

Types of Audio Control Consoles

There are five popular types of audio control boards or consoles. On-air and production consoles are two different types found in radio stations. Virtual consoles exist within digital audio workstations, with some being used in on-air applications. Portable mixers are used for live remote radio broadcasts and sporting-event coverage. Although less common in radio production, large-format mixing desks are found in recording studios and in pro-

duction houses that specialize in custom audio production for the media. Regardless of the console type, they all operate on the same universal principles of amplifying, routing, and mixing across the input, master, output, and monitoring sections of the console.

On-Air and Production Consoles

Broadcast audio control consoles for radio come in two variations that are as different as night and day—on-air consoles and production consoles.

On-Air Console Overview

On-air consoles are designed for controlling a number of audio sources, including microphones, CD players, digital audio storage systems for commercials and music, radio networks, and so forth.

An on-air console's only real role is to route and control the volume levels of all the sources, eventually sending them to the broadcast transmitter. The key design element of an on-air console is simplicity and ease of operation under pressure. In effect, the console designer tries to prevent operator errors with the layout and design of the console. The true role of an on-air console is not creativity but control.

Production Console Overview

A production console is just the opposite of the on-air console; it is designed for creativity. Production consoles have many more features and signal-routing abilities. These include individual channel equalization, the incorporation of outboard gear such as echo and reverbs through auxiliary audio sends and returns, and the ability to mix audio in three dimensions including width, depth, and height. Naturally, to be able to perform all of these functions a production console has more controls and metering.

The Impact of Digital Consoles

Both on-air and production consoles can be either analog or digital, the difference being that in a true digital console all inputs are from digital sources, the audio remains in the digital format in the console, and the console's digital outputs are fed to digital recorders, audio processors, or transmission equipment.

Digital consoles are available in two different physical styles. The first is a console that appears to be a traditional analog console with the same simple, familiar knobs and controls. Each knob and fader has an assigned function, or knob per function. The only difference is that the console's internal components and the audio moving through the console are digital. This type of digital console requires no retraining on the part of the operator. In fact, the operator might not even know he or she is using a digital console. Typically, this is the type of digital console found as an on-air console and is simple and easy to operate.

The second type of digital console is the “any-switch-can-be-anything-any-day” console. The heart of the console is an on-board microprocessor (computer) used to set up the console and assign the input and output functions to each of the console's switches and faders. The channel assignments, switching setup, and functions can be changed at any time, using the console's on-board microprocessor. Literally, any input, output, or function can be assigned to any channel or control; it's like moving checkers on a checkerboard.

Each operator sets up, or programs, the console the way he or she prefers, rearranging console configuration to suit the operator's particular needs. The operator saves the configuration in the console's memory under his or her name or on a removable flash memory card. At any time the operator can recall the settings and reset the control board to that operator's original preferences. This type of digital console requires much more attention to detail and is usually found in production studios.

A digital console allows each operator to set up, or program, the console the way he or she prefers, rearranging the console configuration to suit the operator's particular needs. The operator saves the configuration in the console's memory. At any time the operator can recall the settings and reset the console to his or her original preferences.

On-Air Consoles in Depth

Most radio stations have already installed, or are in the process of installing, upgraded audio consoles. The new consoles are needed to meet competitive and consumer demands for high-quality audio and digital broadcast transmission requirements. On-air consoles are generally custom-ordered to meet the station's specific control room needs. This is easy to do since most consoles are made up of a number of different modules and individual channel strips and can be ordered in just about any configuration a station engineer desires. Prices for on-air consoles, such as the example in this chapter, start in the \$10,000 range and rise quickly, depending on the features that are ordered. Most radio stations operate in what is termed the combo mode, meaning that the talent serve as their own engineer, running the control board and other elements of their show.

The easiest way to understand a console is to break it down into components, looking at just one channel strip at a time, top to bottom. The Element on-air console has the appearance of an analog console. It has identical channel strips; if you know how to operate one, you know how to operate all twelve, and if you know how to operate twelve, you can operate twenty-four.

Selecting the Channel Input

At the top of the first channel strip, located on the left side of the console, is the Options key. Among other functions, this switch lets you choose the input feeding this particular channel. For clarity and ease of operation, an alphanumeric display above the fader indicates the actual radio station source, such as a CD player.

After selecting the Options key, choose the Source key located at the top left of the Monitor/Navigation module. Use the knobs above the Monitor section to scroll through the list of available sources, and choose "Take" to select a highlighted source. Press the Options key above the fader twice to exit the Source Selection screen and return to the monitor screen.

After choosing an input to the channel, the next step is to use the output section of the channel strip and as-

sign where you want the channel's audio to go after the level has been adjusted with the fader.

The Output or Assignment Section

Traditionally, on-air consoles have two main buses (common outputs) to which audio is assigned or routed, program and audition. Element has four main buses, giving you extra room for "utility" mixing.

Element's Program-1 bus generally feeds audio to the radio station transmitter. Any channel with this button lit is routed to the transmitter. The other buses are entirely separate circuits and are used to preview audio on the studio monitor speakers without putting the audio on the air. These buses can also feed recorders, so that while the program bus is feeding the transmitter, other buses can be used to record a telephone call or other program elements.

Element has three additional buses that audio can be assigned to, Program-2, Program-3, and Program-4. The outputs of these buses can be sent anywhere the operator needs the additional audio feeds. For example, one bus might feed a station's IFB (interruptible foldback system) so that the on-air talent can talk to, or feed, an on-air signal to the remote talent's headphones during a live remote or sports broadcast. With Element, it is possible to route a signal from one fader channel to four separate places simultaneously, if necessary.

Listening In Preview

Continuing down the channel strip, you'll find a Preview button. Preview (traditionally called Cue) is an isolated bus routed to a separate amplifier and speaker so that you can listen to a channel to preview music or check microphones before you put them on the air. When a channel is placed in Preview, its signal cannot go over the air; Preview is a part of the monitoring section of the console. Most preview speakers are small, crummy-sounding speakers placed in an odd location in the control room, such as behind the operator or under the desk the console is sitting on. If an operator makes a mistake and inadvertently leaves a channel in cue that is supposed to go on the air, the odd sound from the tiny, crummy cue speaker alerts him or her to the error.

Faders and Level Control

Often, instead of faders or level controls, the term “pot” is used. Pot is the abbreviation for the word potentiometer, the technical term for a volume control. Other terms for the volume control include attenuator, gain control, or just plain volume control.

There are basically two types of pots, rotary and faders. Rotary pots are round volume knobs that are turned in a clockwise/ counterclockwise manner to raise and lower the volume. Faders are volume controls that require more space to operate, sliding a knob along a short linear scale. They are sometimes referred to as linear faders. These are the kind your Element is equipped with.

The fader slides along a linear scale to raise and lower the channel’s audio volume. At about three quarters up the scale there is a simple line; this mark is the unity gain position. Unity gain is an optimum audio level near 100 percent modulation and is preset by the station’s engineer. This makes it very easy to run audio levels on the console.

Below the fader are the channel’s on/off switches. The switches can also be used to trigger remotely start/ on and stop/off devices such as a CD or minidisc player or other digital playback source.

Monitoring Your Work

The monitor section controls what you hear in the control room monitor speakers and headphones. Monitoring levels should be set to a comfortable level in the control room. Keep in mind that your job is not to stand and listen to music, your job is to produce a radio show. Most beginning radio operators are surprised to find out how little music they listen to or even remember hearing during their show. A simple rule of thumb is to listen at an “office” level. You want to hear what is on the air, but you do not want it to dominate your control room since you are going to be doing other things in the control room while the music is playing, such as taking phone calls and talking to other staff coming and going from the control room.

With Element, there are multiple monitor inputs to se-

lect from on the monitor channel section located on the Monitor/Navigation module. In addition to the program buses, there selectors for Aux buses and two External monitor sources. The External sources are for off-the-air monitors, referred to as an off-air signal. Any one of these sources can be listened to in the control room monitor speakers or headphones without affecting what is going on the air. Simply select the source and turn up the monitor speaker or headphone volume.

On-air talent use headphones to monitor with when they are talking on the air. The reason for this is that the moment the on-air microphone is turned on, the monitor speakers were muted (turned off) by the control board to prevent the feedback (the deafening howling sound that public-address systems sometimes produce) that would occur if the sound coming out of the monitor speakers were picked up by the control room microphone, fed back through the control board and broadcast transmitter, and back out of the monitor speakers into the microphone again in a vicious, screeching, never-ending circle.

Automatic muting of the monitoring speakers is a universal feature of broadcast control boards. Production control boards do not always have the feature and the operator may have to turn the monitor speakers down manually before using the studio microphone.

It is critical that you be able to hear your work accurately. The monitor section of a control board provides you the controls to do just that. The one control that can get you into the most trouble is the monitor volume control for either the headphones or monitor speakers.

A common beginner’s error is listening to the Program-1 bus (what is coming out of the control board going to the transmitter) and not the actual air signal after it has been broadcast. When working in an on-air control room, your default monitor is always the actual air signal so that you are the first to know if there is a signal-quality problem or the station goes off the air. Many an operator has been embarrassed beyond belief when a station manager or chief engineer frantically entered the control room and the operator did not know the station was off the air. A station can go off the air for any number of rea-

sons, including power outages, storms, lightning strikes, and so on. The point is, in an on-air control room, you should always listen to the off-the-air monitor.

Many beginners like to work with the monitors “cranked” (nearly loud enough to tear your clothes off). Big mistake. Learn to monitor your work at lower levels. Your listeners are not listening on studio monitors and certainly not at the levels you can listen at. And although it may seem easier to hear everything in the mix at louder levels, a good rule of thumb is to mix at a low level to make sure that everything in the mix can be heard clearly at that level. Most importantly, the voice needs to stand out above the mix.

If your Element is equipped with a Call Controller module, you can control audio levels from a telephone hybrid (a hybrid is an interface between the telephone system and your console) and monitor multiple telephone callers at once, ideal for talk shows or taking requests.

On-Air Operational Tips

Control consoles are pretty straightforward with regard to their functionality; they are all basically the same. The key to being able to handle any console is not to learn just the knobs and faders of one console but to understand the overall operational principles that have been discussed here, which you can then apply to all consoles. As an operator you are going to have to learn to “walk and chew gum at the same time” so that you can talk on the air and operate the console without really thinking about it. Given a little experience, operating a console is like riding a bicycle; they all have pedals and handlebars. The trick to smooth operation often lies in simple things that can make the job of the operator easier.

The first rule of working in a studio is never, ever, to take drinks, food, or lighted cigarettes into a control room. Sugar-saturated liquids and electricity do not mix. A soda spilled into a control board can cause thousands of dollars worth of damage and take the station off the air, costing thousands more in lost advertising revenue. Food is a real problem as crumbs find their way into equipment and greasy, sticky fingers leave fingerprints

on control boards, computer keyboards, and computer touch screens. Cigarette smoke leaves a film on switch contacts and sensitive digital equipment, eventually causing equipment failure.

To help you work smoothly and more naturally, Element allows you to configure the layout of your console and save your layout as a Profile, which can then be recalled anytime you use that console. Talk to your station’s operations director or engineer about saving your personal profile.

From an operational standpoint, similar console inputs should be grouped together and logically arranged to assist you. We read left to right, so it is not unusual for a console to be laid out from left to right. For example, what is the most important input on the console? Your microphone is usually on pot 1. That is followed by any other microphones in the studio or adjoining studios. Since you can assign any source you need to any fader, it is up to you as to how the control board is laid out.

The simplest consoles to operate are those that have been laid out ergonomically, or in relation to the studio and operator. For example, if the CD decks are located on the left side of the console, locate your CD deck pots on the left side of the console, too. If the screen and computer keyboard for the music and commercial hard drives are located to the right, locate the pots for these to the right of the console as well.

As a side note, never mark or doodle on a console. Consoles are expensive. If you want to see an engineer get ugly, write on his or her console.

With regard to accepted operating procedures, follow some simple suggestions to avoid problems. Leave all of the pots you are not using closed, or down. This eliminates the chance of something you don’t want to go on the air making it there, and it also reduces any noise that may be present in the source. When opening a mic, turn the mic on and then raise the level on the pot. Likewise, when closing a mic, lower the level and then turn the pot off. This avoids the noise of the switch getting on air and makes a smoother transition from the music to the

acoustic environment of the studio.

Advanced Controls

Element has many features designed to tailor the sound of audio sources. These tools may be enabled or disabled for you, depending upon your engineer's desires.

Pan Control

The pan (panoramic) control permits the sonic positioning of a channel between the left and right stereo channels and works in conjunction with the Options key at the top of each fader channel (select the Options key, then the Pan & Mode key at the top left of the Monitor/Navigation module).

For example, if a fader is assigned to send audio to Program-1, then turning the pan control to the far left or right places the audio on the left or right output channel. Leaving the pan control in the center positions the sound equally between the left and right stereo channels. Normally you will not need to adjust this control for on-air work; but if an incoming audio source is horribly out of balance, you can use this tool to center the unbalanced audio between the left and right speakers.

Equalization

Choose a fader's Options key, then the EQ & Dynamics key located in the Monitor/Navigation module to enter the EQ setup screen.

Equalization is the electronic process of altering a signal's frequency response by boosting or cutting certain frequencies (sophisticated bass and treble controls). Element features equalization that is divided into three frequency bands: low, mid, and high.

Briefly, EQ is used to sweeten or correct minor problems with an audio source's sound. Properly used, EQ can be a powerful tool; improperly used, EQ can easily ruin the sound of the audio. Unless you have had specific training in this area, you would be wise to leave these controls bypassed (inactive).

Auxiliary Sends and Returns: Connecting Outboard Gear

Pressing a fader's Options key and then the Aux Sends key located on the Monitor/Navigation module allows you access to the Aux Send assignments for that fader.

An auxiliary send is an additional output from the individual channel used to send the channel's audio to an outboard signal processor or recorder, or to be used for a separate mix. An aux send taps into the audio on the channel and feeds the like-numbered auxiliary bus output. Each Element channel has four aux send buses and is capable of sending the channel's audio to four different locations outside the mixer.

Naturally, if you use the aux send to route audio to an outboard signal processor, such as a reverb, you need a way to get the processed signal, with the reverb, back to include in the mix. This is accomplished using an aux return input to the console. Controls for the Aux Return bus levels are accessed by pressing the Sends & Returns key in the Global Options section of the Monitor/Navigation module.

An unaltered audio signal being sent to an outboard processor is called a dry signal, and the returned signal, with the reverb or effect added, is called a wet signal.

There is one other feature of the aux sends that adds to their functionality. Aux sends have the option of tapping into the channel's audio before or after the fader. In the pre mode, an aux send taps into the audio before the fader, EQ, pan, and assignment switches. This provides a signal straight out of the preamp, which the console operation cannot affect in any way. A "post-aux" send taps into the audio after the fader and before the pan and assignment switches. This provides a signal over which the fader has level control. The pre/post control is located in each fader's Aux Sends setup screen.

Metering: Being Able to See Audio

Why You Must Watch the Meter

One of the biggest mistakes new operators make is to rely on their ears to tell how loud or soft a signal is. Just as pilots are taught to rely on their instruments, you have to learn to rely on your audio meters to verify actual audio levels. Otherwise, you will crash and burn like a pilot in a storm. Production studio monitors do not, cannot, and will not allow you to judge audio levels accurately, since the speaker's volume can be adjusted up and down to compensate for low- or high-level audio signals.

Many a new operator, with speakers blaring, has said "it sounds fine to me," only to be asked by the program director to look at the control board meters, where the audio level is barely registering. Proper audio levels are adjusted by FIRST observing the meters and setting the audio level, and then adjusting the monitor speakers to a comfortable listening level. Always refer to the meters to observe or set levels. Failure to rely on audio meters has ruined many a program.

Audio Meters

Element meters work on the peak meter principle. Traditional needle-style VU meters were designed for old-fashioned analog consoles and display only average levels. Peak meters display the signal peaks, or the absolute maximum signal level present. This is the kind of meter best suited for digital mixing, because by indicating the peak signal levels it allows you to avoid digital overloads, or splatter.

Element's meter scale is designed to help you keep audio levels at their optimum. The scale is divided into three parts. The lowest part, from -50 to -20 dB, is colored yellow. The area from -20 to -10 dB is colored orange, and the area from -10 to 0 dB is red. The red area of the meter indicates levels that are likely to cause distortion and clipping. In a digital console like Element, audio levels beyond the 0 dB mark will cause the audio to become extremely distorted and unlistenable.

As a general rule, your audio should peak mostly in

the orange part of the meters. To combat transients and instantaneous audio peaks that can surprise anyone, manufacturers build headroom into the console. Headroom is the difference between the normal operating levels and the point when distortion (clipping) starts to occur in the console. Because of this safety net, it is OK to allow occasional peaks into the red area of the meters, since there is some headroom before actual distortion occurs.

Audio that is driven so hard that causes the PEAK indicator to light consists of nothing but distortion and hard clipping. If these indicators come on, reduce audio volume immediately to keep from distorting your programming!

What's Next

Ready to dig deeper? Join us in Chapter Three for a comprehensive, detailed walk-through of Element operations. [🔗](#)

Chapter Three:

Element In-Depth

Once it's installed in your studio, Element looks a lot like a traditional radio broadcast console — but that's where the resemblance ends.

Element isn't a console; it doesn't actually mix or process audio. It would be more accurate to say that Element is a control center, allowing the operator to take charge of the delivery system, the StudioEngine mixing/routing engine, the phone system, the recorder/editor and other sources.

With Element, our goal is to provide you with the most efficient man-machine interface possible for fast-paced, complex programs where board operators must multi-task without error. As such, Element brings a whole new level of control and sophistication to the broadcast studio, while providing very intuitive operation.

Overview

Element was designed to perform equally well in air or production studios. In the air studio, the clean interface enhances speed and accuracy without clutter or confusing controls. In the production room, deeper levels of sophistication are accessible with the touch of a button.

Because different people work differently, we've made it possible to access many functions in multiple ways. By providing several paths to access a function, the board operator has less to remember and is less likely to get trapped. This also serves to bring new operators up to speed easily.

In this chapter we'll first give you a high-level overview of Element capabilities to help convey exactly what Element is capable of. Then, we'll give you detailed explanations of the various controls and functions.

Displays

Unlike other control surfaces, Element does not include built-in meters or other displays. Instead, Element GPIO/Power Supply Nodes provide a VGA output to feed your choice of display monitors.

Element's display normally shows meter levels for all PGM buses, a time of day clock, countdown and time-remaining timers, and well as important status messages. When the operator adjusts console options using the Direct Access Keys on the Master Module, the screen helps navigate the options and, in the case of audio dynamics and EQ adjustment, shows graphic representations of the effects of changes.

Show Profiles

Element can be completely reconfigured, instantly, to suit different types of shows. By using pre-defined Show Profiles, talent can change board settings from a phone and microphone intensive morning show to a personality-based music show — or any other kind — at the touch of a button.

Show Profiles are easily accessed by pressing the **Profile** key located at the top of the **Global Options** section of the Master Module. Element's on-screen display will change to display a list giving instant access to all stored Profiles. Construction and administration of Show Profiles is covered in Chapter Five. "Show Profiles."

Sources and Channels

During the course of this manual, we'll refer often to **sources** and **channels**. *These are not the same!* "Sources" are microphones, CD players, outputs from the playout system, telephone hybrids, etc. Your Livewire network may have a very large number of sources, in different locations, used at various times.

"Channel" usually refers to the individual Element faders and their on/off controls, alpha displays, bus assignment keys, etc. "Channel" is also used when discussing numbered Livewire network streams. To avoid confusion, we'll call them "Surface Channels" and "Network Channels" where appropriate.

Sources are assigned to surface channels for use on-air or in production. During initial configuration, the sources are programmed by the engineer for appropriate logic and options (as outlined in previous chapters); afterwards, whenever that source is assigned to a surface channel, source logic follows.

Surface channel settings are instantly reconfigured whenever a source is selected, to accommodate the unique requirements of the source. For example, a surface channel controlling a microphone source also controls appropriate monitor mutes. A surface channel controlling a line source sends “start” and “stop” commands when the channel is turned on and off.

Sources can also be fed directly to the monitors for auditioning without being assigned to a surface channel.

Mix-Minus

Mix-minus setup, especially for live broadcasts, has always been one of the most confusing aspects of running a radio board.

Element makes mix-minus easy by automating it. The operator never has to worry about sending a source back to itself — it just can’t happen!

Several mix-minus choices – fixed and switching – are possible, and are configured for each source when Source Profiles are defined by the engineer (see Chapter Two for details). Once this setup is done, no further tweaking is needed; the operator simply uses the source. A Status Symbol display on each surface channel tells operators when a particular source has a mix minus output, and even tells them which audio mix is being backfed.

Element will support as many outputs as there are installed faders, each with its own automated mix minus feed. Got 24 faders? You can have 24 mix-minuses!

GPIO

As explained in Chapter Three, each input source can have associated GPIO (General Purpose Input/Output) control associated with it. Element GPIO interfaces are

connected to Element via the Livewire network, so you can locate the actual machine connections either physically close to source equipment or in a central equipment room.

During installation, the engineer configures sources to accept and provide logic commands by selecting various options. Example: studio microphones may be set up to automatically mute the studio monitor speakers and illuminate an on-air warning light when turned on. The same sources can be configured to accept GPIO logic inputs from buttons so each studio guest can have their own **On**, **Off**, **Talk**, and **Mute** keys (see Chapter One, “Setup and Configuration” for details).

Line sources can be configured to receive *start* and *stop* commands from the Element as well. *Start/stop* messages can also be passed, via Ethernet, to a computer-based playout system.

Master Module and SoftKnobs

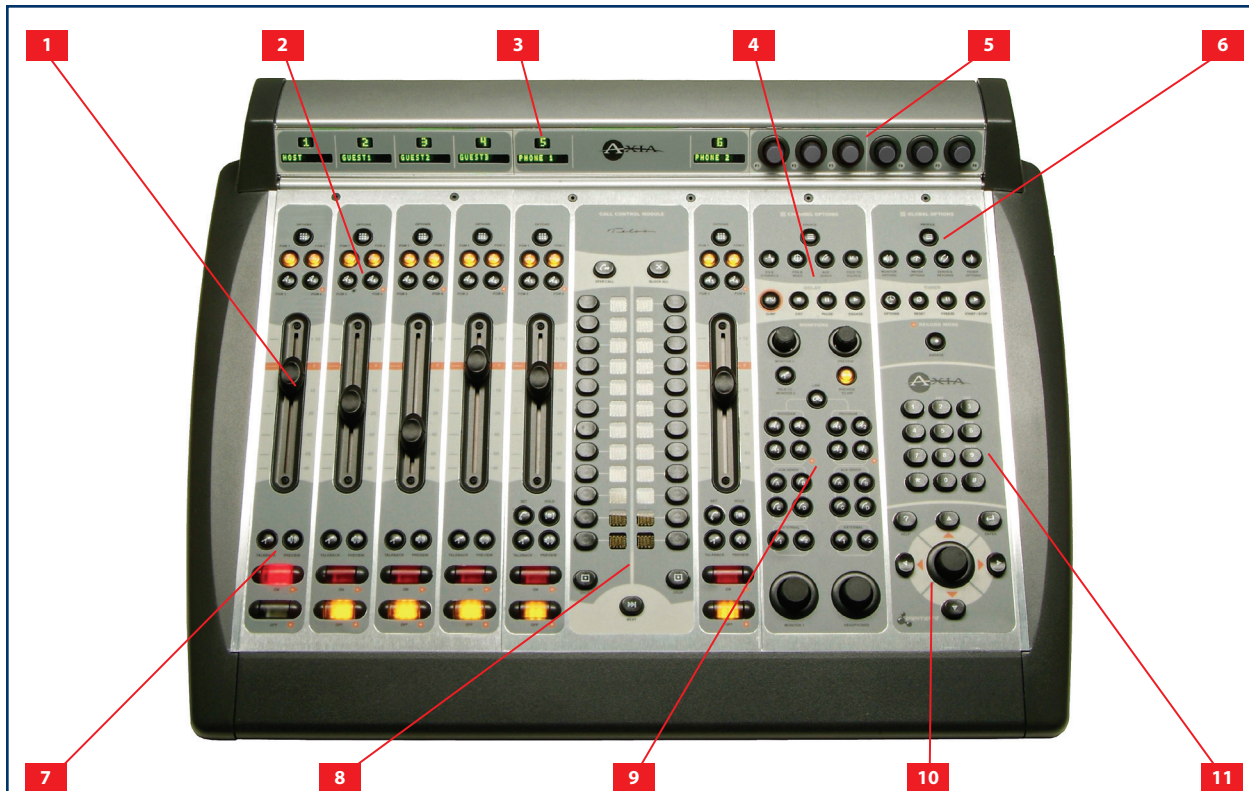
The Master Module and SoftKnobs contain Element’s operating controls. This is where you’ll find volume and selection controls for monitors and headphones, as well as Direct Access keys for fast access to show profiles, EQ & dynamics, Aux feeds and more. A numeric keypad gives direct control of broadcast phones and codecs; the scrub wheel and nav keys help navigate on-screen options.

Software

The Element operating system uses a high-performance embedded Linux kernel to help ensure bullet-proof, 24/7 operation. Software updates can be obtained from Axia via the Internet and applied by the station engineer. Updating system software will never affect Show Profiles or other configuration data already programmed into the system.

Control Callout

On the following page is a “bird’s-eye view” of a typical Element configuration, with major controls located and their functions explained.



1. **Surface Channels (faders)** can accommodate any input source. Each channel's **Options** button calls up expanded channel features.
2. **Assignment Buttons** for the four stereo main output buses – **Program-1, Program-2, Program-3** and **Program-4/Record** – are located atop each channel fader.
3. **Channel Displays** contain a 10-character text readout and a smaller accompanying display with Status Symbol icon displays (where appropriate) or the fader position number. Text is fixed for static sources like microphones and CD players, but can be set to show song/artist information, etc., when computer-based audio sources are connected.
4. **Master Module Channel Options & Profanity Delay Controls.** Channel Options section contains Direct Access Keys for quick access to a channel's Source assignment, EQ, Dynamics, Pan, Mix-Minus and Aux Sends settings. These keys become active whenever a channels Options key is pressed. Below the Channel Options keys, a set of keys are provided for convenient on-console control of external time-delay units.
5. **SoftKnobs** are context sensitive; when any Direct Access key is pressed, the on-screen display shows the function affected by rotating and/or pushing each knob.
6. **Master Module Global Options & Timer Control Keys** let operators recall Show Profiles, select monitor and metering preferences and affect other Element behaviors. Timer controls help keep track of on-screen elapsed and remaining time displays.
7. **Talkback & Preview Keys** let operators preview (cue) selected sources off-air, or quickly talk to any source(s) with associated backfeed(s).
8. **Optional Telos Phone Control Module** provides line control and Status Symbols for Telos TWOx12 or Series 2101 dual-hybrid broadcast telephone systems. Includes two dedicated faders.
9. **Monitor Section** contains controls for monitors, headphones and Aux Send buses.
10. **Scrub Wheel/Nav Keys** help navigate through on-screen options..
11. **Numeric Keypad** lets operators dial phone systems or codecs directly. Also provides numeric input during surface setup.

Control Details

General Channel Controls

Although Element’s individual surface channels can be assigned any audio source you like, some fader functions change their behavior depending upon the type of source assigned. Also, certain fader modules have “extra” controls for specific uses. The controls found on the Element 4-Fader Line Module (Figure 4-1) are common to all Element modules, so we’ll look at these first.

- **Channel Status Display**

The 10 character alphanumeric display shows the operator the source presently assigned to this channel. If the fader’s **Options** button is pressed and the operator changes sources, this display changes too.

As a safety feature, if the channel is ON while the operator is changing sources, the display will flash between the source currently on-air and the newly selected source. The channel will not “take” the new source until the operator has turned the channel off, preventing on-air errors.

Just above the alpha display is a Status Symbol display. This display indicates the fader’s channel number or, if there is a mix-minus defined for that channel’s source, displays a Status Symbol to indicate when the backfeed is active.

- **Channel Options Key**



Pressing this key activates the **Channel Options** keys located on Element’s Master Module. Refer to the “Master Module Controls” section later in this chapter for more information on these keys.

- **PGM-1, -2 & -3 Keys**



Each channel is assigned to either of three main stereo output buses by selecting any (or all) of these program keys. Generally, **PGM-1** is the main air bus and **PGM-2** are used for production and special program-

ming requirements. All program outputs are post-fader and post on-off function.

- **PGM-4 (Record) Key**



The fourth stereo bus can be used either as a regular output bus or as a dedicated recording and dubbing output. Surface channels assigned to **PGM-4/Record** are mixed and sent to a special output to feed your recorder/editor. This record bus is fed after the fader, but is independent of on/off status — the **Record** bus is fed whether the channel is on or off. This makes recording off-air phone calls or even producing commercials off air very easy; the operator only has to assign any channel whose source they wish to record to **PGM-4/Record** and activate the recording device.

- **Talkback Key**



Each fader is equipped with a **Talkback** key, located between the bottom of the fader and the channel **ON/OFF** keys. If the source assigned to a fader has an associated backfeed, pressing this key will allow the board operator to communicate with the person at the other end of the backfeed.

Multiple **Talkback** keys can be selected to talk to more than one source at a time.

Auto-Talkback: If a source’s Backfeed mode is set to “Auto”, the caller or remote talent will hear PGM-1 minus himself (mix-minus) whenever the surface channel he’s assigned to is **ON**, and will have his Talkback channel automatically activated when the channel is turned **OFF**. See Chapter Two: “Configuring Inputs” for more information on configuring backfeeds.

- **Channel Preview Key**



Preview (called “cue” in the analog days) allows the operator to listen to sources before they air. Element’s Preview bus allows auditioning of content in full stereo.

The **Preview** key acts like a latching switch.

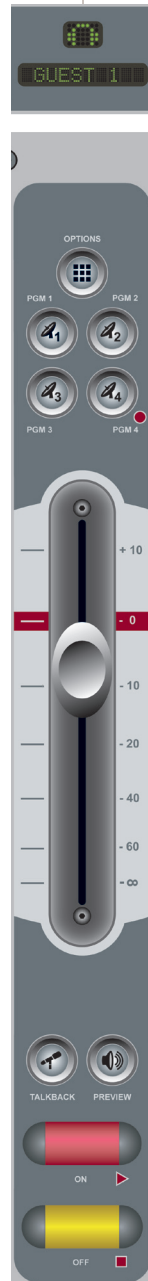


Figure 4-1:
Line Module
Fader Channel

Momentarily pressing any channel's **Preview** key assigns that channel to the Preview bus; pressing it again removes the channel from Preview. To enable an operator to quickly listen to various sources, the Preview function is interlocked: pressing **Preview** on any channel will remove any other channel from Preview mode.

If you want to preview multiple channels, pressing and *holding* any **Preview** key disables the interlock; other sources can be added to the Preview mix by pressing them.

Conversely, channels can be removed from a multiple Preview selection by pressing and *holding* any lit **Preview** key and deselecting the individual channels you wish to remove from Preview. A *momentary* press of any lit **Preview** key will remove all channels from Preview.

The audio from the Preview bus feeds the operator's headphones, as well as any speakers dedicated to the Preview bus. These speakers will be muted whenever the console operator's microphone is **ON**, or assigned to the Preview or Record buses.

• Fader

The fader controls the volume of the input source. There are two modes for the fader: **fader-start** and **fader-normal**:

1. When the channel is used in **fader-start** mode, pushing the fader all the way to the bottom of its travel turns the module off, and sends appropriate logic commands to source equipment. Moving the fader up turns the channel on and sends logic commands.
2. In **fader-normal** mode, the on/off status and start/stop commands will follow the channel on/off buttons independent of fader position.

For information on how to switch between **fader-start** and **fader-normal** modes, see the "Master Module Controls" section later in this chapter.

• On and Off Keys

As you'd expect, when the channel is in **fader-normal** mode the **On** key (▶) turns the channel on and the **Off** key (■) turns it off.

In **fader-start** mode, the **On** and **Off** keys act more as indicators of channel status; the **On** key will not actuate the channel if the fader is at the bottom of its travel.

If a channel is being used for a control-room mic, the control room monitors and preview speakers will mute when this channel is turned on or preview is actuated. Also, if a channel is configured to be the board operator's mic, pressing the **On** button will mute the channel, providing a "cough" function, until the button is released.

The channel **On** and **Off** functions also provide logic (start and stop pulses, monitor muting, etc.) appropriate to the selected source.

4-Phone Module Channel Controls

You already know that you can assign any type of source to any type of Element fader module. But, as mentioned previously, some fader modules include special keys that provide extra functions for certain types of sources.

The 4-Phone Fader Module (Figure 4-2) is one of these modules. If a normal line source is assigned to a Phone Fader, the fader and its controls operate exactly like a standard 4-Fader Module (see p.33). But when a **Telos TWO** ISDN phone hybrid is assigned to one of these faders, extra functionality is enabled.

Taking Calls

When controlling a Telos phone system, the Status Symbol box above the alpha display will indicate an available line by displaying a single dot icon. A call ringing in will show a "ringing" icon. It is answered by pushing either the **On** or **Preview** buttons. If **Preview** is already on, toggling to **Off** and then **On** will

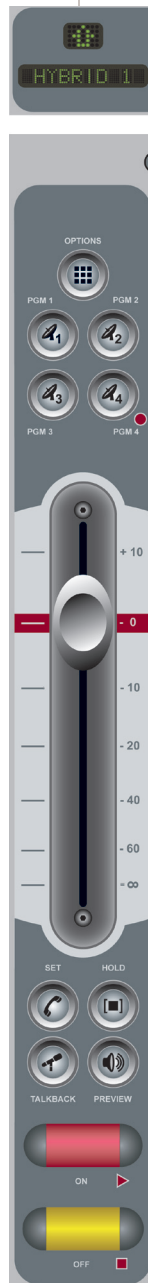


Figure 4-2:
Phone Module
Fader Channel

answer the call. When the call is answered, a “up arrow” Status Symbol icon will show. Also, a display window on the Element screen will show call status and incoming caller ID (if available).

When a channel is off, pressing **Off** drops the call. This is normally used to drop a call that has been on **Preview**. If a channel is on, pressing **Off** *once* will not drop the call. This is so you can keep a call active after using it on the air. A *second* press will drop the call.

Making Calls

To dial out, the Numeric Keypad on Element’s Master Module is assigned to the last fader channel that had a button press (**Off**, **On**, **Hold**, **Set**, or **Preview**). The dialing works like a mobile phone: you press all digits and then Enter to start the actual dialing. The Status Symbol icon above the fader indicates first that a line is active and then that the call is proceeding. A window will also open on the Element display screen showing the numbers dialed and call status.

- **Set Key** 

The **Set** button connects the telephone line assigned to a fader to a phone set connected to the Telos phone system, so that operators may use a normal handset to take or make calls. Since there is usually only one phone set used with a given system, these buttons are mutually exclusive across all the phone faders in an Element;

touching a **Set** key on any Phone Fader switches the handset to the line assigned to that fader and cancels any previous line selection.

- **Hold Key** 

Pressing **Hold** sends a hold command to the connected Telos phone system. When connected to European ISDN lines, a hold signal is sent to the network. With USA ISDN lines, no network hold function is available, so the hybrid enters a virtual hold state.

Call Controller + 2-Phone Module Controls

This module (Figure 4-3) works with Telos **TWOx12** and **Series 2101** multi-line talk show systems. Its center section features the twin hybrid/12-line controller found on Telos **Desktop Director** and **Call Controller** talk show controllers, flanked by two Phone Faders. The left Phone Fader is linked to the left hybrid control, and the right Phone Fader to the right hybrid control.

The Phone Faders controls on this module operate identically to the fader controls described previously. For details on hybrid control operation and description of the Status Symbols used

therein, please refer to your **Telos TWOx12** or **Series 2101** Operator’s Manual.



Figure 4-3: Call Controller + 2-Phone Fader Module

Master Module Controls

The Master Module (Figure 4-4) consolidates controls for all of Element's options and features into a single panel. Constantly-used controls like headphone and monitor selection and volume, plus command keys for the on-screen timers, controls for an external profanity delay unit, and a numeric keypad for convenient dialing of telephone systems and codecs are included.

Nav Keys & Scrub Wheel

Located at the bottom right of the Master Module is a **Scrub Wheel** surrounded by a set of navigation keys (**Up, Down, Right and Left**), an **Enter** key (↵), and a **Help** key (?). These controls augment the SoftKnobs and provide another way to navigate Element option screens.

Operators can use the directional keys to navigate through on-screen lists, and the **Enter** key to "take" a selection. The **Scrub Wheel** can also be used to scroll through on-screen lists.

Channel Options Section

These keys allow the operator to customize the behaviour of individual faders (surface channels). The operator first presses the **Options** key atop the fader s/he wishes to modify; the Channel Options screen (an overview of current channel options is displayed on Element's screen (Figure 4-5), and the Channel Options keys then become active.

Channel Jumping: If any channel is in **Options** mode, you can "quick jump" to set options for another channel by pressing the Options key on that channel.

For example, if you need to change **Source** on several channels, select the first channel using its **Options** key, and then choose the **Channel Options Source** key. The Source Selection list will appear on-screen. Once the source reassignment is made, simply press the **Options** key on the next channel you want to assign a new source to; the **Source** screen will remain active, but you're now adjusting settings for the new channel. The channel number is displayed onscreen as confirmation.

Placing a channel in **Options** mode lets you change inputs, apply EQ, change pan position, invert phase, set Aux Send and Aux Return levels, and adjust mix-minus settings for that channel. The **Options** displays (Figure 4-5) give detailed channel information. Both the Element channel number (the physical fader position, from left to right) and the source assigned to that channel are displayed for confirmation.

- **Source Key** 

Pressing **Source** displays the Source Selection menu. The Source Selection screen displays the currently selected source alongside a list of available sources Operators may select. Using the SoftKnobs, or Nav Keys, board-ops can scroll through and select from a list of available sources. They can also preview a source before



Figure 4-4: Master Module + SoftKnobs

Using the SoftKnobs, or Nav Keys, board-ops can scroll through and select from a list of available sources. They can also preview a source before

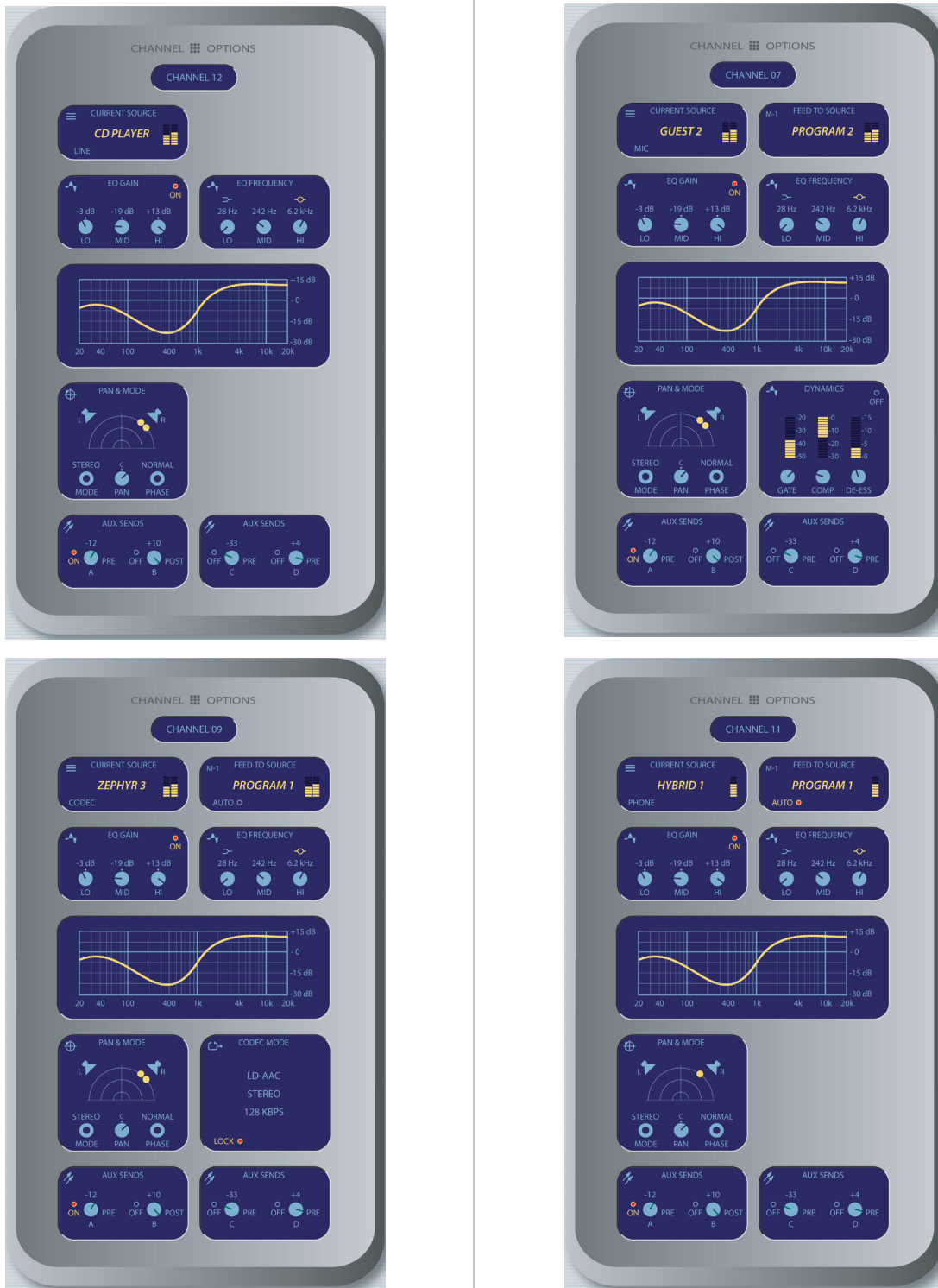


Figure 4-5: Channel Options screens. Information displayed differs based on the type of source currently assigned to the active fader. Shown clockwise from top left: Line Source Channel Options screen, Mic Source Channel Options screen, Phone Source Channel Options screen, Codec Source Channel Options screen.

selecting it.

- » **Select/Take:** rotating this SoftKnob scrolls through the available sources. Push to take.
- » **Previous:** When you assign a new channel source, Element remembers the last assigned source. The **Previous** command functions as a “speed key,” highlighting the previously assigned source in the source list. This is convenient when talent wishes to load a source for a short on-air segment, then return to the previous source when the segment is over. Press to take.
- » **Preview** allows you to hear the highlighted source through the Preview speakers prior to selecting it. Press and hold to listen; release when finished.
- » Pressing **Source** again returns to the Channel



Figure 4-6: Source menu screen.

Options screen.

• **EQ & Dynamics Key**

This command allows three-band parametric equalization to be applied to the selected audio source and, if that it is a mic source, to apply dynamics processing by Omnia. Pressing the **EQ & Dynamics** key causes the **EQ**

screen to appear; pressing it a second time displays the **Dynamics** screen, and pressing a third time exits.

When the **EQ** screen is displayed, a graphical representation of the equalization curve is displayed as the controls are adjusted. The screen also indicates the

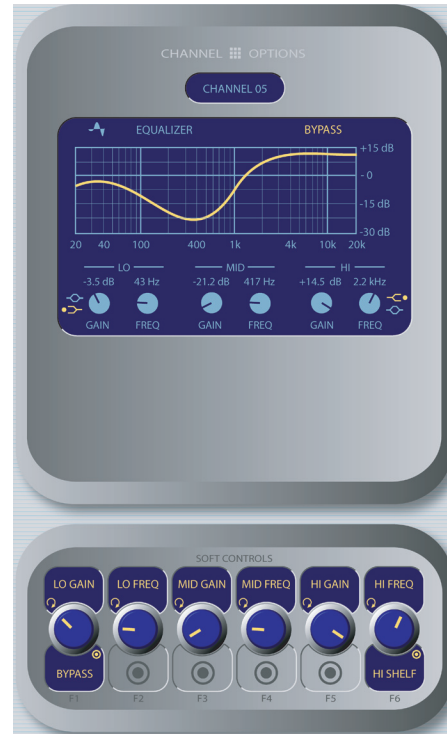


Figure 4-6: EQ menu screen.

amount of cut/boost applied, and the center frequency of each band.

- » **Lo Gain/Bypass/Active:** Rotating this SoftKnob adjusts the amount of gain or cut applied to the low EQ band. Pressing this knob toggles EQ between **Active** and **Bypass**.
- » Rotating the **Lo Freq** knob adjust the center frequency of the low EQ band (25 - 400 Hz).
- » Rotating the **Mid Gain** and **Mid Freq** SoftKnobs adjust gain and center frequencies of the mid-range EQ band (100 Hz - 1 kHz).
- » The **Hi Gain** knob adjust the boost/cut of the high frequency band.
- » **Hi Freq/Hi-Shelf/Bandpass:** Rotating this SoftKnob adjusts the center frequency of the high EQ band (1 kHz - 16 kHz). Pushing this knob toggles the high-band EQ method between **Band-pass** and **Shelf** modes.

- » Pressing **EQ & Dynamics** again displays the **Dynamics** screen.

Note: High, Mid and Low band adjustments provide up to 25 dB of cut and 15 dB of boost. Element EQ features SmartQ™ automatic bandwidth system that varies the Q of the selected parametric band to provide the most pleasing EQ effect. For details on SmartQ, please refer to Page 7 of this manual.

Dynamics controls are available if the selected source is a microphone input. Pressing the **Dynamics & EQ** key twice invokes the **Dynamics** menu.



Figure 4-7: Dynamics menu screen.

From here, you can add **Gating**, **Compression** and **De-essing** to a microphone channel. Normally, these values are part of a pre-set Show Profile; however, on-the-fly control is provided for talent to make adjustments when needed.

What is this stuff? Radio engineers don't need a lesson in dynamics processing, but for the benefit of any jocks reading this, here's a quick explanation of what these controls can do.

A **Gate** is a dynamics device whose function is to remove unwanted audio material below a certain threshold. A gain circuit is employed to

raise or lower the volume of the audio signal; when the signal falls below a certain set threshold, the audio level drops down to a predetermined level. The reason they are called gates is because when they "close" it sounds as if the audio has suddenly stopped, or has been "gated." You might use this on a mic channel to eliminate background noise during times when you're not speaking.

Threshold settings determine at what level the processor will begin working. For example, on a compressor, when signal level exceeds the threshold setting, it will be compressed; below the threshold the signal will remain uncompressed. On a gate, threshold determines the minimum input level required to cause the gate to open up and pass signal; when input level drops below the threshold, the gate will be closed preventing signal from passing. Carefully setting the threshold allows you to very specifically control when processing is being applied to a signal.

Compression controls the dynamic range of an audio signal. It generally reduces the volume of very loud audio, helping to keep from "blowing out" the mic channel with too much volume. It can be used to make your mic channel sound "fatter," evening out the highs and lows of your vocal pattern. It can also increase the overall perceived fullness of your voice. Too much compression, however, can make your voice sound hard and "squashed." Use it sparingly!

To use compression, you must first set your threshold value (usually expressed in dB). When the audio signal is louder than this threshold, its gain is reduced. The amount of gain reduction applied depends on the **Compression Ratio** setting. For example, with a 2:1 ratio, for every 2 decibels the input signal increases, the output is allowed to increase only 1 decibel.

The **Freeze Gate** is used to prevent "suck-up" of room noise during speech pauses. When active, gain is not increased by the compressor when audio is absent. "Freeze" would be the default, since it almost always improves results.

De-Essing is just what you think it is - a way to electronically remove extra sibilance from your mic channel. If your mic is particularly sensitive to "s" and "th-" sounds and accentuates them on-air, De-Essing helps reduce these sounds to more manageable levels. As with compression, too much de-essing will produce annoying and unnatural results - you'll wind up sounding like you're talking through a cotton sock! Stick to small amounts to get the effect you want.

- » Rotating the **Exp Gate Thresh/Bypass/Active** SoftKnob varies the level of signal threshold needed for the **Gate** function to activate (between -50 and 0dB). Pushing this knob toggles Dynamics between Active and Bypass states.

- » The **Exp Gate Depth** knob sets the amount of gain reduction when audio level falls below the **Exp Gate Threshold** value (adjustable between -3 and 0dB).
- » **Comp Thresh/Bypass/Active** sets the upper limit of the level at which the compressor will begin to work (between -30 and 0dB). Pushing the SoftKnob toggles between Active and Bypass states.
- » **Comp Ratio/Freeze/No Freeze** sets the compression ratio (variable between 1:1 and 16:1). Pushing the SoftKnob activates and deactivates the compressor's Freeze Gate function.
- » **De-Ess Thresh/Bypass/Active** sets the audio level above which the de-esser will be engaged (from -20 to 0dB). Pushing the SoftKnob toggles between Active and Bypass states.
- » **De-Ess Ratio** sets the amount of sibilance reduction, in dB, that will be applied (adjustable between 1:1 and 8:1).
- » Pressing the **Dynamics & EQ** key again will return you to the **Channel Options** screen.

● **Pan & Mode Key**



Figure 4-8: Pan / Mode menu screen.

This function allows the operator to adjust the pan or balance of the channel source and correct signal phase errors.

- » **Pan/Balance/Bypass/Active:** Rotating this SoftKnob adjusts the the Pan setting for mono sources and the Balance setting for stereo sources. Pushing this knob toggles between Active and Bypass settings.
- » **Mode:** If the selected source is in stereo and the Source Profile for the selected source permits it, (see Chapter Two for information on Source Profiles), rotating this control lets the board op switch source audio between **Stereo** (discrete left/right), **Left** (left source channel fed to L/R input), **Right** (right source channel fed to L/R input) and **Sum** (left and right source channels summed to mono and fed to L/R input).
- » **Phase:** If the selected source is in stereo, rotating this control lets the board op compensate for out-of-phase audio sources. **Normal** is used for in-phase sources; **Inv Left** inverts the phase of the left input channel only, **Inv Right** inverts the phase of the right input channel only, and **Invert** reverses phase of both stereo channels. If the selected source is in mono, the choice are **Normal** and **Invert** only.
- » Pressing **Pan & Mode** again returns to the Channel Options screen.

» **Aux Sends Key**



Element has four stereo Aux buses that can be used as utility buses for mixing, for constructing custom IFB mixes, or as effects buses for production. Pressing this key displays the Aux Sends screen.

- » More information about these controls is forthcoming. Watch this space.
- » Pressing **Source** or **Options** a second time exits this menu.

● **Feed To Source Key**



If the assigned source (i.e., a codec or phone hybrid) has an associated backfeed or mix-minus, pressing this key allows the board op to change the source of the audio being backfed to the channel source. This selection is normally pre-set when a Source Profile is constructed



Figure 4-9: Aux Sends menu screen.



Figure 4-10: Feed to Source menu screen.

(see Chapter Two, “Configuring Inputs,” for information on Source Profiles), but can be modified on-the-fly.

- » The **Select/Take** knob, when rotated, scrolls through the list of available audio sources that can be backfed to the phone or codec.
- * **Automatic** backfeeds audio from the Talkback bus when the channel is **Off**, and audio from the Program-1 bus when the channel is **On**, with mix-minus if needed.
- * **Program 1 - Program 4** backfeeds audio from the selected Program bus, with mix-minus if needed.
- * **Aux Send 1 - Aux Send 4** backfeeds audio from the selected Auxiliary bus, with mix-minus if needed.
- » Rotating the **Dim** knob allows the operator to adjust, in dB, the amount by which the studio monitors are dimmed when this fader channel’s **Talkback** key is pressed.
- » Pressing **Feed to Source** or **Options** a second time exits this menu.

Delay Section



These keys can be linked to provide from-the-board

control on an external profanity delay unit. Provides activation of **Dump**, **Exit**, **Pause** and **Engage** functions. Refer to your delay unit’s Owner Guide for information on how these functions work.

Monitor Section

This controls in this section provide direct operator access to monitor selection and volume controls.

• Monitor 1 Knob + Source Keys



This knob, at the bottom left of the Monitor section, controls the selection and volume of the source heard in the **Monitor 1** speakers (typically the Control Room monitors).

- » Rotate the knob to vary the gain of the **Monitor 1** speakers.
- » Use the Direct Selection Keys in the column directly above the **Monitor 1** knob to send the output of **Program 1 - Program 4**, **Aux Send A - Aux Send D**, or the two **External** monitor sources to the Monitor 1 speakers. These keys light to denote the selected source.
- » You can listen to any frequently-used audio stream

in your facility instantly (such as a network feed, off-the-air monitor or another studio's Program output) by assigning it to one of the two **External** selector keys - like preset buttons on a car radio. If you press and hold either of these keys for five seconds, a list of available audio streams will pop up onscreen. You can then scroll through this list and select the source you want to hear in the monitors when that key is pressed. Each **External** key is independently programmable.

- **Headphone Knob + Source Keys** 

This knob, at the bottom right of the Monitor section, controls the selection and volume of the source heard in the board operator's headphones.

- » Rotate to vary the gain of the **Headphone** feed.
- » Use the Direct Selection Keys in the column directly above the **Headphones** knob to send the output of **Program 1 - Program 4, Aux Send A - Aux Send D**, or the two **External** monitor sources to the headphones. These keys light to denote the selected source.
- » Press and hold either of the **External** selector keys to display and select from a list of available audio streams to monitor on these keys.

- **Link Key** 

This key toggles the **Monitor 1 / Headphone** source **Link** function. When active, a change to the **Monitor 1** source automatically selects that source for the **Headphone** feed, and vice versa. The **Link** key illuminates when this function is active.

- **Monitor 2 Knob** 

This knob, at the top left of the Monitor section, controls the selection and volume of the source heard in the **Monitor 2** speakers (typically Studio or News booth monitors).

- » Rotate to vary the **Monitor 2** speaker gain.
- » Push the knob to see an on-screen display of sources that can be assigned to the **Monitor 2**

feed. Rotate the knob to highlight the desired source, and push again to "take".

- **Talk To Monitor 2 Key** 

The board operator can push and hold this key to talk to the talent listening to the **Monitor 2** feed. While the key is depressed, audio from the Control Room mic channel is fed, pre-fader, to the **Monitor 2** speakers and headphones (if used). Releasing the key ends the talkback function.

- **Preview Knob** 

This knob, at the top right of the Monitor section, controls the volume of the source heard in the **Preview** (cue) speakers. Rotate the knob to vary the gain of the **Preview** speakers.

- **Preview To HP Key** 

This key toggles the **Preview To Headphone** function. When active, this function sends **Preview** audio to the board operator's headphones. When inactive, audio from the **Preview** bus is heard exclusively in the **Preview** speakers. The **Preview To HP** key illuminates when this function is active.

Global Options Section

The **Global Options** keys are used to load Show Profiles and to customize Monitor, Meter, Aux Sends/Returns and Fader settings.

- **Profile Key** 

Pressing this key allows the board operator to browse and select from the list of saved Show Profiles, custom surface configurations that can be loaded as needed to reconfigure Element for different shows and talent.

As soon as you "take" a new show profile, Element begins loading it. During the profile loading process, a status line reads **Loading (Show Profile Name)** to inform the operator of the operation in progress.

Note: Element will never interrupt an active source in order to load a new one.



Figure 4-11: Show Profile selection screen.

Let's say that the operator has a satellite feed assigned to Channel 10, and this source is currently on-the-air (assigned to **Program 1**, fader up, channel **On**). The board op loads a new **Show Profile** to ready Element for the next show; the Show Profile he's loading has a CD player assigned to Channel 10. Obviously, there's a conflict.

Instead of blindly changing the channel source and interrupting audio that's on-the-air, Element queues the new source. When the board op is finished with the satellite feed, he turns the channel **Off** and the CD player automatically loads.

Along with selecting a new **Show Profile** to load, users can reload the currently active profile, resetting the board to a "nominal" state if they wish to return to their normal configuration after making on-the-fly changes.

- » **Select / Take:** Rotate this knob to scroll through the on-screen list of available Show Profiles. Push the knob to "take" the new Show Profile.
- » **Previous:** When you load a new Show Profile, Element remembers the prior Profile. The **Previous** command functions as a "speed key," highlighting the previously assigned source in the

source list. This is convenient when talent wishes to load a special-purpose Show Profile for an on-air segment, then return to the previous Show Profile when the segment is over. Press to "take."

Useful Tip: Element is full of options intended to make life on-air easier and more intuitive for today's talent. However, it's possible for talent not yet familiar with the way Element works to "get lost" in the option settings.

Don't Panic! Pressing and holding the **Profile** key for five seconds will instantly clear changes made to the current Show Profile, resetting the Surface to the Show Profile's default settings. All is well, and your jock once again knows where his towel is.

• Monitor Options Key



Pressing this key reveals all sorts of neat options relating to Monitor and Headphone behavior. The Monitor Options screen is shown in Figure 4-12.

- » Pushing the **Talk Levels** SoftKnob selects between adjusting the levels of the **Monitor 1 Talk To** and **Dim** levels, and the **Monitor 2 Dim** level. The selected function will highlight in yellow on

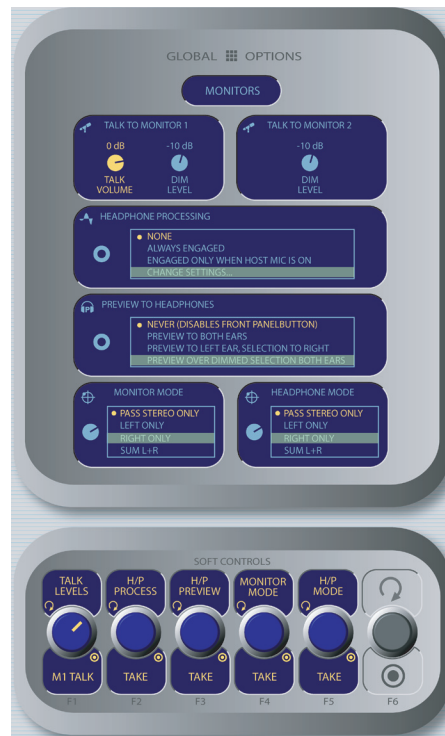


Figure 4-12: Monitor Options menu screen.

the screen. Turn the knob to adjust the values of these settings.

- » The **H/P Preview Knob** controls whether audio from the **Preview** bus can be fed to the board op's headphones, and if so, in what manner. Rotate this knob to select from the following options:

- * **Never (Disables Front Panel Button):** selecting this option locks out the **Preview To HP** key located in the Master Module's **Monitor** section. The operator will not be able to send **Preview** audio to the headphone channel.

- * **Preview To Both Ears:** When a surface channel's **Preview** key is pressed, audio from the **Preview** channel overrides the **Headphone Monitor** selection and is sent to both the left and right channels of the operator's headphones.

- * **Preview To Left Ear, Selection To Right:** When a surface channel's **Preview** key is pressed, audio from the **Preview** channel overrides the **Headphone Monitor** selection *in the left headphone channel only*. Program audio is still present in the right channel of the operator's headphones.

- * **Preview Over Dimmed Selection Both Ears:** When a surface channel's **Preview** key is pressed, the **Headphone Monitor** selection is dimmed in both ears and audio from the **Preview** channel is heard over the Program audio in both the left and right channels of the operator's headphones.

*

After making a selection, push the knob to **Take** the new value.

- » **Monitor Mode Knob:** Rotate this knob to select from the following options.

- * **Pass Stereo Only:** The selected source will be presented in stereo on the Monitor speakers.

- * **Left Only:** The left channel of the selected source will be presented in both Monitor speakers.

- * **Right Only:** The right channel of the select-

ed source will be presented in both Monitor speakers.

- * **Sum L+R:** Both channels of the selected source will be summed and presented to both Monitor speakers.

After making a selection, push the knob to **Take** the new value.

- » **Headphone Mode Knob:** Works like the **Monitor Mode** knob described above, but affects the board operator's **Headphone** channel.

That "Radio" Sound: Over the years, talent has become used to listening to themselves off-air during raps, hearing their voices with dynamics processing applied, and relying on the levelling effects of processing to "ride the compressor" when the mic is open.

In the age of HD Radio and 8-second delay, monitoring off the air signal is sadly no longer possible. But monitoring the **Program** bus leaves most talent feeling that something's missing from their headphones.

To help combat this, Element offers customizable headphone EQ and built-in headphone dynamics processing by Omnia. When this feature is used, Talent can monitor **Program** audio with no delay and still hear the comfortable "air" sound they're used to without the hassle and expense of outboard processors dedicated to the headphone channel.

As with nearly every Element option, talent's individual headphone channel preferences can be saved in their customized Show Profile for instant recall later.

- » The **H/P Process** knob allows the board op to turn Headphone Processing on and off, define how it operates if active, and change Headphone Processing settings. Turning the knob selects one of the following:

- * **None.** No processing is applied to the **Headphone** channel.

- * **Always Engaged.** Processing is applied to the **Headphone** channel at all times.

- * **Engaged Only When Host Mic Is On.** Processing is applied only when the surface channel with the Host mic assigned has been turned **On**.

- * **Change Settings...** enters the **H/P Process** screen.

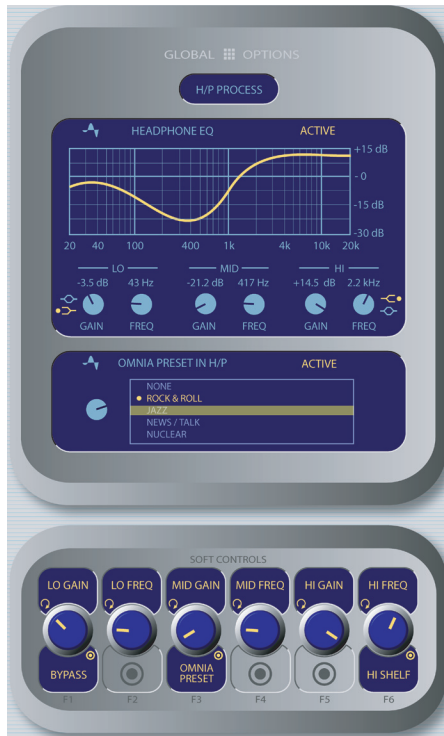


Figure 4-13: H/P Process settings.

Highlight the desired setting and push the knob to **Take** that setting.

Selecting **Change Settings** takes the operator to the **H/P Process** screen shown in Figure 4-13.

The operator is presented with a graphical representation of the EQ curve applied to the headphone channel, and a list of format-tuned Omnia dynamics processing presets that may be selected.

From this screen, talent can apply three-band parametric equalization to his or her headphone channel, and select headphone dynamics processing. The screen indicates the amount of cut/boost applied, and the center frequency of each band as well as a list of Omnia processing presets.

- » **Lo Gain/Bypass/Active:** Rotating this Soft-Knob adjusts the amount of gain or cut applied to the low EQ band. Pressing this knob toggles EQ between **Active** and **Bypass**.
- » Rotating the **Lo Freq** knob adjust the center frequency of the low EQ band (25 - 400 Hz).

- » **Mid Gain/Omnia Preset:** Rotating this knob adjust the amount of gain or cut applied to the midrange band. Pressing this knob does something with the dynamics processing; Mike still needs to tell me how it works.
- » The **Mid Freq** SoftKnob adjust the center frequency of the midrange EQ band (100 Hz - 100 kHz).
- » The **Hi Gain** knob adjust the boost/cut of the high frequency band.
- » **Hi Freq/Hi-Shelf/Bandpass:** Rotating this SoftKnob adjusts the center frequency of the high EQ band (1 kHz - 16 kHz). Pushing this knob toggles the high-band EQ method between **Band-pass** and **Shelf** modes.
- » Press the **Monitor Options** key to exit and return to the **Monitor Options** menu.

Note: High, Mid and Low band adjustments provide up to 25 dB of cut and 15 dB of boost. Element EQ features SmartQ™ automatic bandwidth system that varies the Q of the selected parametric band to provide the most pleasing EQ effect. For details on SmartQ, please refer to Page 7 of this manual.

- » Pressing **Monitor Options** or **Options** exits this menu.

• Sends & Returns Key

This key affects global settings for the master send and return levels of the four stereo **Aux Send** and two stereo **Aux Return** buses. The outputs can be turned **On** or **Off** and can be sent to the preview speakers and headphones (if preview is enabled in headphones).

Element's Aux buses can be used as utility buses for mixing, for constructing custom IFB mixes, or as effects buses for production. Pressing the **Sends & Returns** key once takes you to the **Aux Sends Master** screen (Figure 4-14); pressing it twice displays the **Aux Returns Master** screen (Figure 4-15), and pressing a third time exits the menu.

- » The **Send A - Send D/On/Off** SoftKnobs, when rotated, set the Master Send Gain for each **Aux Send** bus. Pushing any of these knobs toggles that individual **Aux Send On** or **Off**.
- » Rotating the **Select/Preview** knob allows you



Figure 4-14: Aux Sends menu screen.

to select any of the four **Aux Send** mixes for auditioning in the **Preview** channel; the on-screen menu indicates which **Aux Send** is selected.



Figure 4-15: Aux Returns menu screen.

Pushing this knob previews the contents of the selected bus; releasing it ends the preview.

- » Pressing the **Sends & Returns** key advances you to the **Aux Returns Master** screen.
- » Rotate the **Return A/On/Off** SoftKnob to set the audio level of **Aux Return A**. Push the knob to toggle **Aux Return A On** or **Off**.
- » Rotate the adjacent **Assign/Take** knob to choose which of the four **Program** buses to assign the output of **Aux Return A**. Push the knob to **Take** the assignment.
- » Rotate the **Return B/On/Off** SoftKnob to set the audio level of **Aux Return B**. Push the knob to toggle **Aux Return B On** or **Off**.
- » Rotate the adjacent **Assign/Take** knob to choose which of the four **Program** buses to assign the output of **Aux Return B**. Push the knob to **Take** the assignment.
- » Rotating the **Select/Preview** knob allows you to select either of the **Aux Return** mixes for auditioning in the **Preview** channel; the on-screen menu indicates which **Aux Send** is selected. Pushing this knob previews the contents of the selected bus; releasing it ends the preview.
- » Press the **Sends & Returns** key to exit the menu.

• Fader Options Key

Element supports fader-normal and European-style fader-start channel activation methods. In default **Normal** operation, Element conforms to the US method of requiring talent to manually turn the channel on and off. **Fader Start** mode follows the European convention of activating the channel and associated machine logic when the fader is raised from ∞ . The **Fader Options** menu allows you to choose which method of operation you wish to use. This choice applies to all faders.

- » Rotate the **Start Mode/Take** SoftKnob to choose between **Normal** and **Fader Start** modes. Press this knob to **Take** your selection.
- » Pressing the **Fader Options** key again exits.

Additionally, *pressing and holding* the **Fader Options Key** for five seconds enters the **IP Address Book** discussed in Chapter 1.



Figure 4-16: Fader Options menu screen.



Figure 4-17: Timer menu.

Timer Section

The Timer control keys operate Element's onscreen event and countdown timers, and allow users to set options determining the behavior of these timers.

• Timer Options Key

Pressing the **Timer Options** key opens the Timer Menu (Figure 4-17). This menu allows users to adjust time & date-related settings, as well as entering values for Element's countdown timer.

- » Rotate the **Select/Set** SoftKnob to scroll through the list of available options. To select an option to modify, push this knob. While modifying an options, rotate the knob to change an option's value, and push the knob to move to the next field.
- » Available options include:
 - * **Set Time Of Day Clock:** If you do not have an NTP (Network Time Protocol) server configured as part of your Livewire IP-Audio network, use this option to set the correct time for Element's on-screen clock.
 - * **Set Calendar Date:** Choose this option to set Element's on-screen calendar.

- * **NTP On/Off:** If you have an NTP server tied to your IP-Audio network, set this option **On**. If you do not, or if your NTP server is supplying incorrect time data, turn this option **Off** to keep time with Element's internal clock.
- * **12/24-Hour Format:** This option switches the on-screen time display between 12-hour (normal-style) and 24-hour (military-style) timekeeping.
- * **Time Zone GMT+:** Use this field to set the global time zone your station is located in, relative to Greenwich Mean Time.

Time And A Word: The term "Greenwich Mean Time" refers to Britain's Royal Greenwich Observatory, which lies athwart the Prime Meridian. Time at this location is referred to as "base time," to which all other time zones are relative. GMT is properly referred to as UTC - "Coordinated Universal Time."


If you're at a loss as to what the GMT offset is for your location, a complete list is located online at www.wikipedia.org. Enter a search for "Time Zone" and scroll down the resulting page.


- * **Countdown From:** When the operator enters


an interval of time into this field, Element's countdown timer will count backwards from this number to zero upon exit of this screen.

- » Pressing the **Timer Options** key again exits this menu.

Additionally, *pressing and holding* the **Timer Options Key** opens the Clock & Timer Options Menu, discussed in Chapter 1.


- **Reset Key:**  Pressing this key resets Element's on-screen Event timer to zero.

- **Freeze Key:**  Momentarily pressing this key pauses Element's Event timer but does not reset it to zero. Releasing the key restarts the timer's count from its last position.

- **Start/Stop Key:**  If the Event timer is not running, pressing this key resets any previous value to zero and re-starts the count. If the timer is running, pressing this key stops the countdown and retains the last displayed timer interval.

Record Mode Section

Element's **Record Mode** is a special "one-touch" mode which can be set to automatically start a designated record device or application, assign pre-defined sources to the **Program-4/Record** bus, and automatically change Monitor sources so that talent can record a phone call or remote interview with a single button-press. **Record Mode** settings are unique to each **Show Profile** and are defined using the Show Profile setup screen via Element's HTTP interface (see Chapter 5, "Show Profiles", for details).

- **Engage Key:**  Press this key to put Element into **Record Mode**. Pressing this button can activate a specified recording device, and assign certain surface channels to the **Program-4/Record** bus. Options are unique to each Show Profile. Pressing the key a second time deactivates **Record Mode** and returns the surface to its normal state.

Source-Specific Channel Controls

Some fader functions change their behavior to suit the type of source assigned to the fader. A Host Mic input has unique functions different from those of a Codec input, and so forth. In this section we'll explain the channel functions unique to each type of source input, as well as providing functional block diagrams of each type.

Control Room Host Microphone Channel Operation

The Control Room Host microphone is the board operator's mic. It's always located in the Control Room, so activating a channel designated as the CR Host mic affects CR monitor speaker and Preview speaker muting.

When the board operator pushes a **Talkback** key, all output bus assignments for the Control Room Host microphone are temporarily muted, while his mic's audio is routed to the requested **Talkback** destination. Releasing the button returns the channel outputs to normal. (Pushing your own **Talkback** key does nothing.)

Pressing the channel **On** key for the CR Host Mic while the channel is already on will mute the channel until the button is released, acting as a "cough" function for the board operator. On-air status and speaker mutes are unaffected by this action. The CR Host microphone channel will mute the Control Room speakers and the Preview speakers whenever the channel is **On**, or if **Preview** or **Record Mode Engage** keys are selected.

The block diagram for this channel selection is shown in Figure 4-18.

Control Room Guest Microphone Channel Operation

There are often microphones in the Control Room other than the board operator's; i.e., an in-studio guest wing, or perhaps a producer or announcer located physically near the board op. These microphones control the muting of the Control Room monitor speakers.

The Control Room monitor speakers and Preview speakers will mute when this channel is turned on, or when **Preview** or **Record Mode** are selected.

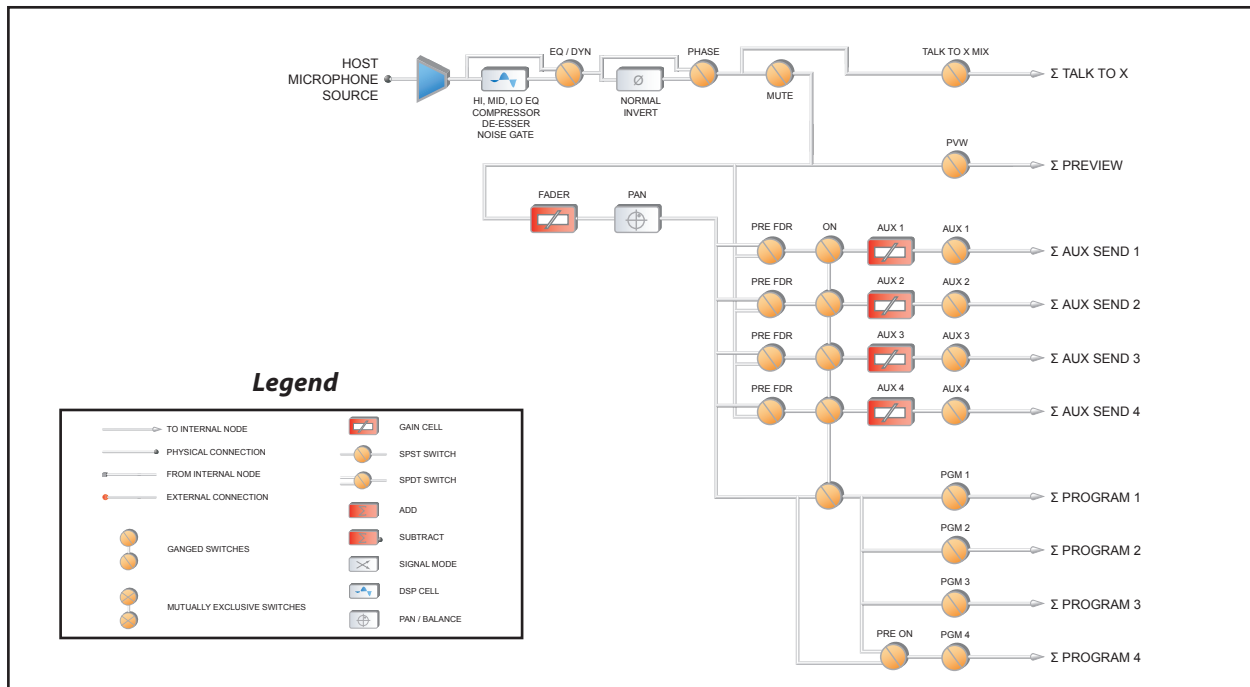


Figure 4-18: Control Room Host Microphone channel block

Control Room Guest mics can be turned on and off remotely, and provide on/off status to a remote logic device. **Remote Talk** and **Remote Mute** functions are also provided, and can be activated using a remote On/Off/Talk/Mute panel.

Remote Talk lets talent and guests communicate with the board operator. When **Talk** is remotely activated on a CR Guest mic channel, outputs to program buses for that channel are muted and the mic audio is fed to the Talkback bus. While this is active, the **Talkback** key will be lit and the Status Symbol will display the word **TALK**. When the user stops talking, the channel returns to normal, and its **Talkback** key flashes for three seconds, alerting the board operator to the guest who spoke to him, and enabling him to reply using the **Talkback** key.

When the **Remote Mute** function is activated, the Status Symbol displays **MUTE**, and all the channel outputs are muted until the command is released.

Control Room Guest Microphone channels will mute the Control Room speakers and the Preview speakers whenever the channel is **On**, or if **Preview** or **Record**

keys are selected. If only the **Phone** key is selected, the Control Room speakers will mute independently, leaving the Preview speakers un-muted.

Finally, Control Room Guest Microphone channels have provisions for an individual Headphone feed. This is especially useful for talent and guests in the same room as the board operator; their headphone feeds will all monitor the audio selected on the **Monitor 2** selector, but the board op can communicate with each person privately using **Talk To Previewed Source**.

The block diagram for this channel selection is shown in Figure 4-19.

Studio Microphone Channel Operation

The third type of mic channel is a Studio Microphone. This mic is typically located in a location separate from the control room — perhaps a voice-over booth, a talk studio separated by glass, or a news booth.

Logic for Studio Microphone channels is identical to that of the Control Room Guest Microphone channel. Studio Microphone channels can be turned on and off remotely and provide on/off status to a remote logic

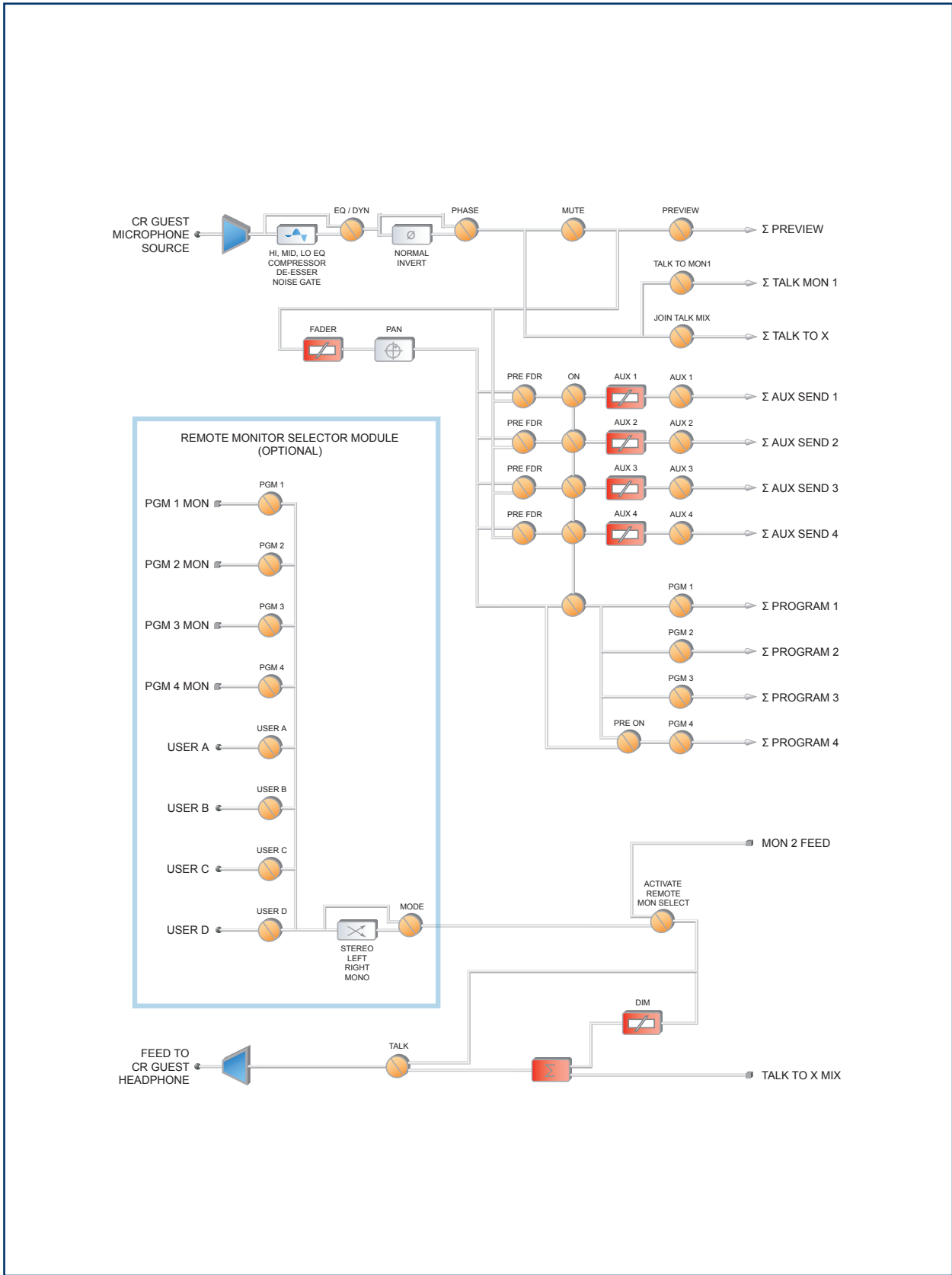


Figure 4-19: Control Room Guest Microphone channel block

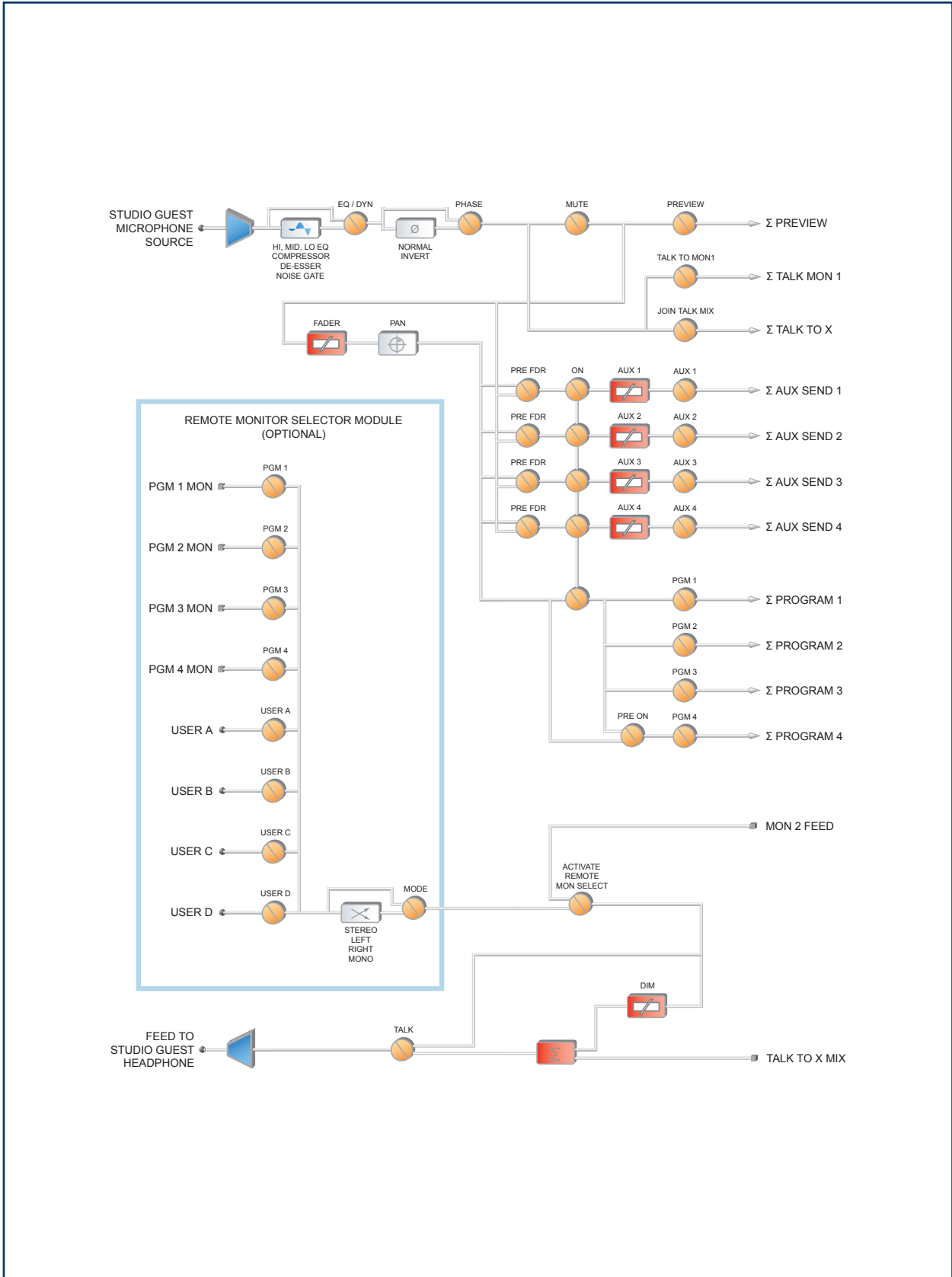


Figure 4-20: Studio Microphone channel block

device. **Remote Talk** and **Remote Mute** functions are provided for use with a remote On/Off/Talk/Mute panel.

Remote Talk lets talent and guests communicate with the board op. When **Talk** is remotely activated, outputs to program buses for that channel are muted and the source audio is fed to the **Talkback** bus. While active, the **Talkback** key is lit and the Status Symbol displays **TALK**. When the user stops talking, the channel returns to its previous state, and its **Talkback** key flashes for three seconds, alerting the board op to the guest who spoke to him, and enabling reply via the **Talkback** key.

When **Remote Mute** is activated, the Status Symbol will display **MUTE**, all channel outputs are muted until the key is released, and the channel is again active.

The **Monitor-2** speakers mute when a Studio Microphone channel is turned on, or if it is assigned to the **Program-4/Record** bus.

The Studio Microphone channel can be associated with an individual Headphone feed. All Studio Micro-

phone-associated feeds will monitor the audio selected on the **Monitor 2** selector, but the board op can communicate privately using the **Talkback** function.

The block diagram for this channel selection is shown in Figure 4-20.

Line Channel Operation

Line-level devices like CD players, delivery system outputs, tape players, etc., use the Line Channel profile. The Line Channel is stereo, but the user can change the input mode to mono, left only or right only by using the **Pan & Mode** key in the Master Module's **Channel Options** section.

Line channels have a variety of remote logic functions available at the GPIO interface. **On** and **Off** keys can be remotely controlled, as well as the **Preview** key. Each function has a corresponding lamp driver.

A **Reset** function is provided to remotely turn the channel **Off** while suppressing the **Stop** command, use-

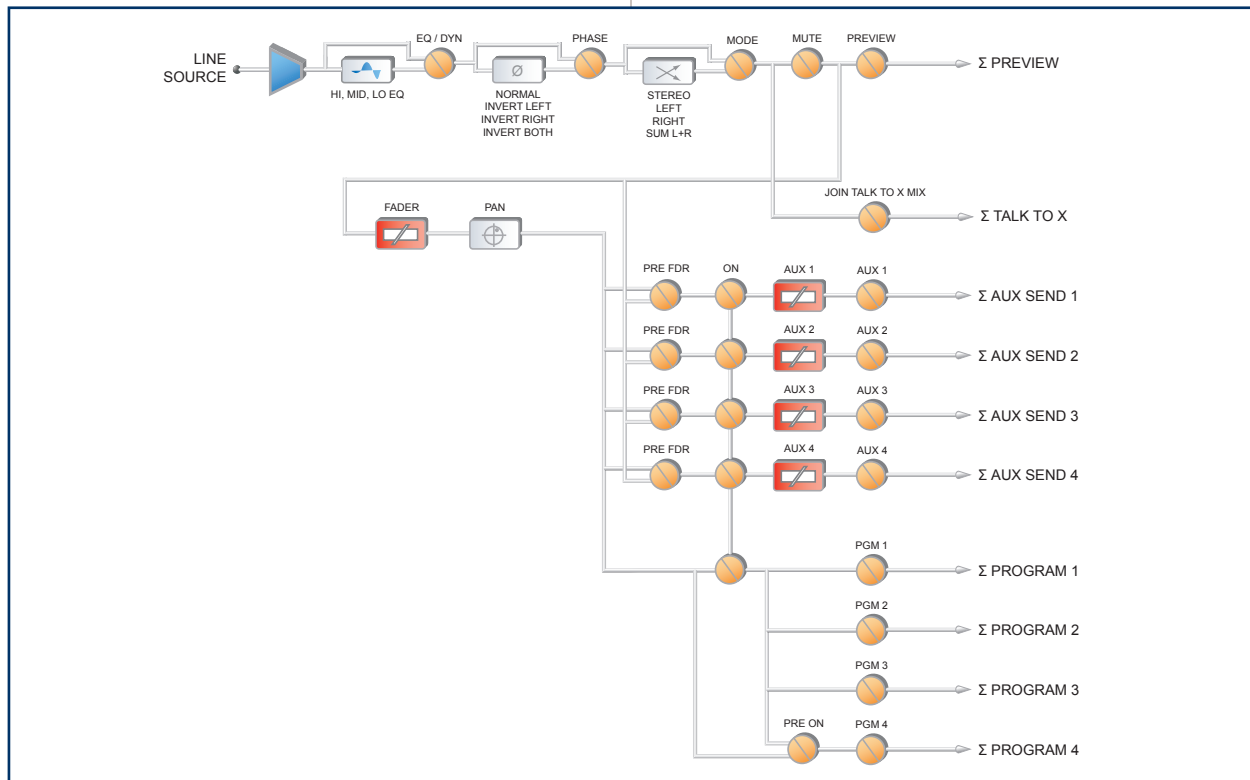


Figure 4-21: Line channel block

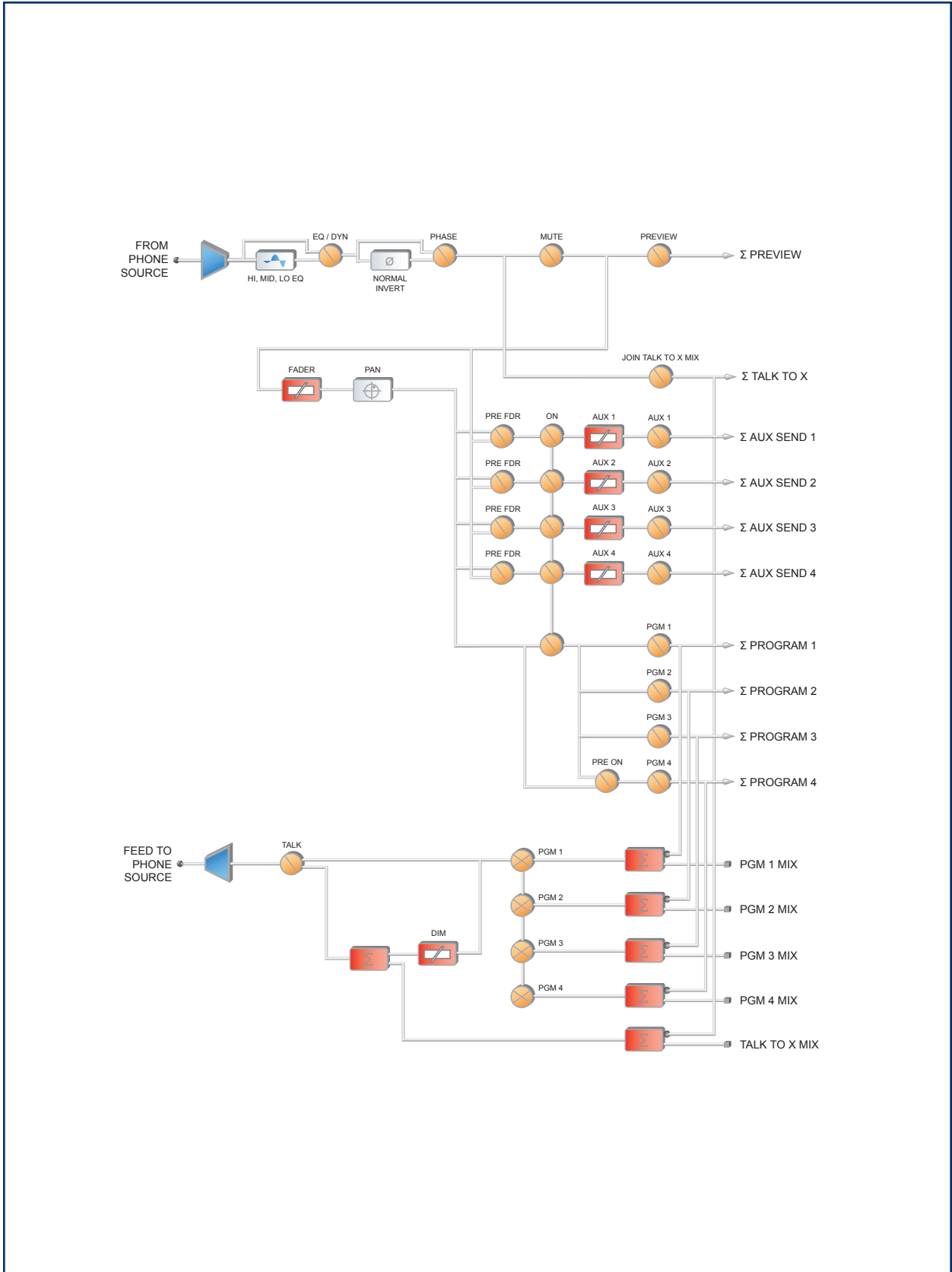


Figure 4-22: Phone channel block

ful for allowing certain types of tape machines to re-cue. You'll also find a **Ready** command which will remotely illuminate the **Off** lamp on the Element channel and activate the GPIO **Off** lamp driver, allowing source equipment to signal the operator with either a steady-state **Off** button (indicating source ready) or a flashing **Off** button (indicating source next). Refer to the GPIO Node User's Manual for information on interfacing to the GPIO's inputs and outputs.

The block diagram for this channel selection is shown in Figure 4-21.

Phone Channel Operation

The Phone Channel profile is used, naturally, for telephone hybrid audio sources. Each Phone Channel has its own discrete Feed-to-Source mix-minus output.

Each Element can accommodate as many unique, automatically-generated mix-minus feeds as it has faders. If the Feed-to-Source mode is set to "Auto" (the most common option - see Chapter Two, "Configuring Inputs," for the section entitled "Source Profile Options"), phone callers hear the output of the **Program-1** bus, minus himself, when the channels they're assigned to are **On**. When the caller's channel is **Off**, their channel's **Talkback** key is lit and they hear the Board Operator's mic routed to their **Talkback** feed.

When a Phone Channel has been assigned to an Element fader, the Status Symbol display located above the channel name will display an icon to show the board operator what is being fed to the caller.

Phone feed-to-source outputs can also be locked to backfeed only **PGM-1**, **PGM-2**, **PGM-3** or **PGM-4**, using the options provided when constructing the Source Profile for your hybrid. This automatic switching mode makes it very easy to prep callers and then seamlessly put them on the air with a minimum of button pushes. (See Chapter Two).

Channels configured for Phone sources can be turned on and off remotely, and will provide on/off status to a remote logic device. **Remote Preview** is also provided,

as are lamp drivers for **On**, **Off** and **Preview**. **Start** and **Stop** pulses can be sent when the channel is turned on or off, respectively. See the *GPIO User's Manual* for information interfacing to GPIO inputs and outputs.

The block diagram for this channel selection is shown in Figure 4-22.

Codec Channel Operation

The Codec Channel is used for codec sources. Like the Phone Channel, each codec source is brought up on its own channel, and an individual Feed-to-Source (mix minus) output is provided for each codec source.

Codec Channels can be turned on and off remotely and will provide on/off status to a remote logic device. **Remote Talk** and **Remote Mute** are also provided, allowing remote talent to take control using their own On/Off/Mute/Talk panels.

Talk enables remote talent to communicate with the board op using the **Talkback** bus. When **Talk** is activated, all outputs for that channel are muted and the source audio is fed to the **Talkback** bus. While **Talk** is active, the channel's **Talkback** key lights, and its Status Symbol will display the word **TALK**. When the user stops talking, the channel returns to normal, and its **Talkback** key flashes for three seconds, alerting the board operator to the guest who spoke to him, and enabling him to reply using the **Talkback** key.

When the remote mute function is activated, the Status Symbol displays **MUTE**, and all the channel outputs are muted until the command is released.

The mix-minus output normally feeds **Program-1** audio to the remote, but can be switched to **Program-2**. The Feed-to-Source output on a Codec Channel is a dual mono output, occupying both sides of the stereo pair. The Left output is normally used for remote talent's headphone feed, and is interrupted by **Talkback** audio when the board op talks to the remote. The Right output sends uninterrupted program audio for use as a PA feed.

The block diagram for this channel selection is shown in Figure 4-23.

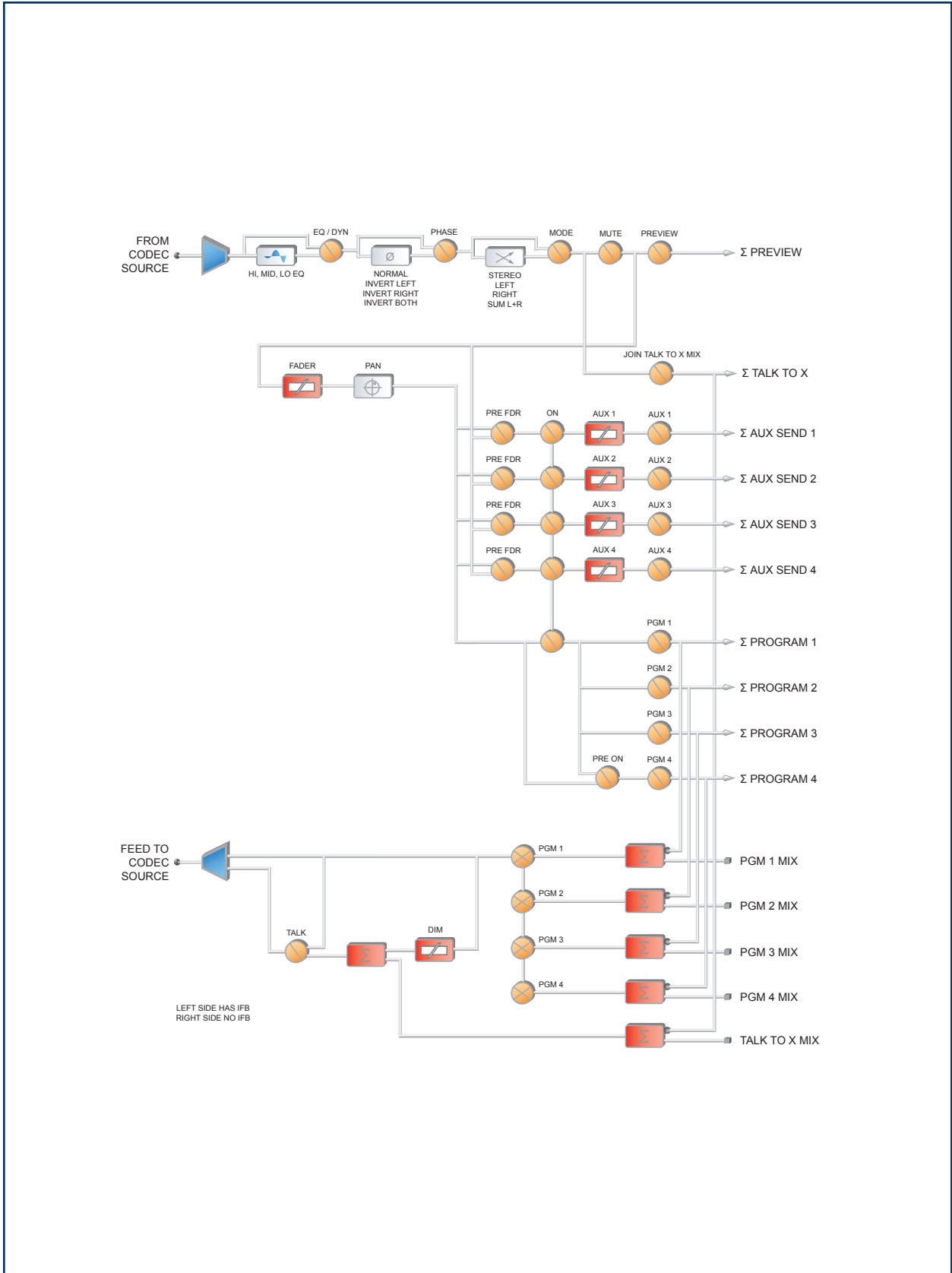


Figure 4-23: Codec channel block

What's Next

One of Element's most powerful features is the ability to instantly reconfigure the entire surface for different show types. Join us in Chapter Four to learn how to set up and use Show Profiles. [🔗](#)

Chapter Four:

Element Show Profiles

If you've already read Chapter Two, "Configuring Inputs," you already have a good understanding of how Source Profiles are essential to Element operation.

In the same way that Source Profiles allow you to pre-determine how an individual channel is configured when a source is loaded, Show Profiles let you build configuration files that can be loaded to determine how the entire Element behaves, and which sources are loaded, and what channels they appear on.

By assembling sets of Source Profiles, each user can have the board set up his or her favorite way — sources placed where they're most useful, monitors set to the appropriate feed, headphones conforming to personal preference. Or, you can use Source Profiles to define different types of broadcasts — one for the morning show, one for talk segments, one for musical guest interviews, one for unattended operation — that need only to be loaded to instantly reconfigure the board for use with a different situation.

Creating A Show Profile

Build A Show

Element's Web interface, which you've used to construct Source Profiles, is also used to help build Show Profiles, so make sure a computer with a Web browser is connected to your Livewire network within easy reach.

Your Element is shipped from the factory with a default Show Profile. This loads the first time you turn on your Element; it provides the "blank slate" you'll use to construct custom Show Profiles. You've no doubt been exploring your Element, so you might want to reload the Default Show Profile using **Show Profiles** key located on Element's Master Module to make sure you're in

"ready mode."

The first step in building a Show Profile is to set up Element using the Channel Options, Monitor Options and other board options as described in Chapter Four, "Element Operations." Get started by assigning a source to each Element channel using the **Options** keys at the top of each channel.

Reminder: You can speed the setup process by "Channel Jumping" while using the Options menus.

For example, to quickly assign sources, select the **Options** key on Channel 1 and choose **Source** from the Channel Options menu. Use the Soft Knobs, Scrub Wheel or Nav Keys to select a source from the list; "take" the source and then press the **Options** key on Channel 2. You'll see the indicator at the top of the Channel Options screen change to read **Channel 02** to confirm that you're now programming Channel 2.

Repeat the procedure until you've programmed all your channels. You can now choose other Channel Options to configure using the same method.

After you've assigned a source for each channel, you can configure the rest of the Channel Options:

- Equalization,
- Pan / Balance,
- Aux Sends, and
- Feed to Source.

When you've configured all Channel Options, use the Monitor Options menu to set Control Room monitor and headphone preferences, the Studio Options menu for Studio monitor settings, and finally the Control Options menu if you need to adjust Aux Send / Aux Return settings.

When you've got the board configured to your satisfaction, proceed to the next section.

Capture It!

It's now time to save your Element configuration. You'll need your Engineer to help with this part.

Enter Element’s IP address in your Web browser. When the Main Menu appears, choose “Configuration” under the “Sources & Profiles” heading on the sidebar menu. This is where you’ll come whenever you need to capture, construct or administer a Show Profile.

From the main menu, choose Show Profiles. You’ll then find four options:

- » **Default Show Profile.** This is the “starter” profile that comes with Element. If you wish, you can overwrite this with your preferred default options; otherwise, consider it as “return to zero” configuration option, for use in returning Element to a “blank slate” condition.
- » **New Show Profile.** Choose this option when you want to construct an entire Show Profile completely from scratch.
- » **Capture Show Profile.** Takes a “snapshot” of the current board state and saves it for recall. This is the option we’ll use in just a moment.
- » **Sort by Time.** Sorts saved Show Profiles in order of creation date / time.

Since you’ve already set up your Element, all that’s left is to capture your settings for further use. Choose **Capture Show Profile** from the menu. You’ll be prompted to give your new profile a unique name; after you do, it’ll be added to the list of available profiles. Voila! You’ve just created and saved a Show Profile, which can now be recalled for use whenever you wish.

However, there may be a few more options you’d like to tweak — options not accessible from Element itself. Let’s go a little deeper into Show Profiles to find out exactly what’s available to you.

Suggestion: As part of Element training, show your talent how to use the Channel Options and Monitor Options menus (discussed in Chapter Four) to assign sources and set monitor options. Then, let them set up the board exactly the way

they want it, then use the Capture command to save their work. This way, talent gets the satisfaction of personalizing their own Element configuration — and saves you the work!

(Sneaky, aren’t we?)

Show Profile Options

Although the ability to create Show Profiles using Element’s “capture” function is powerful, there are more options available exclusively via the Element Web interface.

In the last section, we showed you how to capture the current configuration of your Element, easily creating a Show Profile, which was immediately added to the Show Profiles list.

Now, click on that newly created profile. It’s probably named something like “Capture 2004-2-12 0:800” — the date and time the capture was made.

You’ll see the menu shown in Figure 5-1. Notice that there is an entry for each of Element’s installed channels, as well as a link for **Auxiliary Send/Return Data**, **Monitor Section Data**, and **Record Mode**.

The Channel Description Screen

Click on any of the **Channel** links to examine the settings for a fader. Many of the options will seem familiar to you; you’ve seen them before, during Source Profile setup.

Let’s look at these options one-by-one; notice as we go that the fields match the choices you made for this channel, using Element’s **Options** menus, prior to capturing the Show Profile.

Note: in addition to “active” options – choices that are actively made to Element settings

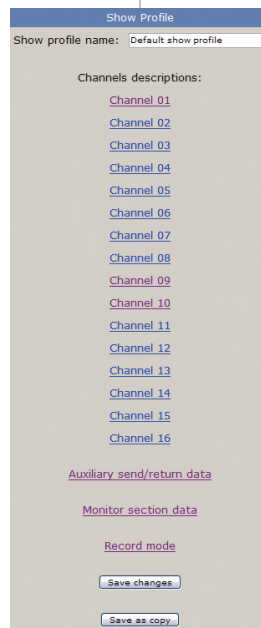


Figure 5-1: Show Profile Settings Menu

when loading a new Show Profile – there is also a “passive” option for many items: **Retain Source Setting**. Choosing **Retain** allows the setting for this option to “cascade” from the Source Profile attached to the source loaded on this channel.

For example, you’ve set a Show Profile to load the Control Room Host mic on Channel 1. You specified an EQ boost of 2 dB @ 1200 Hz in the Source Profile for the Control Room Host mic; checking the **Retain Source Setting** option here in the Channel Description Screen allows that previously-defined EQ setting to “ride along” when the source is loaded – without your having to specify it again.

- **Source ID:** The name of the source you assigned to this fader. You can change the source using the drop-down box.
- **Feed To Source Mode:** Specify **PGM-1**, **PGM-2**, **PGM-3** or **PGM-4** or **AUTO mix-minus** for this source, or **Retain Source Setting**. Choosing **Auto** for any phone source feeds the caller **PGM-1 minus** when the fader is **On**, and **Talkback** when the fader is **Off**.
- **Auto-Start Timer:** Determines whether turning this fader on will start/reset the Event Timer. You can **Enable**, **Disable** or **Retain Source Setting**.
- **Signal Mode:** Choose from **Stereo**, **Left** (left channel fed to both sides), **Right** (right channel fed to both sides), **Sum** (sum L+R to mono and feed to both sides) or **Retain Source Setting**.
- **Signal Mode Locked:** **Enable** or **Disable** the user’s ability to change the Signal Mode, or **Retain Source Setting**.
- **Panorama Position:** Trim Pan/Balance for this source, or **Retain Source Setting**.
- **Phase:** Adjust for phase errors. May be set to **Normal**, **Invert Left**, **Invert Right**, **Invert Left And Right**, or **Retain Source Setting**.
- **EQ Active:** Set source EQ for **Bypass**, **Active** or **Retain Source Setting**.

- **EQ High Mode:** Switches high-frequency EQ mode between **Shelf** and **Bandpass**, or **Retain Source Setting**.
- **EQ High Frequency:** You can specify the top of the High Frequency Shelf, or the middle frequency of the Bandpass filter (depending upon which mode was chosen for EQ High Mode), or **Retain Source Setting**.
- **EQ Mid Frequency / EQ Low Frequency:** Choose the middle frequency of the Low and Middle EQ bands, or Retain Source Setting.
- **EQ High Gain / EQ Mid Gain / EQ Low Gain:** dial in boost or cut for the three EQ bands, or **Retain Source Setting**.

Note: For details on Element EQ operations, please refer to Chapter Two, “Working With Sources.”

- **Assign to PGM1 / PGM 2 / PGM3 / PGM4 / AUX A / AUX B / AUX C / AUX D:** Checking **ON** for any of these buses assigns the channel to that bus when the Show Profile is loaded. You can also choose **OFF** or **Retain Source Settings**.
- **AUX A / AUX B / AUX C / AUX D Pre/Post Fader:** Allows you to send this channel to the Aux Send 1 bus either **Pre-Fader**, **Post-Fader** or **Retain Source Settings**.
- **AUX SEND A / AUX SEND B / AUX SEND C / AUX SEND D Gain:** Turn this channel’s feed to the AUX1 bus **Off**, specify boost or cut, or **Retain Source Settings**.
- **Channel On/Off Status:** Allows you to turn this channel **On** or **Off** when the Show Profile is loaded.
- **Control Lock Map:** Placing a check mark in any of these boxes “locks” the selections for this channel’s bus assignments, On/Off state and fader position.

About Channel Lock and Fader Operation:

Say that your station runs syndicated programming overnight, and you want to eliminate the possibility that the feed could be inadvertently disrupted. Choose **Lock** for the channel the feed is assigned to, and the channel cannot be turned off, or bus assignment changed, until the channel is unlocked.

Note that there is currently no option to allow presetting the fader gain. When you load a Show Profile which specifies a **Locked** fader, the gain value is locked to the position of the fader at the moment the Show Profile is loaded.

When you **Unlock** the fader using the Channel Options menu, the fader will resume working normally from the current position. You can move it and lock again in a different position if you wish.

If the fader has been moved, the first touch after **Unlocking** immediately resets the gain to the fader's current position.

- **Save Changes** button applies any changes made to the current Channel Description.
- **Back to Show Profile** link takes you back to the Show Profile Settings menu.

The Auxiliary Send Description Screen

In the same way that the Show Profile Channel Description screens mirror Element's **Channel Options** menus, the Auxiliary Send Description Screen captures the global Aux Send settings that can be set using the **Global Options Sends & Returns Key** located on the Master Module.

Note: Instead of the **Retain Source Settings** option found in the Channel Descriptions pages, the Auxiliary Send Descriptions Screen features a new option: **Retain Existing Settings**. Using this option with any of the Aux Send/Return settings allows the Show Profile to be loaded without disturbing the Aux Send/Return master settings in use by the board operator.

From the Show Profile Settings web page, choose the **Auxiliary Send/Return Data** link. You'll find the following options:

- **Aux Send A / B / C / D Master Gain:** Selecting

Off turns off master gain adjustment for each **Aux Send's** outbound feed. Select **Use** and enter a gain value between -25 dB and 10 db to trim overall gain if desired, or select **Retain Existing Setting**.

- **Aux Send A / B / C / D On/Off Status:** Choose **Off** to turn the selected Aux Send 1 off completely, **On** to turn it on, or **Retain Existing Setting**.
- **Aux Return 1 / 2 Master Gain:** Selecting **Off** turns off master gain adjustment for the selected Aux Return bus. Select **Use** and enter a gain value between -25 dB and 10 db to trim overall return gain if desired, or select **Retain Existing Setting**.
- **Aux Return 1 / 2 On/Off Status:** Choose **Off** to turn the selected Aux Return off completely, **On** to turn it on, or **Retain Existing Setting**.
- **Aux Return 1 / 2 Signal Mode:** Choose from **Stereo, Left** (left channel fed to both sides), **Right** (right channel fed to both sides), **Sum** (sum L+R to mono and feed to both sides) or **Retain Existing Setting**.
- **Aux Return 1 / 2 Panorama Setting:** Trim Pan/Balance for the selected Aux Return, or **Retain Existing Setting**.
- **Aux Return 1 / 2 Assign to PGM1 / PGM2 / PGM3 / PGM4:** Choose **On** to fold the selected Aux Return into any of the four PGM buses, or choose **Retain Existing Setting**.
- **Aux 1 / 2 Source ID:** If this drop-down box looks familiar, that's because it's just like the ones you used earlier to construct Source Profiles and map I/O to Axia Audio Nodes. Click in the box and select the audio device that will be providing source audio for the Aux 1 or Aux 2 Return.
- **Save Changes** button applies any changes made to the Auxiliary Send Description and returns you to the Show Profile Settings menu.

- **Back to Show Profile** link takes you back to Show Profile Settings without saving changes.

Monitor Section Data Screen

The Monitor Section Data screen captures the Monitor settings found in Element’s **Monitor Options** menus. Since you’re working on a “captured” Show Profile, this page will, of course, mirror the choices you made when you set up Element in the beginning of this chapter.

From the Show Profile Settings web page, choose the **Monitor Section Data** link (refer to Figure 5-4). You’ll find the page divided into four sections: general options, Control Room Monitor options, Control Room Headphone options, and Studio Monitor options.

General Monitor Options

- **Scrub Volume Mode:** Assign the Scrub Wheel to control **Control Room Monitor** volume or **Control Room Headphone** volume, choose **Scrub Volume Off** to have the Scrub Wheel assigned to nothing, or choose **Retain Existing Setting**.
- **Preview Interlock Mode:** If **Enable** is chosen, pressing any fader’s **Preview** key causes any other, previously-lit **Preview** key to be extinguished, removing that source from **Preview**. The board operator can assign multiple faders to **Preview** by pushing and holding any lit **Preview** key while pushing other **Preview** keys to add or subtract.

If **Disable** is chosen, each **Preview** key works like a press-on, press-off latching switch. Multiple faders can be assigned to **Preview** simply by pushing once; a second push cancels any individual **Preview** assignment.

- **Switched Meter Source Select:** Choose whether the far-right on-screen meter bars will display levels for the **PGM-4** bus, the source feeding the Control Room Monitors (**Monitor Section**) or **Retain Existing Setting**.
- **Preview Speaker Master Gain:** Lets you **Use** up

to -85 dB of attenuation on the Control Room Preview speakers, turn gain adjustment **Off**, or **Retain Existing Setting**.

- **Source ID for EXT1 / EXT2 Preview:** This drop-down box lets you specify the source that will be fed to the monitors when **External 1** or **External 2** is selected by talent for monitoring (typically an off-air monitor or a dedicated headphone processing loop).
- **Talkback Gain:** **Use** up to -30 dB attenuation or 10 dB gain on the contents of the **Talkback** bus. You can also turn gain adjustment **Off** or **Retain Existing Setting**.
- **GPIO Channel For Profanity Delay:** Enter the Channel Number of the GPIO port connected to your station’s Profanity Delay Unit. This assigns the **Delay** keys on the Element Master Module to control your external delay unit.

Control Room Monitor Options

- **Monitor Assignment:** Choose to load **PGM1, PGM2, PGM3, PGM4, AUX A, AUX B, AUX C, AUX D** or an **External** source to the Monitor 1 speakers when the Show Profile is loaded. You can also choose **No Source** if you wish the Profile to be loaded with nothing assigned to Monitor 1.
- **Source ID for EXT 1 / EXT 2 Input:** Use these drop-down boxes to choose what source will be auditioned when **External 1** or **External 2** is chosen from the Monitor Assignment list above.

Note: The word “External” is used to refer to any source not generated by the studio you’re working in. When a source is assigned to an **External** input, that source will appear in Element’s Monitor Selection menus.

- **CR Monitor Master Gain:** Allows you to **Use** as much as -85 dB of gain reduction on the audio signal sent to the Monitor 1 Speakers. You can also turn gain control **Off** or **Retain Existing Setting**.
- **Signal Mode: CR Monitor** lets you set the audio

feed to the Monitor 1 speakers to **Stereo, Left, Right, Sum** or **Retain Existing Setting**.

- **CR Monitor Dim Gain:** Specify the amount of volume by which the Monitor 1 speakers will dim when **Talk To...** or **Preview** is in use. **Use** as much as -30 dB of cut, turn the option **Off**, or **Retain Existing Setting**.
- **GPIO Channel for CR Monitor:** In this box, you assign Monitor 1 a channel number, effectively making it a “source” to which logic commands – to activate Mute, Dim, tally lights – can be associated. You’ll enter this channel number in the GPIO Setup web page to assign a GPIO port for these logic functions. See Chapter Three, “GPIO Configuration,” for more information.

Control Room Headphone Options

- **Headphone Assignment:** Choose to load **PGM1, PGM2, PGM3, PGM4, AUX A, AUX B, AUX C, AUX D** or an **External** source to the Control Room headphone channel when the Show Profile is loaded. You can also choose **No Source** if you wish the Profile to be loaded with nothing assigned to the board op’s headphones.
- **Source ID for EXT 1 / EXT 2 Input:** Use these drop-down boxes to choose what source will be auditioned when **External 1** or **External 2** is chosen from the headphone assignment list above.
- **CR Headphone Master Gain:** Allows you to **Use** as much as 85 dB of gain reduction on the audio signal sent to the Control Room headphones. You can also turn gain control **Off** or **Retain Existing Setting**.
- **Signal Mode: CR Headphone:** Lets you set the audio feed to the Control Room headphones to **Stereo, Left, Right, Sum** or **Retain Existing Setting**.
- **CR Headphone Independent:** Lets you choose whether, for this Show Profile, the Control Room headphones should be fed the same audio as the Monitor 1 Speakers (**Follow Monitors**) or if the board op is allowed to select a source for headphones independent of the Monitor Speakers (**Use Headphone Source Select**). You may also **Retain Existing Setting**.
- **Preview-in-Headphone Mode:** Choices are **Off, Normal, Auto** and **Retain Existing Setting**. If set to **Off**, sources assigned to the Preview bus will only be heard in the Control Room Preview speakers, leaving the board operator’s headphones to continuously monitor the selected source. In **Normal** mode, audio from the Preview bus will feed the board op’s headphones, *in stereo*, whenever **Preview** is active, interrupting the selected source. **Auto** mode sums Preview audio to mono and feeds it to the *right headphone*; the selected source audio is also summed to mono and fed to the *left headphone*.
- **CR Headphone EQ Active:** Set headphone EQ for **Bypass, Active** or **Retain Existing Setting**.
- **CR Headphone EQ High Mode:** Switches high-frequency EQ model between **Shelf** and **Bandpass**, or **Retain Existing Setting**.
- **CR Headphone EQ High Frequency:** You can specify the top of the High Frequency Shelf, or the middle frequency of the Bandpass filter (depending upon which model was chosen for EQ High Mode), or **Retain Existing Setting**.
- **CR Headphone EQ Mid Frequency / EQ Low Frequency:** Choose the middle frequency of the Low and Middle EQ bands, or **Retain Existing Setting**.
- **CR Headphone EQ High Gain / EQ Mid Gain / EQ Low Gain:** dial in boost or cut for the three EQ bands, or **Retain Existing Setting**.

Studio Monitor Options

- **Monitor Assignment:** Choose to load **PGM1, PGM2, PGM3, PGM4, AUX A, AUX B, AUX C, AUX D** or an **External** source to the Monitor 2 channel

when the Show Profile is loaded. You can also choose **No Source** if you wish the Profile to be loaded with nothing assigned to the Control Room monitor.

- **Source ID for EXT 1 / EXT 2 Input:** Use these drop-down boxes to choose what source will be auditioned when **External 1** or **External 2** is chosen from the Studio 2 assignment list.
- **Studio Monitor Master Gain:** Allows you to **Use** as much as -85 dB of gain reduction on the audio signal sent to the Monitor 2 Speakers. You can also turn gain control **Off** or **Retain Existing Setting**.
- **Studio Monitor Dim Gain:** Specify the amount of volume by which the Monitor 2 speakers will dim when **Talk To...** or **Preview** is in use. **Use** as much as -30 dB of cut, turn the option **Off**, or **Retain Existing Setting**.
- **GPIO Channel for Studio Monitor:** In this box, you assign Monitor 2 a channel number, effectively making it a “source” to which logic commands can be associated. You’ll enter this channel number in the GPIO Setup web page to assign a GPIO port for these logic functions. See Chapter Three, “GPIO Configuration,” for more information.

Save and Exit

- **Save Changes** button applies any changes made to the Auxiliary Send Description and returns you to the Show Profile Settings menu.
- **Back to Show Profile** link takes you back to Show Profile Settings without saving changes.

Record Mode

Element’s Record Mode (described in Chapter Four) acts as a one-button “macro,” assigning channels to the Record bus, assigning monitor feeds, and starting a recording device.

However, unlike most other Element options, Record Mode cannot be configured from the surface itself. You

must define Record Mode here, in the Show Profiles web page.

Like the Monitor Section Data page just described, the Record Mode page is divided into sections. Click the **Record Mode** link from the Show Profile Settings screen to access these Record Mode definitions:

Record Mode Configuration

- **Record Mode Activation:** If **Disabled** is chosen, the board op will not be able to use **Record Mode** while this Show Profile is loaded. Choose **Enabled** to allow Record Mode to be used.
- **GPIO Channel for Recorder Control:** Enter the channel number of the GPIO port assigned to your chosen recording device. This device will automatically begin recording with the board op presses Element’s **Record Mode Engage** key. See Chapter Three, “Configuring GPIO,” for a complete reference of Element GPIO functions.
- **GPIO Channel for Hybrid Control:** Enter the channel number of the GPIO port assigned to your telephone hybrid, if you want **Record Mode** activation to control your hybrid’s **Answer** and **Drop** functions, among others. Refer to Chapter Three for a complete reference of Element GPIO functions.

Mixing

Fader input	Source	Assign to record	Assign to preview	Mute	source input
9	Sony DVD	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
10	Hybrid	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
11	CD 3	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
12	CD 4	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

Figure 5-2: Record Mode Mixing section

- This section lists every source assigned to a fader in this Show Profile. Placing checkmarks in the appropriate boxes lets you instantly auto-assign your choice of sources to the **Record** and/or **Preview** buses when the **Record Mode** key is pressed. You may also choose to **Mute Source Input** to prevent a source from being recorded.

CR Monitor

- **CR Monitor Source:** Choose to load **PGM1**, **PGM2**,

PGM3, PGM4, AUX A, AUX B, AUX C, AUX D or an **External Source** to the Monitor 1 speakers when **Record Mode** is engaged. You can also choose **No Source** and **Retain Existing Source**. Monitor 1 will be returned to its previous selection when **Record Mode** is disengaged.

- **CR Monitor External Source ID:** Use this drop-down box to choose what source will be auditioned when **External** is chosen from the CR Monitor Assignment list above.

CR Headphones

- **CR Headphone Source:** Choose to load **PGM1, PGM2, PGM3, PGM4, AUX A, AUX B, AUX C, AUX D** or an **External** source to the Control Room headphone channel when **Record Mode** is engaged. You can also choose **No Source** if you wish **Record Mode** to be loaded with nothing assigned to the board op's headphones.
- **CR Headphone External Source ID:** Use this drop-down box to choose what source will be auditioned when **External** is chosen from the CR Headphone Assignment list above.

Studio Monitor

- **Studio Monitor Source:** Choose to load **PGM1, PGM2, PGM3, PGM4, AUX A, AUX B, AUX C, AUX D** or an **External Source** to the Monitor 2 speakers when **Record Mode** is engaged. You can also choose **No Source** and **Retain Existing Source**. Monitor 2 will be returned to its previous selection when **Record Mode** is disengaged.
- **Studio Monitor External Source ID:** Use this drop-down box to choose what source will be auditioned when **External** is chosen from the Studio Monitor Assignment list above.

Save and Exit:

- **Save Changes** button applies any changes made to the Auxiliary Send Description and returns you to the Show Profile Settings menu.

- **Back to Show Profile** link takes you back to Show Profile Settings without saving changes.

What's Next

By now you should have a pretty thorough understanding of your Element control surface. Next, we'll look at the Axia Audio Nodes in Chapter 5. [🔗](#)

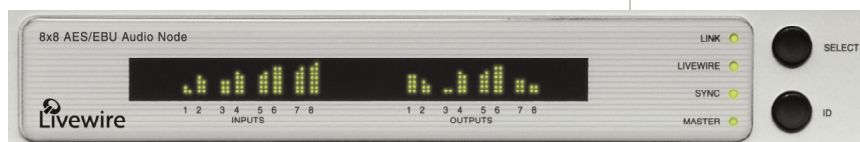
Chapter Five:

Introducing the Axia Audio Nodes

This section will help you understand the function and displays of the the Axia audio nodes you'll encounter in your Axia system. These displays and controls are for information only and should not be altered unless given specific instructions to do so by your engineer.

Axia Analog and AES/EBU Nodes

The Axia Livewire 8x8 Audio Node has eight stereo inputs, and eight stereo outputs. 8x8 Analog nodes have balanced analog inputs and outputs whereas the 8x8



AES nodes have AES3 inputs and outputs. Therefore, each 8x8 Node can create 8 stereo Livewire streams, each of which becomes available to other devices on the Livewire network.

Each stereo output can be assigned to output a stereo Livewire stream acquired from the network.

Basic point-to-point (e.g. “Livewire Snake”) applications require only two Livewire nodes and a category 6 “Crossover Ethernet cable”. More sophisticated multipoint networks can be used by connecting multiple Livewire nodes to an appropriate Ethernet Switch.

NOTE: Only an approved, and properly programmed Ethernet switch, incorporating the proper Multicast and QoS standards may be used.

Front Panel Controls and Indicators

The Livewire 8x8 Audio Node incorporates a number of front panel indicators to allow the operator to verify proper operation quickly and confidently.

Status LED indicators

Four LEDs indicate the status of the Livewire and Ethernet connections, as well as system synchronisation as follows:

- » LINK – When illuminated continuously, this LED represents the presence of a live Ethernet link to another Ethernet 100 Base-T device. This LED indicates that a connection is present and some device is connected. It does not indicate the quality of the connection however. If no Ethernet link is present, this will flash slowly.
- » LIVEWIRE – This LED indicates that the connected Ethernet segment has Livewire traffic present. If the link LED is illuminated, and the LIVEWIRE LED fails to illuminate, there are either no other Livewire devices connected, or the Ethernet switch has not been programmed to pass such traffic to the port to which this node is connected.

SYNC & MASTER

Only one of these two LEDs should be illuminated, with the one exception noted below. If neither LED illuminates, something is not correct. The SYNC LED indicates the receipt of clock information from another (Master) Livewire Node. The MASTER LED indicates that this node is acting as the master clock source for the Livewire network. More specifically:

- » SYNC – If Sync packets are being received by the Livewire node, this LED will begin to flash. The LED will continue to flash until the Livewire node has locked its local clock to the network master. Once the local node's PLL is locked, the LED will illuminate solidly.

- » MASTER – The Livewire system employs a sophisticated master/slave clocking system over the Ethernet network. By default any device may become the clock master, however this can be changed if desired. The system has the ability to automatically change to a different clock master should the current master become disconnected, or otherwise inoperable. This happens transparently without audio glitches. This LED indicates that this node is currently acting as MASTER.

SYNC & MASTER with the 8x8 AES Node

When the AES Node is used in the default configuration the SYNC and MASTER LEDs operate as described above. However, if the AES sync input as Livewire master timebase is set to YES from the QOS web pages (see section 3, Advanced Programming) these lights will operate differently, as described here:

- » SYNC – If a valid AES signal received by the AES input designated as AES sync source in the QOS web page, this LED will begin to flash. The LED will continue to flash until the Livewire node has locked its local clock to the AES input signal. Once the local node's PLL is locked, the LED will illuminate solidly.
- » MASTER – This LED indicates that this node is currently acting as MASTER clock source for the Livewire network.

The Livewire 8x8 nodes have a bargraph meter for each input and each output.

Bargraph LED Meters

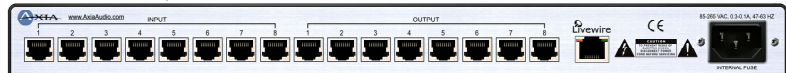
- » INPUT (Transmit to network) METERS – The left eight meter-pairs represent the left and right channels of each input. The meters are continuously active, and indicate that audio is present at the associated input. The lowest LED segment will illuminate at a signal level of -42 dBfs. The top-most LED segment represents a level of 0

dBfs – this segment should not illuminate except in cases of extreme input overload.

- » OUTPUT (Receive from network) METERS – The rightmost 8 LED meter pairs represent the left and right channels of each audio stream being received. Each meter is associated with the audio on the corresponding output. The meter calibration is the same as for the input meters.

The lowest Output LED segment has a special meaning. These LEDs will be illuminated whenever the stream designated for that channel is present, and the Livewire node is locked to that stream. Therefore, the lowest LED segment will be illuminated during silence, to indicate that the associated Livewire stream is correctly being received.

Rear Panel



The rear panel of both the 8x8 Analog Node and the 8x8 AES Node are essentially identical and are pictured above.

AC (Mains) Power

The AC receptacle connects mains power to the unit with a standard IEC (International Electrotechnical Committee) power cord. The power supply has a “universal” AC input, accepting a range from 85 to 265 VAC, 47-63 Hz.

Livewire (100 Base-T) Connector

This connector is for connection to another Livewire node, or an approved Ethernet switch. It has two integral LEDs. The green “Link” LED indicates the presence of a live signal (same as the front panel “Link” LED). The “Activity” LED indicates that Ethernet packets are being sent or received over the link.

Input Connectors

Each pair of audio inputs share a 8-position / 8-pin

miniature modular jack (e.g. RJ45 style). Each jack functions as a single stereo input or two mono inputs.

Output Connectors

Each pair of audio outputs share a 8-position / 8 pin miniature modular jack (e.g. RJ45 style). As with the input connectors, each jack functions as a single stereo or dual mono output.

Checking the Livewire Node Name

With the meter screen displayed, press SELECT twice. The name of this Livewire node will be displayed.

Checking the Software Version

With the meter screen displayed, press SELECT three times. The software version will be displayed.

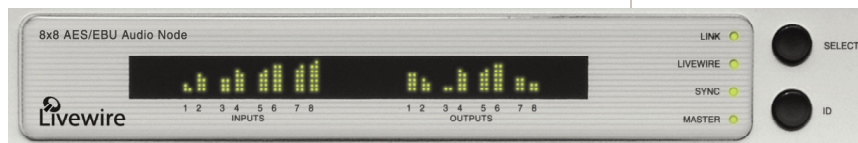
Axia Microphone Node

The Axia Microphone Node has eight microphone-level mono inputs, and eight stereo outputs. Each Microphone Node can create 8 mono Livewire™ streams, each of which becomes available to other devices on the Livewire™ network.

Each of the Microphone Node's stereo outputs can be assigned to output a stereo Livewire™ stream acquired from the network.

Front Panel Controls and Indicators

The Livewire Microphone node is similar in operation to the 8x8 Analog and 8x8 AES Nodes. These all incorporate a number of front panel indicators to allow the operator to verify proper operation quickly and confidently.



Status LED indicators

Four LEDs indicate the status of the Livewire™ and Ethernet connections, as well as system synchronisation as follows:

- » **LINK** – When illuminated continuously, this LED represents the presence of a live Ethernet link to another Ethernet 100 Base-T device. This LED indicates that a connection is present and some device is connected. It does not indicate the quality of the connection however. If no Ethernet link is present, this will flash slowly.

LIVEWIRE

This LED indicates that the connected Ethernet segment has Livewire™ traffic present. If the link LED is illuminated, and the LIVEWIRE LED fails to illuminate, there are either no other Livewire™ devices connected, or the Ethernet switch has not been programmed to pass such traffic to the port to which this node is connected.

SYNC & MASTER

Only one of these two LEDs should be illuminated. If neither LED illuminates, something is not correct. The SYNC LED indicates the receipt of clock information from another (Master) Livewire Node. The MASTER LED indicates that this node is acting as the master clock source for the Livewire network. More specifically:

- » **SYNC** – If Sync packets are being received by the Livewire™ node, this LED will begin to flash. The LED will continue to flash until the Livewire™ node has locked its local clock to the network master. Once the local node's PLL is locked, the LED will illuminate solidly.
- » **MASTER** – The Livewire™ system employs a sophisticated master/slave clocking system over the Ethernet network. By default any device may become the clock master, however this can be changed if desired. The system has the ability to automatically change to a different clock master should the current master become disconnected, or otherwise inoperable. This happens transparently without audio glitches. This LED indicates that this node is currently acting as MASTER.

Bargraph LED meters

The livewire Microphone Node has a bargraph meter for each input and each output.

- » **INPUT (Transmit to network) METERS** – The left eight meters indicate the levels for each input. The meters are continuously active, and indicate that audio is present at the associated input. The lowest LED segment will illuminate at a signal level of -42 dBfs. The top-most LED segment represents a level of 0 dBfs – this segment should not illuminate except in cases of extreme input overload.

OUTPUT (Receive from network) METERS

The rightmost 8 LED meter pairs represent the left and right channels of each audio stream being received.

Each meter is associated with the audio output on the corresponding audio output. The meter calibration is the same as for the input meters.

The lowest Output LED segment has a special meaning. These LEDs will be illuminated whenever the stream designated for that channel is present, and the Livewire node is locked to that stream. Therefore, the lowest LED segment will be illuminated during silence, to indicate that the associated Livewire stream is correctly being received.

Rear Panel

AC (Mains) Power

The AC receptacle connects mains power to the unit with a standard IEC (International Electrotechnical Committee) power cord. The power supply has a “universal” AC input, accepting a range from 85 to 265 VAC, 47-63 Hz.



Livewire (100 Base-T) Connector

This connector is for connection to another Livewire node, or an approved Ethernet switch. It has two integral LEDs. The green “Link” LED indicates the presence of a live signal (same as the front panel “Link” LED). The “Activity” LED indicates that Ethernet packets are being sent or received over the link.

Input Connectors

Each input is on a female XLR-3 connector. 48 Volt phantom power is available and is enabled through a configuration web page.

Output Connectors

Each pair of audio outputs share an 8-position / 8 pin miniature modular jack (e.g. RJ45 style). The connector pin functions are the same as for both the 8x8 Analog node.

Checking the Livewire Node Name

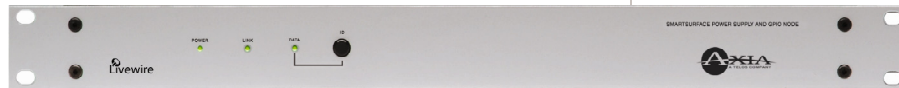
With the meter screen displayed, press SELECT twice. The name of this Livewire node will be displayed.

Checking the Software Version

With the meter screen displayed, press SELECT three times. The software version will be displayed.

Axia GPIO Node

The GPIO Node provides the ability for logic (closure) inputs and outputs to be sent via Livewire streams. The GPIO node has 8 groups of inputs and outputs. Each group of Livewire closure inputs and outputs is on a separate connector and each must be assigned a Livewire “channel”. The function of the GPIO Node is to provide 40 optically-isolated “General-Purpose I/O” ports to interface to enable the box to control customer equipment, like CD players, tape decks, etc (40 input and 40 output



ports total). The console or Axia Audio Engine can send commands to the GPIO box over a 100 Base-T Ethernet interface. Using these input/output ports, it can control, or convey the status of, connected equipment.

Front Panel Controls and Indicators

The Livewire GPIO node incorporates a number of front panel indicators to allow the operator to verify proper operation quickly and confidently.

Status LED indicators

Three LEDs indicate the status of the Ethernet connection, as well as system synchronisation as follows:

- » POWER – Indicates that the unit is connected to AC Mains power and operating.
- » LINK – Indicates that the Ethernet LINK is connected correctly, and that Ethernet “Link Active” pulses are being correctly received. This does NOT mean, however, that the unit is transferring any data; simply that the Ethernet link is OK from a hardware point of view. This LED does not indicate the quality of the connection however. If no Ethernet link is present, this will flash slowly.
- » DATA – This indicates several different status states depending on it’s state. The meaning varies on whether the ID button has been pressed in the previous 10 seconds:

- * OFF – Unit is booting up, or the processor has stopped functioning.
- * ON – Processor and system are operating normally.

ID Button

The I.D. button is used as part of the configuration process. When this button is pressed, it triggers this GPIO box to identify itself and request an IP address from a bootp server on the Ethernet network. This button should not be pressed unless instructed to do so by your engineer.

Rear Panel

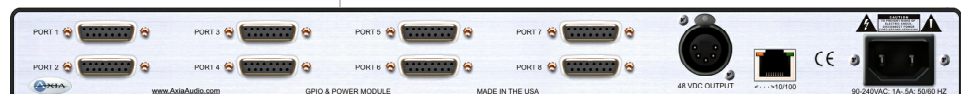
The rear panel of the Router Selector node is pictured below:

AC (Mains) Power

The AC receptacle connects mains power to the unit with a standard IEC (International Electrotechnical Committee) power cord. The power supply has a “universal” AC input, accepting a range from 85 to 265 VAC, 47-63 Hz. A fuse is located inside on the power supply circuit board.

Livewire (100 Base-T) Connector

This connector is for connection to another Livewire node, or an approved Ethernet switch. It has two integral LEDs. The green “Link” LED indicates the presence of a live signal (same as the front panel “Link” LED). The



“Activity” LED indicates that Ethernet packets are being sent or received over the link.

I/O Port Connectors

These are 15-pin D-Sub connectors, used to provide I/O drivers for customer devices. Eight of these connectors are available; each one providing 5 optically-isolated inputs and 5 optically-isolated outputs for interfacing to audio equipment.

Axia Router Selector Node

The Axia Livewire Router Selector Node has one stereo input, and one stereo output. The user can feed audio into either a balanced analog input or an AES input.

The Router selector node has both AES and analog balanced outputs. Both the AES and analog outputs are active simultaneously; both outputs have the same audio present.



Therefore, each Router Selector Node can create 1 stereo Livewire stream, which becomes available to other devices on the Livewire network.

Similarly, the node's stereo outputs can be assigned to output a single stereo Livewire stream acquired from the network.

The stream received can be easily selected from a list of available Livewire streams using the front panel. Eight frequently-used streams can be assigned to the front panel "radio buttons" for instant access.

A front panel headphone jack, LCD display, and control knob round out the front panel. The control knob defaults to the volume control function.

In addition to being a handy audio selector for use in the studio, the Router Selector node can be a handy Livewire tester/monitor node, for use in the Technical Operations Center or rack room.

Front Panel Controls and Indicators

The Livewire Router Selector Node incorporates a number of front panel controls to allow the operator to quickly and confidently select Livewire network audio sources for local playback.

Fast Select Radio Buttons

The 8 buttons on the left side of the node permit instant stream access. These work identically to the channel select buttons present on most radios. The built-in amber LED indicates that a particular button is active and the associated stream is being played. A label area above the buttons can be used to label the buttons, using commonly available thermal printed pressure sensitive labels (3/8 inch).

LCD Display and Control Knob

The built in LCD display and control knob permit most functions of the Router Selector node to be controlled from the front panel.

Volume Control

The control knob defaults to a volume control function for the built-in headphone jack. Turning the knob causes a volume indicator bar to be displayed showing the volume level setting. The volume indicator disappears three seconds after one ceases to change the volume.

Meters

The top section of the default LCD display shows the current Send (local input) and Receive (local output) levels for the left and right channels.

Source Selection

The control knob is also used to select the Livewire stream to be played. Clicking on the control knob before rotating it puts it in selection mode.



Rear Panel

AC (Mains) Power

The AC receptacle connects mains power to the unit with a standard IEC (International Electrotechnical Committee) power cord. The power supply has a "universal" AC input.

Livewire (100 Base-T) Connector

This connector is for connection to another Livewire node, or an approved Ethernet switch. It has two integral LEDs. The green “Link” LED indicates the presence of a live signal. The amber “Activity” LED indicates that Ethernet packets are being sent or received over the link.

Input Connectors

The node will automatically switch between analog and AES inputs; if a valid AES input signal is present, this will be used, otherwise the analog input will be used. Each analog stereo audio input is wired to an 8-position / 8-pin miniature modular jack (e.g. RJ-45 style) as well as two XLR female connectors. The AES input is wired to an 8-position / 8 – pin miniature modular jack as well as an XLR female connector.

Output Connectors

Each stereo analog audio outputs share an 8-position, 8-pin miniature modular jack (e.g. RJ-45 style) and a pair of XLR male connectors. The AES output is also available on both an RJ-45 connector and an XLR male connector.

Powering up

When the Livewire Router Selector node is powered on you should observe the following: The eight Radio buttons should illuminate briefly, and after a short period of time the LCD display will illuminate and display the default screen (showing the meters and the stream currently selected).

Checking the Software Version

With the meter screen displayed, press and hold the control knob for three seconds. The software version will be displayed in the top left corner of the System Menu.

Operation

Operation of the Router Selector from the front panel is quite simple. Streams can be selected, assigned to an instant access Radio Button if desired, and the headphone volume may be adjusted.

Selecting a stream for playback

From the default menu screen click the control knob once. The source selection screen, shown below, will be displayed.

Rotate the control knob to highlight the desired Livewire stream and click to select it.



Streams can be displayed in order by their Livewire channel number, or by their Livewire name. To change the sort criteria rotate the knob clockwise until you reach the end of the menu. You can now highlight and select from the following options:

- Name – clicking this will cause the streams-list to be sorted by stream name.
- Chan – clicking this will cause the streams-list to be sorted by stream Livewire channel number.
- Exit – clicking this will return you to the default menu without selecting a new source.

Assigning streams to radio buttons

Once a stream has been selected and is playing (e.g. by using the control knob and LCD display, see above), it can be assigned to any button by pressing and holding that button. After 3 seconds, the button’s built-in lamp will blink indicating the new setting is being stored.

Adjusting headphone playback volume

Whenever the unit is at the default screen (the meters will be shown at the top of the screen) the headphone volume may be adjusted by rotating the control knob. This adjusts the front-panel headphone levels only; the line level outputs on the rear panel are not affected.

What’s Next

Since Axia networks are computer-based, it’s easy to interface computer audio applications directly to the network. We’ll discuss the IP-Audio Driver next. 🎧

Chapter Six:

Introducing the Axia IP-Audio Driver

Here we'll tell you what the Axia IP-Audio Driver is, and give some examples of typical applications. As a general rule, the IP-Audio Driver will be pre-installed for you by your Engineering staff and will not need to be configured or adjusted by you. This information is supplied so that you can better understand how the technology is employed.

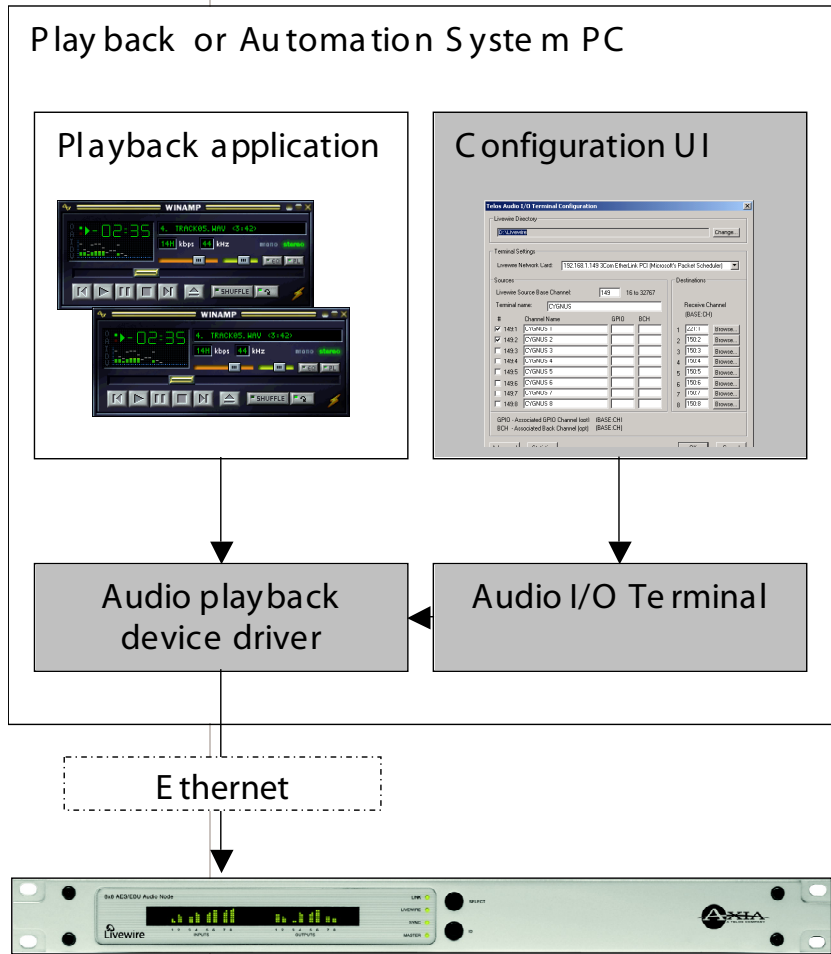
Description

The Axia IP-Audio Driver is the software interface between your PC audio applications and the Livewire network. It provides the following functions:

- Software interface for audio “sources” to be sent to the Livewire network from PC/Windows audio applications, such as delivery systems and other audio players.
- Software interface to receive audio from the Livewire network to “destinations” on the PC/Windows system such as audio editor applications.
- A ‘GPIO’ function to convey “button-press” data from the Livewire network to “destination” applications, such as when a control surface fader start button commands PC/Windows-based audio player to start playback.
- A software interface to allow Livewire streams to be played on standard media players “destinations” such as the Windows Media player.

Axia IP-Audio Driver for Windows allows integrating the Windows multimedia system with an Axia Livewire audio network. There are wide variety of applications where Axia nodes devices and the Windows software are used together.

Axia Nodes can be simply used as high quality remote 8-channel audio input/output devices (Scenario 1.) Or, Livewire technology can completely eliminate audio



input/output devices by simply allowing multiple computers to send audio over the Livewire network (see Scenario 2). In both cases the Axia IP-Audio Driver interfaces through Windows to appear as an audio input/output device(s).

This software package also contains GPIO module that allows controlling any application from your on-air or production-room console.

Scenario 1: Axia node as remote sound card

In this example a remote Axia node acts as a sound card, permitting audio input and output to the computer from the node:

Scenario 2: Automation PC (no sound card required)

In this scenario an Automation System places a

provides all software components needed to receive Livewire streams using standard players, like Windows Media player.

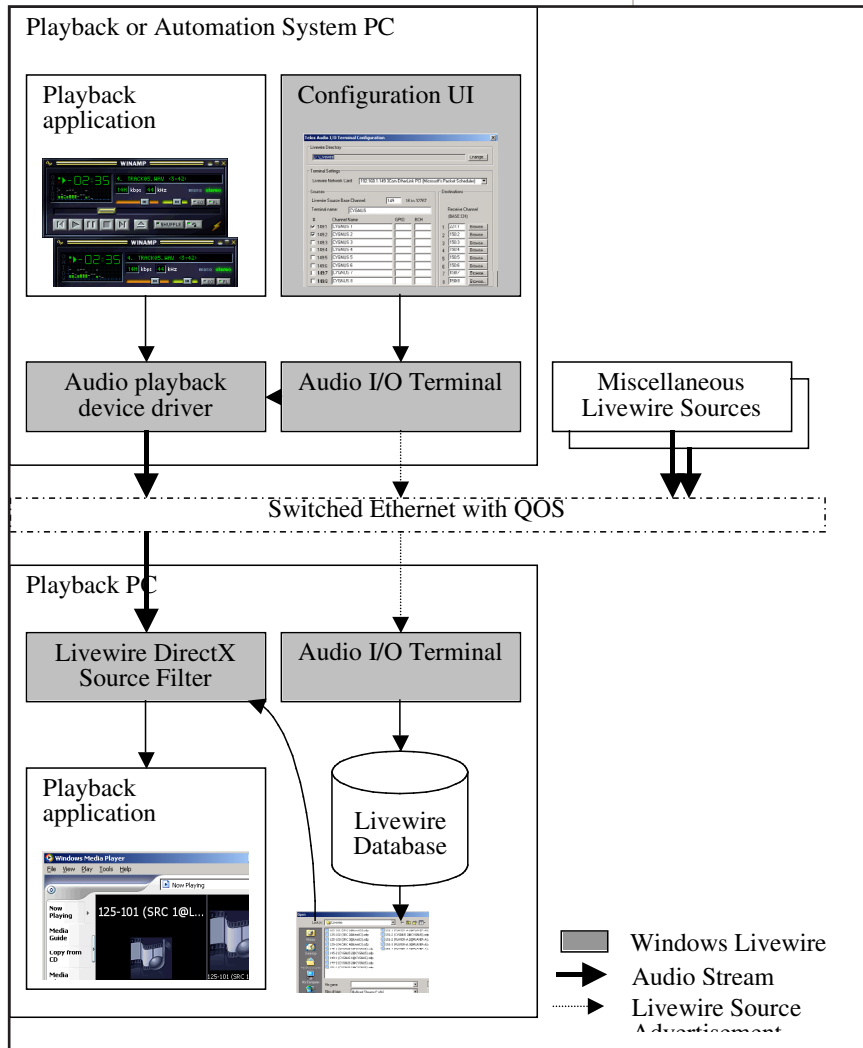
Axia IP-Audio Driver vs Axia IP-Audio Multichannel

The Axia IP-Audio Driver version supports emulates

a single sound card, with one stereo audio output device and one stereo audio input device. This version is suitable for typical playback or recording applications. Axia IP-Audio Multichannel (OEM version) emulates 16 sound cards, with one stereo audio output device and one stereo audio input device per “sound card”. It is intended for more complex professional applications. Other than the number of inputs the two drivers are the same.

**Advanced Info:
Using the Axia IP-Audio Driver as a Playback Device**

As mentioned earlier, your engineer will typically install and configure your IP-Audio Driver Software for you. This section is provided in the unlikely event that you need to manually modify that configuration.



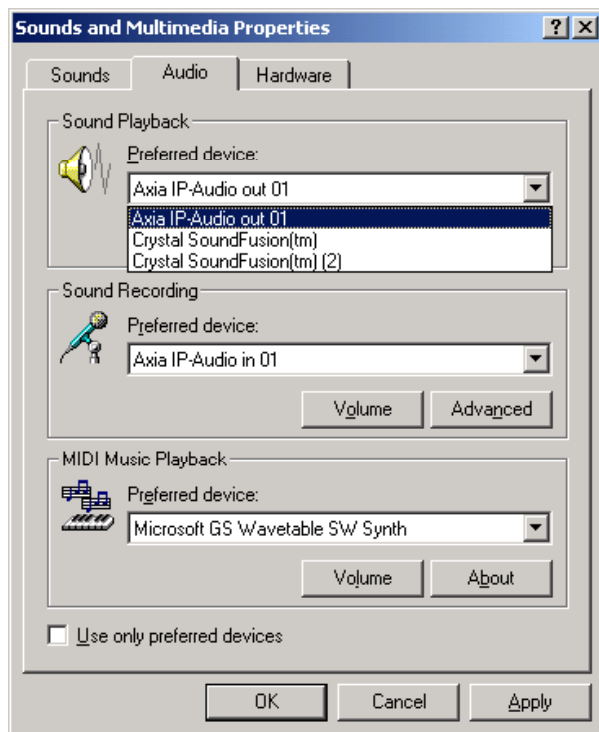
Livewire stream on the Livewire audio network. This stream is available on the network and can be monitored by Axia nodes or Axia IP-Audio Driver-equipped PC, as shown here.

Note that Scenario 2 also demonstrates how the IP-Audio Driver enables any PC on the Axia audio network can serve as a monitoring station and can be used to monitor any audio stream on the network. The driver

Using Axia IP-Audio for Playback/ Recording To/From the Livewire LAN

Make sure that your player application is configured to use the Axia IP-Audio Input device. Many recording applications use a drop down menu to list all available audio output devices; simply choose from the list.


You may also wish to specify one of the Axia IP-Audio Input Devices as the preferred “Sound Recording”



device in Windows Sound and Multimedia Properties control panel, as shown above. This specifies the Axia IP-Audio Input device for recording on applications that do not allow input device selection.

Once you start playback on the selected Axia IP-Audio Output device, the driver will start generating stream of network packets to a Livewire channel. Livewire nodes configured to receive audio on that channel will receive the audio data. If the lowest (bottom) segment of the output audio meters are illuminated this indicates that the stream is present, even if the audio source is currently silent.

What's Next

Axia is the networked-audio division of Telos Systems, the world's foremost provider of broadcast telephone systems and makers of famous Zephyr Xstream ISDN codecs. Some of these devices may be supplied with your system; the next chapters will give you tips on how to use these systems. 

Chapter 7:

Telos TWOx12 Phone System

Introduction

The Telos TWOx12 is designed for stations where up to 12 incoming talk lines are required. It is also well suited to dedicated studios within larger facilities. It also has the flexibility to serve two studios simultaneously, with one hybrid for each studio.

At the heart of any broadcast system are the hybrids used. Based on the Telos TWO dual hybrid, the TWOx12 includes simply the best broadcast hybrids made. With a trans-hybrid loss of well over 70dB (when used on digi-



tal lines) it allows for the best possible announcer audio over the air. Performance when used with open speakers and in teleconferencing applications is also superb.

The TWOx12 combines the TWO's dual hybrid design with a twelve line system controller and support for up to two (expandable to eight) Telos Desktop Director control surfaces - typically one for a phone screener and one for talent. Each Desktop Director can be easily reconfigured between the Talent (on-air control) and Producer (screener) modes of operation. The Desktop Director has an intuitive user interface and we are confident your staff will take to the system in no time.

Desktop Director Operation

Although the TWOx12 has front-panel controls, you will generally use the controls on the Telos Desktop Di-

rector to control phone calls.

The Desktop Director acts as a studio telephone and at the same time acts as a control surface for putting lines on either of the Telos TWOx12's built-in hybrids. The Desktop Director has two modes of operation – Producer (Screener) mode and Talent mode. This section gives general information while Sections 5 and 6 give complete details on using the TWOx12 and Desktop Director in Producer and Talent modes, respectively.

Status Symbol™ Icons & What They Mean

The Desktop Director breaks new ground in broadcast telephone control surfaces with the use of easy to understand Status Symbol Icons displayed on the LCD matrixes located between the two rows of line selectors. In most cases the meaning of each symbol is easily and effortlessly understood without study. However, to be complete, here is what each sequence means.

-  Line Available
-  Line In Use
-  Incoming Call
-  Call on Handset
-  Call on Speakerphone
-  Call on Hold
-  Call Screened, Ready For Air
-  Call is Next in Ready-for-Air Queue
-  On Air, Hybrid 1
-  Locked On Air, Hybrid 1
-  On Air, Hybrid 2
-  Locked On Air, Hybrid 2
-  Busy All Lines

Line Control Buttons

These work differently depending on whether the Desktop Director is in Talent or Producer mode.

In Talent mode each line has 2 buttons, the left column for Caller 1 (or whatever you choose to label the fader for hybrid 1) and right column called Caller 2. Pressing a line button places the caller on that line on-air on the corresponding hybrid.

NOTE: The TWOx12 can be “split” to control two studios. When this mode is in use, only the left column will be active and will control the designated hybrid.

Pressing a Line button when caller is already on-air locks the caller. To unlock press that Line button a third time.


If the handset is picked up, or the LED next to the hands free button is illuminated, the Left row of buttons will control the line to be used on the telephone. The right row of buttons will continue to control the caller 2 hybrid as usual. To place a caller currently on the handset on the air, press either the left (caller 1 hybrid) or right (caller 2 hybrid).

In Producer mode the left row of buttons is used to answer calls on the handset (or headset). Pressing a left Line button during a call on the handset, or in hands free mode, locks the caller. To unlock press that Line button a third time.

The right row of buttons is used to place a caller on screened hold. For more details on how screened hold works, see the section on the Next button, below.

Hold & Drop

These buttons place a caller on Hold, or Drop (disconnect) the call, respectively. In Talent mode these affect only the corresponding bank of buttons above them. A locked line will not be affected unless you unlock it first (by pushing the corresponding LINE button). All unlocked lines in that column will be dropped.

In Producer mode the left hold button is normal hold while the right hold button places a caller on “screened hold”. For more details on how screened hold works, see the section on the Next button, below. This right-hand hold button may be replaced by with a keycap labeled with a check mark  if the Desktop Director in question will be used primarily for screening.



Next

This button works differently in Producer versus Talent mode.

In Talent mode, this button will direct the call designated by the screener as NEXT to Caller 1 hybrid. When NEXT is pressed, any unlocked caller on the Caller 1 Hybrid will be dropped. It is also possible to configure the system to put the caller back hold instead.

If a Call Screener is used, s/he may place a caller on screened hold (see above) and the check mark icon will be displayed. This indicates that the caller has been checked by the Screener, and s/he is ready to go on air.

Calls placed on screened hold in this manner are being queued up from the first selected to the last one selected by the Screener. The NEXT caller line icon is represented by a blinking check mark.

If a screener is not used, the line which has been ringing the longest will be selected instead. If no line is ring-

ing, the line on hold the longest will be selected.

Note that if the handset is off-hook the caller will be directed to the handset, using the same priorities outlined above.

In Producer mode the NEXT button will select the line which has been ringing longest. If no line is ringing, it selects the line that has been on (standard) hold longest.

When NEXT is pressed, any unlocked caller will be dropped. It is also possible to configure the system to put the caller back hold instead.

Block All

Pressing this key will clear all lines with the only exceptions being lines on-air, on the handset, on hold and direct dial-in “priority” lines such as the studio hotline. This allows you to prevent early callers from getting in on contests until after the DJ has made the announcement and turned off Block All.

Ringing lines, other than “priority” lines, will be dropped. Those lines will then be blocked from accepting any calls. For ISDN lines, the caller may get a busy signal, or they may continue to hear ringing. For analog lines, the ringing calls will be connected to a silent audio channel. Further calls will get a busy signal.

Pressing Block All again will release the lines and allow incoming calls.

Telephone Mode Buttons

Dial pad buttons

These are used for dialing out, as with any telephone. They remain active during a call for use accessing voice mail and other services.

Speaker phone button

This button places a call that is on the handset onto this Desktop Director’s built-in speaker phone. If the handset is on-hook this button places the next line selected on the left column on the speaker phone.


Mute button

This button mutes the handset microphone and hands free microphone on the Desktop Director™. The LED beside this button will light to indicate that these microphones are muted. Pressing this button again will cancel the muting function.

Hook flash/Transfer

This button performs a hook flash (briefly interrupts loop current) on POTS lines. Depending on the telephone lines used, this key may serve to activate features such as “transfer” or “3-Way Calling. This key does nothing if the line selected is an ISDN line.

User Function Buttons

These buttons can be used for control of equipment external to the Telos phone system. For instance, the  button may be used to control your obscenity delay dump, etc.

?#@! Button

This button controls an output on the parallel control port. This output will go low for the duration that the button is pressed. This output can control equipment external to the TWOx12 such as the “delay dump” input on an obscenity delay unit.

Record Control

1. SINGLE MODE : This is intended to control an external recording device to easily capture conversations from callers. It operated the external equipment by pulsing the “Record Start” (pin 3) and “Record Stop” (pin 4) outputs of the Parallel Control port. These pins are pulsed for approximately 1 second. For details on the electrical characteristics of these outputs, please see Section 2.9.1.

Pressing the button once when nobody is on air, “arms” the recording feature (LED blinking).

Once armed, placing a caller on either of the two hybrids starts the recorder automatically (LED solid). If no caller is on either hybrid, the “Record

Stop” output will be pulsed to stop the recorder and the system will re enter the “armed” state.

Pressing the button, once when somebody is on air, starts the recorder (LED solid) immediately instead of entering the armed state.

While recording (LED solid), pressing the button stops the recorder and puts the system back in the armed (LED blinking).

When in armed state (LED blinking), pressing the button cancels the recording feature.

2. DUAL SHOW MODE: When the TWOx12 is used in any of the dual studio modes, the Record logic will only operate from Desktop Directors accessing Show 1 (Hyb 1). The RECORD button is disabled for any DD connected to SHOW2 (Hyb 2).

Desktop Director LCD Screen & Menu Navigation Buttons

The LCD status is used for general status information and for configuring the Desktop Director. Items can be selected and changed using the 6 buttons immediately below the LCD screen. The buttons in the row immediately below the LCD screen are “soft buttons”. The function of these buttons will be indicated on the bottom row of the screen.

The LCD screen and menu navigation buttons are used to set volume settings, LCD screen backlight and contrast, and to select menus allowing to access some of the TWOx12 configuration parameters. The function of the three “soft keys”, immediately below the LCD screen will vary depending on context, and will always be displayed on the screen. In most cases, the left softkey offers a BACK function. The BACK function takes you back to the default screen without saving unconfirmed changes, except for volume and LCD settings.

The UP & DOWN ARROW buttons scroll through the options in a menu while the Select button chooses a menu selection or confirms an entry.

Changing from Producer to Talent mode

At any given time, each Desktop Director is in either Producer or Talent mode. This can be changed at any time for each Director in the system. Producer mode can be thought of as “Screener phone” mode while Talent mode can be thought of as “on-air control” mode. As discussed earlier, the function of certain buttons is different in these two modes. To change from Producer mode to Talent mode:

- Press the soft key button below the word menu. • The words Change Director Mode should be highlighted by a black cursor bar and the black text will change to white.
- Now press the SELECT button. You’ll be offered Producer or Talent Mode. Press UP or DOWN to change the setting. Press SELECT to confirm your choice or just press the soft key below the word BACK to return the main menu without the change taking effect.

Volume Menus

Pressing the soft key button below the word volume will display the volume adjustment screen.

Press UP or DOWN to select the level to be adjusted. Pressing <-> will decrease, while pressing <+> will increase the volume level for the item displayed. The change will take place immediately. Pressing the soft key button under the word BACK or the SELECT button will return you to the main screen. The following volume levels can each be individually adjusted:

- » Ringer: This adjusts the volume of the ringer. Note that the ringer will usually be turned off whenever a microphone channel is on.
- » Speaker: This adjusts the level of the built in speaker for the Desktop Director’s hands-free option.
- » Handset: This sets the level of the caller on the handset. If you are using a headset attached to the Desktop Director, this will adjust the level in the headset as well.

Other Menu options

When selecting the menu from the main LCD screen, the following other options are offered.

- » Select Ringer: It is possible to change the ringer

tone on each Desktop director. There are four different choices. Press SELECT to apply your changes.

- » Configure Shows: This feature gives access to the TWOx12-Split functions, where each hybrid is assigned to each of two studios.
- » Adjust LCD Panel: gives access to the same menu as the lcd button on the main screen.
- » Diagnostics: This feature gives a status of the connection between the Desktop Director and the TWOx12 unit to which it is connected. Two values are given:
 - * Near End: gives the number of lost or corrupted data packets in the TWOx12 to Desktop Director direction since this Director was powered on.
 - * Far End: gives the number of lost or corrupted data packets in the Desktop Director to TWOx12 direction since this Director was powered on.
- » Version: This feature gives the currently running TWOx12 software version number, and the currently running Desktop Director software version number for easy reference.
- » About: Information about the product.

The Console Director

Your Axia Element consoles are equipped with “Console Directors” that work very much like the Desktop Director in TALENT mode, with the same icons and controls. Use them the same way as you would the Desktop Director to control calls that are on-the-air.

Conferencing Multiple Callers in Producer Mode – Using lock mode

The TWOx12 offers the possibility to create a conference call off-air, in Producer mode. Pushing a Line button once places that line on the handset. Pressing the same Line button a second time will place that line in LOCKED status. The status symbol for that line will display the LOCKED symbol. The Drop and Hold buttons have no effect on a line in LOCKED status. If another line is selected, the line in LOCKED status will remain on the handset. When a LINE button is selected a third time it reverts to standard status with the handset or

speakerphone ICON displayed.

CONFERCING (also called BUTTON MASHING)

- In standard mode, select your first caller on the left column of the Desktop Director.
- If you wish to keep a caller on the handset and add more callers, put the first caller in LOCKED status by pressing the Line button a second time to avoid accidentally hanging up on him/her.
- A second caller can be added by selecting another Line button on the left column using the same procedure. Pressing that Line button again can also LOCK this caller. You can add additional lines in this manner).
- Audio fidelity may decline. Remember, to unlock a line simply press its Line button. Hold or Drop will operate on all unlocked lines.

Placing Callers On-Air

Single caller

To place a caller on-air, push the appropriate Line button for the phone line on which the caller is located. The left Line button will place the call on the Caller 1 Fader (sometimes called Phone 1) while the right Line button will place him/her on the Caller 2 Fader (sometimes called Phone 2).

Note that if the line has already been screened, the Screened symbol (a check mark) will be displayed. If the caller has not been screened the Hold symbol or Ringing symbol will be displayed. Once you select the line the ICON will change to an arrow indicating it is on the Caller 1 <- or Caller 2 -> fader. The line may be dropped or placed on hold by using the buttons below the row indicated by the arrow. If you select another line on the same column, the line selected earlier will be dropped unless the line is placed in locked mode.

You will, of course, need to turn on and bring up the appropriate input on your console (mixing desk). You will usually have two faders associated with the phone system, one for “Caller 1” (or “Phone 1”) and one for “Caller 2” (or “Phone 2”).

Multiple Callers

Pushing a Line button once places that line on-air. Pressing the same Line button a second time will place that line in LOCKED status. The status symbol™ for that line will display the LOCKED symbol. The Drop and Hold buttons have no effect on a line in LOCKED status. If another line is selected on the same column, the line in LOCKED status will remain on-air. When a LINE button is selected a third time it reverts to standard status with the <- or -> ICON displayed.

The usual method of operation would be to lock the guest (if any) on the Caller 2 hybrid and use the Caller 1 hybrid for call-in listeners. If no guest is present, you can follow the directions below.

CONFERENCING on a single hybrid (also called BUT-TON MASHING)

- In standard mode, select your first caller on the left (Caller 1) row of the Desktop Director.
- If you wish to keep a caller on the air and add more callers, put the first caller in LOCKED status by pressing the LINE button a second time to avoid accidentally hanging up on him/her.
- A second caller can be added by selecting another LINE button on the Caller 2 (right) column using the same procedure. Pressing that LINE button again can also LOCK this caller. You can add additional lines in this manner.
- Although audio fidelity may decline, you may have multiple callers on the Caller 1 column or the Caller 2 columns. For best quality always keep the number of calls evenly distributed across the two columns.

Remember, to unlock a line simply press its LINE button. You may now press HOLD or DROP as desired.

Dropping Calls Off-Air

Look at the Status Symbol icon for the line(s) in question and verify that that the LOCK symbol is not showing. If LOCK is showing press the Line button for this line to unlock it.

If you wish to select another caller you can just press another Line button in this column, or you can press the

Drop button.

To place the call on hold press the Hold button instead.

Chapter Eight:

Zephyr Xstream ISDN Codec

Introduction

In 1993, we had a dream. We envisioned a way for CD-quality audio to be sent over common digital phone lines — the perfect marriage of advanced audio coding and digital telephone technologies. Our pursuit of that dream resulted in the Telos Zephyr™, which transformed broadcasting by making ISDN an easy-to-use, effective tool for sending and receiving high-quality audio.

Since its introduction, broadcasters and audio professionals worldwide have made Zephyr the most successful digital broadcast product ever. Its name has become synonymous with easy, instantaneous point-to-point audio transfer: “Just Zephyr it to me!”

In short, the Zephyr Xstream is advanced, economical, flexible, and easy to use audio codec.

Basic Operation

Introduction to Zephyr Xstream’s Menus

The Zephyr Xstream has seven menus. The four primary menus are AUDIO, CODEC, TEL and SYSTEM. Repeated presses of the associated menu button, located immediately below the LCD screen, access these menus.

The Default Display

The default display will appear after the system has been left idle for 60 seconds. It gives summary information on system status. At the bottom, in inverse text is the status of each ISDN “line” or V.35 port.

The Primary Menus; AUDIO, CODEC, TEL & SYSTEM



Pressing of any of these primary menu buttons once gives a summary status screen for the functions related to that menu.

Repeated presses of the menu button pages through the menu’s screen one at a time. The intention is that frequent users will learn the locations of commonly accessed items, and will be able to jump directly to the associated menu page in question.

Once a desired page of menu selections is shown on the display, you must highlight and select the item to be

changed (or field to be edited). The current selection will be highlighted; i.e. it will be displayed with white text on a black background. Other menu items (which are not highlighted) will be shown with black text on a white background. You can change which item is highlighted using the <UP> & <DOWN> buttons.

NOTE: The bottom line of the LCD display will give you a description of the highlighted menu item. This context sensitive help will give you an indication of what an item does. For more information about that item, push the <SEL> button.

Once the desired menu selection is highlighted, you must select it by pressing the <SEL> button. The selected item will move to the top of the screen and detailed information about that menu selection will be shown below it.

In the case of most menu items, you can now view the options for this item using the <UP> or <DOWN> but-



tons. Note that the option currently in effect has square brackets “[]” around it.

TIP: If you change your mind, get lost, or otherwise panic, don't worry. Pressing any menu key (i.e. <AUDIO>, <CODEC>, <TEL>, or <SYSTEM>) will de-select the item currently selected, without saving the changes in progress.

In the case of numeric or alphanumeric text fields, you can begin entering text as soon item has been selected with the <SEL> button. To erase previously entered text, use the <UP> button.

Let's go through a couple of quick examples to see the Zephyr Xstream menus in action.

Example 1: Selecting a setting from a list (i.e. the bit rate):

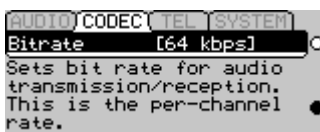
1. Press the <CODEC> button twice. You should see a screen that looks like the following:



2. Now press the <DOWN> button twice so the menu item “bitrate” is highlighted (i.e. is shown with white text on a black background) as shown below:



3. Press the <SEL> button once. Note that the menu item moves to the top of the screen and information about “Bitrate” is displayed.



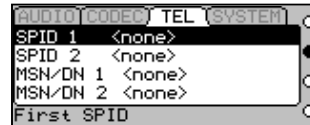
4. Press the <DOWN> button repeatedly to view the possible settings for this menu item (i.e. 56 or 64).

You can press the <CODEC> button at any time to “escape” without saving the changes you have made.

5. Press the <SEL> button to confirm your selection. Or, press <CODEC> to ignore the change.

Example 2: Entering a numeric entry (i.e. entering a SPID):

1. Press the <TEL> button twice. You should see a screen that looks like the following:



2. Verify that the menu item “SPID 1” is highlighted (i.e. is shown with white text on a black background). If not, use the <UP> button to highlight this menu selection.
3. Press the <SEL> button once. Note that the menu item moves to the top of the screen and information about “SPID 1” is displayed.



4. Press the <UP> button repeatedly to erase any existing information.
5. Type in the new SPID number using the Zephyr Xstream's keypad.

IMPORTANT TIP: You can press the <TEL> button at any time to “escape” without saving the changes you have made.

IF THIS UNIT HAS ALREADY BEEN PROGRAMMED YOU SHOULD PRESS <TEL> NOW!

6. Press <SEL> to confirm your entry. Or, press <TEL> to ignore the information and abort the entry process.

More on the Numeric Keypad – Entering Alphabetic Characters

A given keypad button corresponds to the number printed on it. It can also be used for text entry of the letters below it. Additional characters are available on the <*> key (i.e. * ([] <> <Space>) and the <#> key (i.e.

. , ; : ! ? <space>) and underscore is available on the <1> key.



To enter text characters, press the key with the appropriate letter printed near it. Keep tapping it to cycle through the possible choices.

For example, tapping the 2-key in a text field will first enter A. Repeated taps will change that to a B, then C, then the lower-case letters, then the numeral 2. Enter a space by pressing 1 three times. Punctuation characters appear on the * and # keys.

For example:

Press 2 once	See the character A
... press it again	See the character B
... press it again	See the character C
... press it again	See the character a
... press it again	See the character b
... press it again	See the character c
... press it again	See the numeral 2
... press it again	Cycle starts again from A

To move to the next character in a field, tap any other key or press the <DOWN> button. To back up, press the <UP> key.

Special Menus; AUTO, DIAL, & DROP

The AUTO menu

The AUTO menu is used to enter or activate Dial Setups and Location Setups.

Dial Setups are very similar to the “Auto Dial Setups” used in the classic Zephyr. Each of the 100 dial setups consists of several pieces of information.

- A Setup number – this 2 digit number (00 to 99) can be used to instantly access a given setup.

- An alphanumeric Name
- Number 1 (optional) – A telephone number to be dialed on “Line 1”
- A “mode” for number 1 – Either Zephyr for a codec call, or Phone for calls to a standard telephone.
- Number 2 (optional) – A telephone number to be dialed on “Line 2”
- A “mode” for number 2
- Include selection. This allows you to include CODEC menu settings (optional) – This is a snapshot of all settings of the CODEC menu. This allows the unit to automatically change to the settings used with a given site based on the Dial Setup used.

HOT TIP: You can use dial setups as a way to conveniently change your codec settings. Possibly to accommodate incoming calls from differing locations.

Simply leave the Number 1 & Number 2 field blank, and set Include to Codec. As always, be sure to set the CODEC menu items to the proper settings before creating your setup.

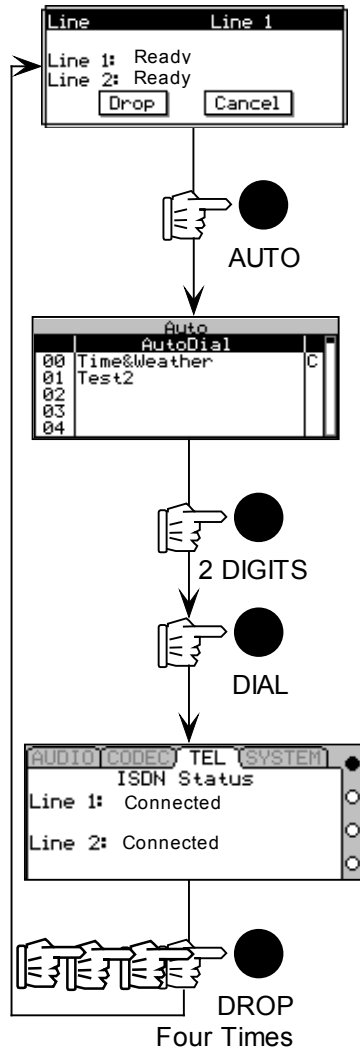
The DIAL menu

Activating a Dial Setup – Step by Step

Before beginning, verify the Zephyr Xstream is connected and ready for use. In most cases the line status should indicate Ready & Ready (with ETS 300 ISDN it is acceptable if Deactivated & Deactivated is displayed)

If necessary, activate the appropriate Location Setup for the location (if you have just arrived there) now (see below). Or you may need to manually configure SPIDs, DNs/MSNs and Telco information in the Tel menu (if no Location Setup exists for this location).

1. Press the <AUTO> key. The most recently used Dial Setup will be highlighted (white text on a black background). To activate this Setup Press <AUTO> again.
2. To choose a different Dial Setup you have two options:
 - a. Use the <UP> and <DOWN> to scroll through the list of 100 Dial Setups.
 - b. Or: Use the Keypad to enter the two-digit number (00 to 99) for the desired setup. That setup will be displayed and will be highlight-



The sequence for activating a Dial Setup

2. (Optional) If you wish to include CODEC settings in your Dial Setup, you may wish to dial the location in question now, to verify that your codec settings are correct. If you choose to “include” the codec settings in step 8, the current codec setting will be stored with the new setup.
3. Press the <AUTO> key. Choose the Dial Setup to be entered. Note: You can edit an existing setup by se-

ed.

3. Once the desired setup is highlighted, press the <AUTO> button to activate the dial sequence.

Note that the notation “C” in the right column when viewing your stored Dial Setups indicates that a particular setup includes CODEC menu settings.

Creating Dial Setups – Step by Step

1. Configure the Zephyr Xstream’s CODEC menu as it will be used when connected to this location. You can skip this step if you are creating a Dial Setup that does not include codec settings.

HOT TIP: You will probably wish to include the CODEC settings in your Dial Setup. This allows you to dial a variety of locations, each with differing codec requirements, without the need for manual configuration.

lecting it. To choose a Dial Setup you have two options:

- » Use the <UP> and <DOWN> to scroll through the list of 100 Dial Setups.
 - » Or: Use the Keypad to enter the 2-digit number (00 to 99) for the desired setup. That setup will be displayed and will be highlighted.
4. Once the desired setup is highlighted, press the <SEL> button to allow entry of (or editing of) the information for this setup. If an empty setup was selected, “Empty Entry” will be displayed.
 5. Press <SEL> to continue or press <UP> and then <SEL> to escape without changing.
 6. Enter a name (see the beginning of this chapter section for an example on how to select and enter information).
 7. Enter one or two phone numbers, as required for the transmit and receive modes to be used when communicating with this site.
 8. For each phone number entered, select a call type. This would be Zephyr if calling a codec or Phone if placing a call to a telephone.
 9. You can choose whether to Include Codec settings or not by selecting this menu selection. The choices are Number (no codec information included with this dial setup) or Number & Codec.
- If Number & Codec is selected the system will include the current settings in the CODEC Menu with this setup. Each time this setup is activated, the system will reset the CODEC Menu items to these setting before dialing the number(s) for this setup.
10. Move the cursor to the word Save and then press <SEL> to complete the entry process.

Editing Dial Setups

1. If this setup includes codec information, configure the Zephyr Xstream’s CODEC menu as it will be used when connected to this location. If the existing codec information included with this setup is correct, simply activate the setup to recall those settings now (then drop the call (s) by pressing drop 4 times). If you are editing a Dial Setup that does not include codec settings you can skip this step.
2. Press the <AUTO> key. Choose the Dial Setup to

be edited. To choose a Dial Setup you have two options:

- » Use the <UP> and <DOWN> to scroll through the list of 100 Dial Setups.
 - » Or: Use the Keypad to enter the 2-digit number (00 to 99) for the desired setup. That setup will be displayed and will be highlighted.
3. Once the desired setup is highlighted, press the <SEL> button. The name of the setup will be displayed.
 4. Press <UP> and then <SEL> to edit the setup.
 5. Edit the name, number 1 or 2, or call type 1 or 2 (see the beginning of this section for an example on how to select and enter information).
 6. You can choose whether to Include Codec settings or not by choosing this menu selection. The choices are Number (no codec information included with this dial setup) or Number & Codec.
 7. Number & Codec is selected, the system will include the current settings in the CODEC Menu with this setup. Each time this setup is activated, the system will reset the CODEC Menu items to these setting before dialing the number(s) for this setup.
 8. Move the black cursor to the word Save and then press <SEL> to complete the editing process.

Location Setups

Location setups are a useful feature that allows you to pre-configure information about the ISDN line (TEL menu information) and the mixer setup/assignment information (AUDIO menu and MX/MXP front panel information) for frequently visited locations. Each dial setup consists of several pieces of information:

- A Setup number – this three-digit number can be used to instantly access a given location setup.
- An alphanumeric Name
- TEL menu settings – This is a snapshot of all settings of the TEL menu.
- AUDIO settings (optional) – This is a snapshot of all settings of the AUDIO menu. Note, the front panel Left/Right and Snd/Rcv mixer assignments on the Zephyr Xstream MX/MXP model’s front panel are included along with the settings in the AUDIO menu.

Activating a Location Setup – Step by Step

Note that this process is the same as accessing a Dial Setup, only a three-digit number is used.

1. Press the <AUTO> key.
2. To choose a Location Setup you have two options:
 - » • Use the <UP> and <DOWN> to scroll through the list of 30 Location Setups. These are setups # 100 to 130 - may need to press and hold <DOWN> to scroll past the Dial Setups.
 - » Or, use the Keypad to enter the three-digit number for the desired setup. That setup will be displayed and will be highlighted.
3. Once the desired setup is highlighted, press the <AUTO> button to activate the configuration sequence. Once configuration is complete, the Xstream will show the following screen:



4. Press <UP> then <SEL> to reboot.

The illustration below shows several sample Location Setups. Note that the notation “A” in the right column indicates that a particular setup includes AUDIO menu settings as well as TEL menu settings.

Auto		
100	Jacobs_Field	A
101	Gund_Basketball	A
102	Gund_Concert	A
103	Productn_Studio	
104		
105		

Creating Location Setups – Set by Step

1. Configure the Zephyr Xstream with the ISDN line to be used at this location (see below). You must make certain that the menu selections in the TEL and AUDIO (optional) menus are set correctly:
 - » All menu settings in the TEL menu. Ideally, you should be connected to the line for the target location, so you can verify the ISDN settings are correct. In that case, the line status should indicate Ready & Ready (with ETS 300 ISDN it is acceptable if Deactivated & Deactivated is displayed). If you are not connected to the line that’s ok too, but you should be certain that the menu selections in the TEL menu are set correctly. NOTE: TEL

menu settings are always included in a Location Setup.

HOT TIP: You can activate an existing setup to recall these parameters, assuming if you have an existing setup for the line in question. Do this first, and then make the necessary changes to the AUDIO menu settings. Then create your new LOCATION setup as described below.

- » If you wish to include AUDIO settings in this Location Setup (optional):
 - * All menu settings in the AUDIO menu.
 - * Front panel Main mixer A/B buss assignments (MX/MXP only)
 - * Front panel Snd/Rcv settings for Local Mix 1 & 2 (MX/MXP only)
2. Press the <AUTO> key. Choose the Location Setup to be entered. Note: You can edit an existing setup by selecting it. To choose a Location Setup you have two options:
 - » Press and hold the <DOWN> key to scroll to the list of 30 Location Setups (these are setups # 100 to 130).
 - » Or, use the Keypad to enter the three-digit number for the desired setup. That setup will be displayed and will be highlighted.

Auto		
100	Jacobs_Field	A
101	Gund_Basketball	A
102	Gund_Concert	A
103	Productn_Studio	
104		
105		

3. Once the desired setup is highlighted, press the <SEL> button to allow entry of (or editing of) the information for this setup. If an empty setup was selected, "Unused Entry" will be displayed:

Unused Entry

4. Press <SEL> to continue to the screen shown below:

Name <none>

Include Tel

or press <UP> and then <SEL> to escape without changing.

5. Enter a name (see the beginning of this chapter section for an example on how to select and enter information).

Name Jacobs_Field

Include Tel & Audio

6. You can choose whether to Include Audio settings or not, by selecting this menu selection. The choices are Tel (no Audio menu information included with this Location setup) or Tel. & Audio.

If Tel is selected the system will include the current settings in the TEL Menu with this setup. Each time this setup is activated, the system will reset the TEL Menu items to these setting before dialing the number(s) for this setup. If Tel. & Codec was selected the AUDIO settings will also be changed when this setup is activated.

7. Move the cursor to the word Save and then press <SEL> to complete the entry process.

Editing Location Setups

1. If this setup includes audio information, configure the Zephyr Xstream's AUDIO menu and front panel mixer assignments as it will be used when connected to this location. If the existing Audio information included in this setup is correct, simply activate the setup to recall those settings now (then drop the call (s) by pressing drop 4 times). If you are creating a Dial Setup that does not include audio settings, you can skip this step.
2. Configure the Zephyr Xstream with the information ISDN line to be used at this location (see below). You must make certain that the menu selections in the TEL and AUDIO (optional) menus are set correctly.
3. Press the <AUTO> key. Choose the Location Setup to be edited. To choose a Location Setup you have two options:
 - » Press and hold the <> to scroll to the list of 30 Location Setups (these are setups # 100 to 130).
 - » Or, use the Keypad to enter the three-digit number for the desired setup. That setup will be dis-

played and will be highlighted.

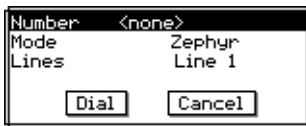
- Once the desired setup is highlighted, press the <SEL> button. The name of the setup will be displayed. Press <UP> and then <SEL> to edit the setup.
- Edit the name (see the beginning of this chapter section for an example on how to select and enter information).

You can choose whether to Include Audio settings or not, by selecting this menu selection. The choices are Tel (no Audio menu information included with this Location setup) or Tel. & Audio.

- Move the cursor to the word Save and then press <SEL> to complete the editing process.

The DIAL menu

The DIAL menu appears after the Dial button has been pressed and looks like the following:



If you wish to dial a data call to another codec (Mode = Zephyr) you can simply enter the desired number and press <DIAL> again. A second call can be placed on “Line 2” by pressing <DIAL> a third time.

To drop all calls in progress, simply press <DROP> four times.

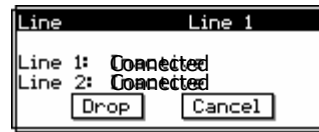
TIP: The Number field in the DIAL menu is different from other entry fields in the Zephyr Xstream. It is NOT necessary to press <SEL> before entering the number to be dialed.

If you wish to specify the Line to be used, or use a mode other than the default (i.e. other than Zephyr,) you can do so. After entering the number, press <SEL>, then use the <UP> or <DOWN> keys to move to the desired menu selection, and press <SEL> again to change the item. Repeat this process till all items are set as desired.

Once you have set the Line and Mode press <DIAL> to make the connection.

The DROP menu

The Drop menu is activated by pressing the <DROP> button once. It looks like this:



The system will default to Line 1 if that line is in use. It will default to Line 2 only if Line 1 is not in use. Press <DROP> a second time to drop the line shown.

To drop a Line other than the line shown, press the <SEL> button, then press <UP> or <DOWN> to select the desired Line(s). To drop the Line(s) press <DROP> again.

Codec Settings

Transmit, Receive, Bit Rate, & Sample Rate

These settings must be set correctly to achieve an end-to-end bi-directional audio connection. Since this is just an introduction, we will just briefly tell you what each is.

Transmit

This determines the format of the audio locally coded for transmission. In order for the far end to receive this audio, the far end codec must:

- Support this transmission mode; and
- Must have its “Receive” (or “decoder”) mode set appropriately to receive the mode being sent.
- bit rate and sample rate settings must be matched between the two ends.

Receive

This instructs your Zephyr Xstream to decode a particular mode being transmitted from the far-end codec.

The easiest option here is to set Receive to the “Auto” setting. This mode will attempt to determine the mode being sent from the far end, and internally set the Receive mode to match. All you’ll need to do is be sure your Sample Rate (see below) is matched to the far end. The mode selected will then be displayed on the default menu and the codec status screen so you can note it for

future reference.

As the auto mode takes several seconds to determine the correct mode, there are times you may wish to use a specific setting for Receive. In this case, your Receive mode must be set to decode the stream being sent from the far end.

If you choose not to use the “Auto” setting, you will need to know the Transmit (“Xmt” or “Encoder”) mode of the far end to determine the correct setting.

HOT TIP: You can determine the correct setting for the Receive mode by first using the Auto mode. Once the Xstream gets a receive lock simply press the <CODEC> button and check what Receive mode was found. Then press <CODEC> again and change the Receive selection to this option.

It is important to understand that the Transmit and Receive settings on two connected units may very well NOT match. This is because there are frequently times when it is appropriate to use a different coding mode coming versus going.

The important thing to remember is that the local Transmit mode must be compatible with the Receive settings and capabilities of the far end unit, whereas the local Receive mode must be compatible with the Transmit settings of the far end unit.

Bit Rate

This setting determines the per-channel bit rate used for digital audio transmission. This setting sets up the codec section and tells the ISDN interface what rate call to place the call over the network.

This parameter must be matched between the Zephyr and the far end codec(s).

The Zephyr Xstream specifies the bit rate per channel. Since certain modes (i.e. stereo or mono-128) require two channels to operate, the total (aggregate) bit rate would be double the number displayed in these cases.

This is important as some other brands of codecs specify the total bit rate.

Since the network must accommodate the rate selected, it is important to hang up (i.e. drop the call) and redial if this parameter is changed during a call.

Since the network has to know the bit rate for a given call, this information is available to the Xstream on incoming calls. Therefore, the Xstream will switch to the rate of the incoming call.

There is one exception however. If the Xstream already has a call in progress at a given rate, and a second call comes in (i.e. from a third codec) the unit will accept the call, but will not change rate (since that would cause a loss of audio from the original site). There will be no lock or audio from the site associated with the second call. In this case, the second remote site should drop the call, change rates to match, and redial - All three units must be at the same Bit Rate for this to work.

Sample Rate

This is the digital sample rate used for transmission/reception by the codec section. Note that this rate may or may not be the same as the AES/EBU sample rate used on the input or output. See section 10 (Detailed Menu Reference) for information on using the AES/EBU ports. The sample rate must be matched between your Zephyr Xstream and the codec at the far end for successful audio transmission.

If the sample rate is not matched, you will have no lock, intermittent lock loss, or audio that sounds pitch shifted.

Chapter Nine:

Axia iProFiler Audio Logger

Introduction

Axia iProFiler is an automated program archiving system that logs your station's audio using a PC.

Traditional audio logging systems used tape technology to record program audio. Cassette recorders, reel-to-reel decks, even VCRs were employed.

In the same way PCs made the tape deck obsolete, these mechanical logging systems and their stacks of tape have also been replaced by digital media.

With iProFiler, radio stations can now make MP3 audio logs of their programs. MP3 audio compression technology helps reduce storage space requirements, eases searches for specific material, and can archive morning shows for on-air replay.

The archived audio is saved as an MP3 file with additional time-of-day information. These files can then be played on any MP3 compatible system, can be sent over the Internet, be posted on Web sites for streaming audio, and be stored efficiently on high-capacity digital media. There's an automatic skimmer that switches from low-bit-rate logging to high-bit-rate skimming on the fly – perfect for telescoped airchecks or high-quality station composites. iProFiler's recording software (the "Server") runs under Windows 2000/XP, and the playback software (the "client") runs under Windows 95 or greater.

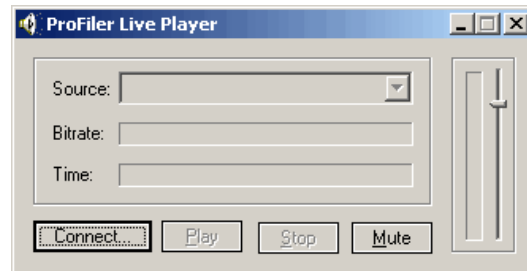
With iProFiler, you can listen to a live feed from a radio broadcast as it is being archived, or you can review radio broadcasts that have been previously archived.

The following applications make up the ProFiler client software:

- Live Player: For listening to a live feed.
- Archive Player: For listening to archived time-stamped audio files stored on the ProFiler server.

Using the iProFiler Live Player

Live Player is a program that allows you to listen to a live audio stream, as it is received by the ProFiler server.



Follow these steps to connect to a ProFiler server and listen to the audio:

1. Click the Connect button to establish a connection to the ProFiler server.
2. Enter the user name and password combination you've been given by your station engineer. You may want to check the **Remember** checkboxes to have the program remember your user name and password for next time.
3. Enter the server address then click the OK button. This address will be provided to you by the server administrator. It should have the following format:

server_name_or_ip_address:server_port

There are no spaces anywhere in the address.

4. The program will connect to the specified server and will display a list of archives available on the server in the Source drop-down box.
5. Select the archive you wish to hear then click the **Play** button.

Using the iProFiler Archive Player

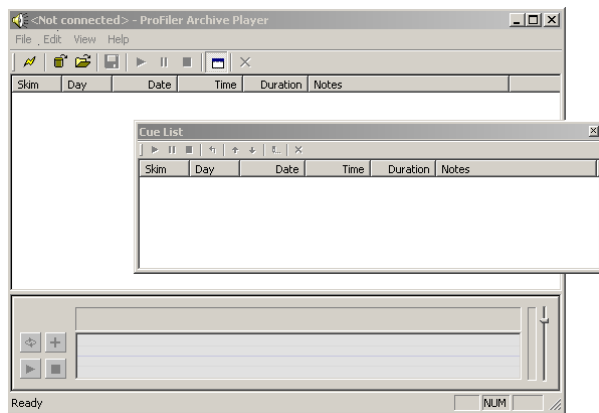
Archive Player is a program that lets you access and listen to stored audio from any computer that is networked to the iProFiler server where the audio is stored.

Using Archive Player, you can extract audio segments and save them on your own computer as iProFiler (AP3) files. AP3 files retain timestamp information for each segment, along with any notes that were entered during the archiving process. These AP3 files can be opened at any time using Archive Player.

AP3 files store the audio data in the MP3 format. As a result, AP3 files can be played by most MP3 players. You can also save the files using the .mp3 extension, or you may rename them with an .mp3 extension later. If you do this they will be opened and played by the default MP3 player instead of the Archive Player.

Menus

Starting the Archive Player will bring up the screen below.



There are five primary sections to the Archive Player screen:

- Menus
- Toolbar
- Audio Segment List Box
- Selection Panel
- Cue List Panel

The **File** menu lets you:

- » **Connect:** Connects to the iProFiler server
- » **Open Archive:** Requests list of segments stored on server. If not connected, will ask you to connect.
- » **Open File:** Opens locally saved files.
- » **Save As:** Saves the current cue list contents as local file.
- » **Exit:** Exits the program.

The **Edit** menu lets you:

- » **Add to Cue List:** Adds selected segment to the cue list
- » **Clear all entries:** Clear all segments from player

The **View** menu commands include:

- » **Cue List:** Displays the Cue List window.
- » **Toolbar:** Displays or hides the Toolbar.
- » **Status Bar:** Displays or hides the Status Bar.

The Help/About menu displays version information about the Archive Player.

Using the Toolbar

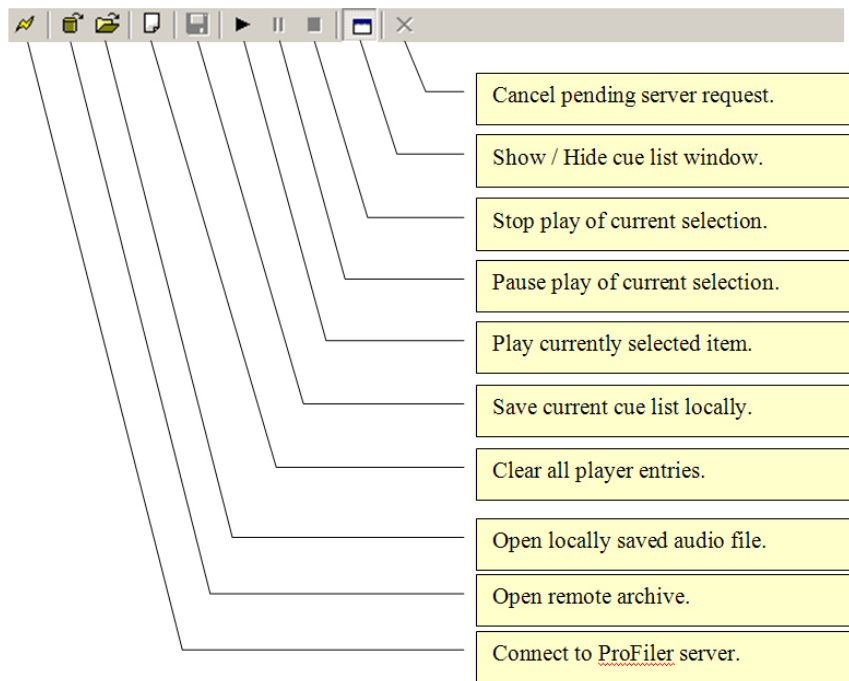
The Toolbar icons allow you to perform many of the same commands as those found in the Menus, only at the click of a mouse. See the illustration on Page 119 for details.

Audio Segment List Box

The Audio Segment List Box has six columns: Skim, Day, Date, Time, Duration, Notes.

Skim	Day	Date	Time	Duration	Notes
No	Mon	04/09/2001	13:40:48	00:00:32	Wall Street is just a casino...
Yes	Mon	04/09/2001	13:41:02	00:00:13	...neutral. Trading volumes down...
No	Mon	04/09/2001	13:44:59	00:03:57	... NSDQ was hitting 5000...
No	Mon	04/09/2001	14:00:00	00:15:00	... look at stock price comparison...
No	Mon	04/09/2001	13:47:44	00:00:20	... get more people to start a bussiness
No	Mon	04/09/2001	14:15:00	00:15:00	CNN Radio News
No	Mon	04/09/2001	14:30:00	00:15:00	... I didn't understand caller...
No	Mon	04/09/2001	14:45:00	00:15:00	CNN Radio News

Archived segments are recorded in 15-minute increments, starting on the quarter hour. However, if the microphone is turned on, the previous segment immediately terminates and a new recording segment begins. The new segment runs until the next quarter hour, at which point a new segment begins, recording until the microphone



The iProFiler Toolbar

is turned off, at which point a new segment recording begins.

Time is recorded on a 24-hour clock, with 0:00-12:00 set as a.m. time and 12:01 – 23:59 set as p.m. time. A “d” or “s” follows the time to indicate Daylight Savings Time or Standard Time respectively.

The list below describes the columns used by the segment list box:

- **Skim:** Indicates the whether the microphone was on or off during recording (“skim yes” means microphone was on).
- A **Microphone** icon quickly illustrates whether or not the mic. was turned on during recording.
- A **line through the microphone** indicates that there was no mic. input recorded during that segment.
- **Day:** The day on which the audio was broadcast and recorded.

Date: The date the broadcast was recorded.

Time: The time the broadcast was recorded.

Duration: The length of the recorded segment. Typically, segments are recorded in 15-minute increments, starting on the quarter-hour for a new segment.

Notes: This column displays notes for the segment. It is generally blank when reviewing archive audio but it is often used with AP3 files stored locally.

Selection Panel

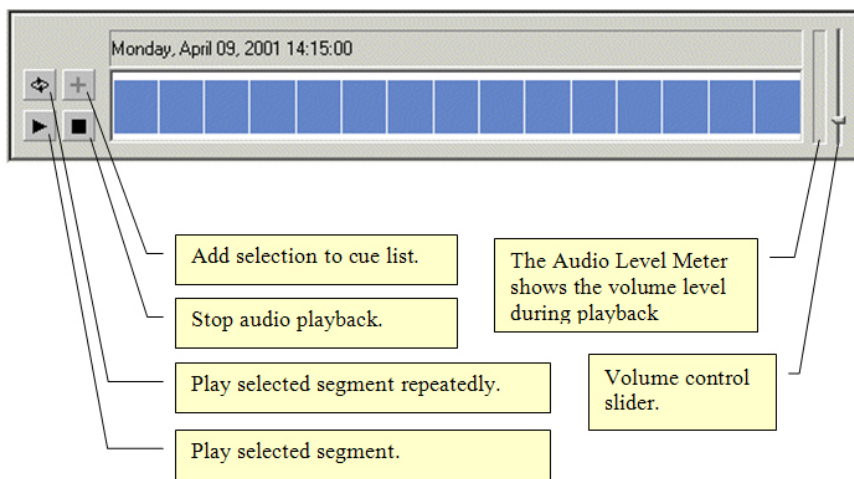
The Selection Panel displays a graphical representation of the currently selected audio segment.

Using the Selection Panel you can

change the playback position within a segment or select a portion of interest for playback or to save for later use. The illustration below illustrates and explains these controls.

The Cue List Window

iProFiler allows you to save lists of selected audio segments to a Cue List, where you can refer to them, play them, and export them for manipulation in an external audio editor if desired.



The iProFiler Selection Panel

Using the Archive Player

To start Archive Player, click the Start button on your computer. Choose **Programs** in the Start menu, then **Axias / iProFiler**, and select the Archive Player. This will display the iProFiler Archive Player window.

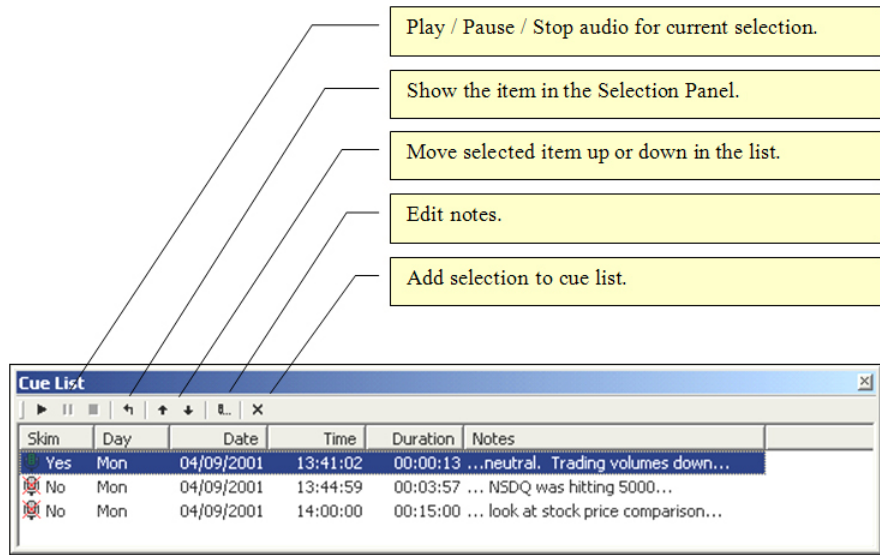
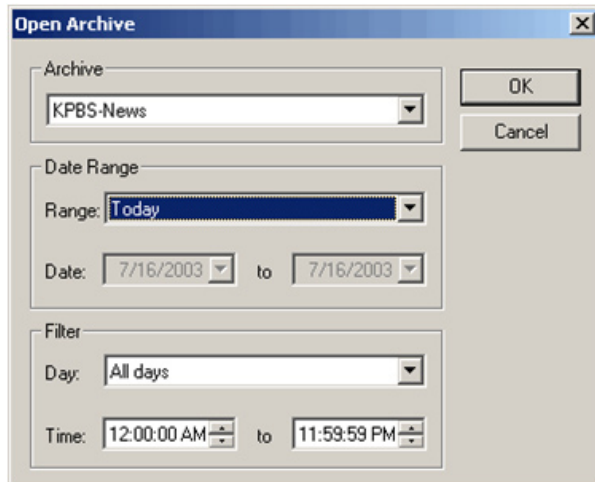
Before you can access a iProFiler audio archive, you must establish a connection to the server. Click the **Connect** icon to display the iProFiler Login dialog box, and enter the requested information. Note that the user name and password are not case sensitive. The server address will have been provided to you by the iProFiler server administrator. Click OK.

If connection is successful, the application will display the address of the connected server in the title bar. If the connect fails, an error message is displayed.

You may change the server connection at any time by choosing the **Connect** command and entering new server information.

Opening iProFiler Archives

To open an iProFiler archive, click the **Open Remote Archive** icon to display the Open Archive dialog box. This is where you will select the date range for the ar-



The iProFiler Cue List

chives you are looking for, and where you can set up filters for the records to be retrieved.

Select the archive you wish to use from the drop-down box, and select a date range by using the drop-down list in the Range box, or choose Selected dates and set a range manually using the date boxes

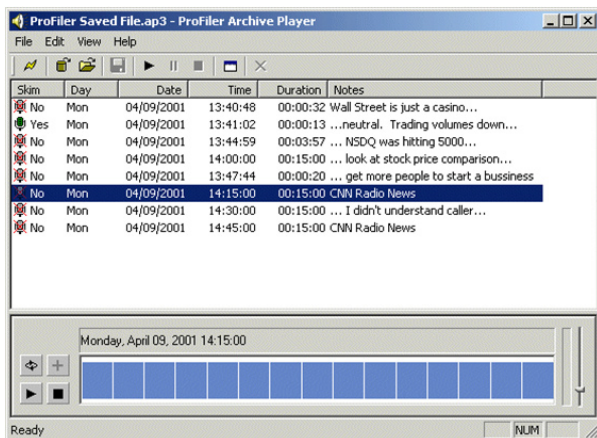
Use the filters to select a day or time for archived records to retrieve. For example, if you want only shows recorded on Sunday, select Sunday in the Day box. If you want something recorded between 10 p.m. and midnight, set those times in the Time boxes.

You can also use Archive Player to access AP3 files that have been stored on your own computer, without connecting to a network. Click the **Open** icon. This displays the Open dialog box. Highlight the file you want to open, and click **Open**. The saved records will be shown in the Audio Segment List Box.

Selecting Logged Segments

To select a logged segment, click anywhere on the archived entry. The entry will be highlighted and is then available for play.

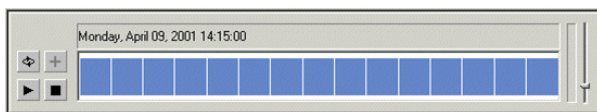
To play the selected segment, click the **Play** button on the toolbar. The selected segment will be shown in the Selection Panel.



The Audio Segment List Box

Archive Player will now play the entire selection from the beginning to the end.

To start play somewhere other than at the beginning, you will need to play the segment from the Selection Panel.



First, click anywhere inside the panel. Then press the Play button in the Selection Panel. The time indicator is displayed at the top of the Selection Panel and the segment will play to the end.

To play a specific segment of the selection, position your mouse cursor over the waveform, then click and drag to highlight the segment to play. Press the Play button, and the highlighted segment will play from beginning to end.

Double-clicking on the selection in the Selection Panel will highlight the entire selection. To loop play the segment, click on the Loop icon. The highlighted segment will play repeatedly.

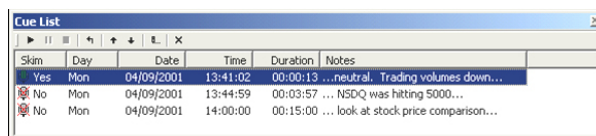
Creating and Using a Cue List

You can create a list of selected recorded segments from one or more entries. These selections are copied to the Cue List, where they can later be edited and saved as a single AP3 file.

You can always show or hide the Cue List by using View menu option or by using Ctrl+Q.

Why would you want a Cue List? If something interesting happened during a recording, such as a joke by the deejay or a particularly important comment by a guest on a show, you are able to capture that portion of the recording, create a note about the segment, and save both the segment and the note to the Cue List. These segments can be used later by the program director for promos for the show, etc.

Cue lists can also be sent using email. You can extract what you want from the archived files, annotate the segments selected, then save both the segments and the notes to an AP3 file, which can then be sent via email and played using the Archive Player. The AP3 file can also be played on an MP3 player if the file extension is changed from “.ap3” to “.mp3”. If the AP3 file is played on an MP3 player, no notes or segment time information will be displayed.



To add a selection or segment to the Cue List from the Selection Panel, highlight the selection or segment to be added and click the Add (+) button.

To add a selection to the Cue List from the Audio Segment List box, double-click on the entry. The selection or highlighted segment will be added to the bottom of the Cue List.

If, upon playing the Cue List, you find that you didn't select the entire segment you had targeted, you can return to the parent entry, and reselect the desired segment. To return to the parent entry, click on the left-angle arrow. This returns the parent entry to the Selection Panel.

You can add notes to the cue list by typing the notes into the Notes field on the Cue List. To add a note, click on the entry where you want to add a note. Then click the Edit Notes icon on the Cue List box or press the Enter

key. A text box opens on the entry you have selected. You can then type your notes. There is a 128-character limit in the Notes field. Click anywhere on the Cue List to finish the note and add it to the list.

To edit a note, click on the entry where you want to edit the note, then click on the Edit Note icon or press the Enter key. The text box for that entry will open. Click inside the text box and scroll through the note using the left or right arrow. Make your changes and click anywhere on the Cue List to finish editing the note entry.

To save your cue list as an AP3 file, type Ctrl-S, or choose File/Save As. This saves the Cue List contents as a single AP3 file containing all of the date, time, notes and audio information.

NOTE: Although an AP3 file can be directly played by most MP3 players (including Windows Media player), it is best played with the Archive Player. The date, time, and notes can be retrieve and viewed only by using the Archive Player.



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