element®

Installation & User's Guide

AXIA
A TELOS COMPANY

Revision 3.0 for:
• Element v2.x and higher
• StudioEngine v2.4.8x and higher
February, 2008

USA Class A Computing Device Information To User. Warning:

This equipment generates, uses, and can radiate radio-frequency energy. If it is not installed and used as directed by this manual, it may cause interference to radio communication. This equipment complies with the limits for a Class A computing device, as specified by FCC Rules, Part 15, Subpart J, which are designed to provide reasonable protection against such interference when this type of equipment is operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference. If it does, the user will be required to eliminate the interference at the user's expense. NOTE: Objectionable interference to TV or radio reception can occur if other devices are connected to this device without the use of shielded interconnect cables. FCC rules require the use of only shielded cables.

Canada Warning:

"This digital apparatus does not exceed the Class A limits for radio noise emissions set out in the Radio Interference Regulations of the Canadian Department of Communications." "Le present appareil numerique n'emet pas de bruits radioelectriques depassant les limites applicables aux appareils numeriques (de les Class A) prescrites dans le Reglement sur le brouillage radioelectrique edicte par le ministere des Communications du Canada."

CE Conformance Information:

This device complies with the requirements of the EEC Council Directives: 93/68/EEC (CE Marking); 73/23/EEC (Safety – low voltage directive); 89/336/EEC (electromagnetic compatibility). Conformity is declared to those standards: EN50081-1, EN50082-1.

Important Safety Information

To prevent risk of electric shock: Disconnect power cord before servicing. If fuse replacement is required, please note: For continued protection against fire, replace fuse only with same type and value.

Caution

The installation and servicing instructions in the manual are for use by qualified personnel only. To avoid Electric Shock, do not perform any servicing other than that contained in the operating instructions unless you are qualified to do so. Refer all servicing to qualified personnel.

Electrical Warning

To reduce the risk of electrical shock, do not expose this product to rain or moisture. Keep liquids away from the ventilation openings in the top and rear of the unit. Do not shower or bathe with the unit.

This equipment is designed to be operated from a power source that includes a third "grounding" connection in addition to the power leads. Do not defeat this safety feature. In addition to creating a potentially hazardous situation, defeating this safety ground will prevent the internal line noise filter from functioning.

Ventilation Warning

The Axia StudioEngine and Power Supply/GPIO Nodes require the free flow of air for adequate cooling. Do not block the ventilation openings in the top and rear of the unit. StudioEngine and the Power-Supply/GPIO Node must be mounted with a blank rack spacer above or damage may occur.

Failure to allow proper ventilation could damage the unit or create a fire hazard. Do not place the unit on a carpet, bedding, or other materials that could interfere with the rear and top panel ventilation openings.

Customer Service

We support you...

By Phone/Fax in the USA.

- Customer service is available from 9:30 AM to 6:00 PM USA Eastern Time, Monday through Friday at +1 216.241.7225. Fax: +1 216.241.4103.

By Phone/Fax in Europe.

- Service is available from Axia Europe in Germany at +49 81 61 42 467. Fax: +49 81 61 42 402.

By E-Mail.

- 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.

Trademarks

Telos Systems, Axia Audio, Livewire, the Livewire Logo, the Axia logo, Element, SmartQ, Status Symbols, Zephyr, Zephyr Xstream, Zephyr Xport, Omnia, the Omnia logo, and the Telos logo, are trademarks of TLS Corporation. All other trademarks are the property of their respective holders.

Axia Audio
2101 Superior Ave. Cleveland, OH 44114 USA
+1 (216) 241-7225
Inquiry@AxiaAudio.com

Axia Europe
Johannisstraβe 6, 85354 Freising, Germany
+49 81 61 42 467
Inquiry@AxiaAudio.com

Copyright © 2006 by TLS Corporation. Published by Axia Audio. We reserve the right to make improvements or changes in the products described in this manual, which may affect the product specifications, or to revise the manual without notice. All rights reserved.

©2008 Axia Audio — Rev. 3.0

Notice

All versions, claims of compatibility, trademarks, etc. of hardware and software products not made by Axia mentioned in this manual or accompanying material are informational only. Axia makes no endorsement of any particular product for any purpose, nor claims any responsibility for operation or accuracy.

Warranty

This product is covered by a one year limited warranty, the full text of which is included in the rear section of this manual.

Service

You must contact Axia before returning any equipment for factory service. Axia will issue a Return Authorization number, which must be written on the exterior of your shipping container. Please do not include cables or accessories unless specifically requested by the Technical Support Engineer at Axia. Be sure to adequately insure your shipment for its replacement value. Packages without proper authorization may be refused. US customers please contact Axia technical support at +1 (216) 241-7225. All other customers should contact their local representative to arrange for service.

We strongly recommend being near the unit when you call, so our Support Engineers can verify information about your configuration and the conditions under which the problem occurs. If the unit must return to Axia, we will need your serial number, located on the rear panel.

Credit Where Credit's Due

Deep bows from the waist go to John Makley, Milos Nemcik, Gints Linis, Michael Dosch and the rest of the Axia team in helping to proofread and fact-check this manual. Thanks also to Bruce Wilkinson. Kudos, guys — you are owed more beers than I can-count. - *c.n.*

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 .

About This Manual

This manual covers the details of the Element Studio Control Surface and StudioEngine. However it is assumed in this document that you are familiar with Livewire's basic concepts, as outlined in the companion *Introduction to Livewire: System Design Reference and Primer.*

If you have not done so, please review that material first. In it we explain the ideas that motivated Livewire and how you can use and benefit from it, as well as nitty-gritty details about wiring, connectors, and the like. Since Livewire is built on standard networks, we also help you to understand general network engineering so that you have the full background for Livewire's fundamentals. After reading *Introduction to Livewire* you will know what's up when you are speaking with gear vendors and the network guys that are often hanging around radio stations these days.

New in Manual Version 3.0

This revision, in February, 2008, details the workings of Element software Version 2.0. This version represents a major upgrade, with many new features and capabilities. Look for the "eyes" symbol that highlights new or changed information throughout the text.

Axia Audio, a Telos Company
2101 Superior Avenue
Cleveland Ohio 44114 USA
Phone: +1.216.241.7225
Web: **www.AxiaAudio.com**
E-Mail: **Inquiry@AxiaAudio.com**

©2008 Axia Audio — Rev. 3.0

Table of Contents

Warranty iv
Service iv
New in Manual Version 3.0 iv
A Note From The President of Telos vii
A Note From The President of Axia viii

Quickstart ix
Making Connections ix
Software Setup ix

Chapter One: Setup and Connections 1
Introduction 1
Element Preliminary Setup 2
Connections and IP Configuration 2
"Capturing" Your Console 3
Short Navigation Tutorial 4
IP Address Configuration 5
Clock Configuration 7
StudioEngine Quick Setup 8
Connections and IP Configuration 8
StudioEngine Channel Settings 9
Set Your Console Type 10
Advanced Settings 10

Chapter Two: Configuring Inputs 11
Understanding Source Profiles 11
Source Profile Setup 11
Source Profile Options 11
Adding Backfeeds and GPIO to a Source Profile . 18

Chapter Three: Configuring GPIO 21
GPIO Port Definitions 21
Assigning GPIO to a Source 34
A word about the GPIO Ports 35
Input Connections 35
Output Connections 35

Chapter Four: Element Operations 37
Overview 37
Displays 37
Show Profiles 37
Sources, Channels and Faders 37
Mix-Minus 38
GPIO 38
Monitor Module and SoftKnobs 38
Software 38
Control Callout 38
Element with Standard Monitor Module 39
Element with 2-Fader Phone Module and 2-Fader + Monitor Module 40
Element Stereo Main Screen Overview 41
Control Details 42
General Fader Controls 42
4-Phone Module Channel Controls 43
Call Controller + 2Fader Phone Module Controls 44
Production Module Controls 45
Standard Monitor Module Controls 46
Nav Keys & Control Knob 46
Channel Options Section 46
Delay Section 52
Monitor Section 52
Global Options Section 53
Timer Section 58
Record Mode Section 59
"Magic Key" Sequences for Standard Monitor Module 59
Monitor + 2-Fader Module Controls 59
Channel Options Controls 60
Monitor Section 66
Other Controls 70
Timer Section 71
Record Mode Section 73
"Magic Key" Sequences for Monitor + 2-Fader Module 73
Source-Specific Channel Controls 73
CR Operator Mic Channel Operation 74
Producer Microphone Channel Operation . . . 74
CR Guest Microphone Channel Operation . . . 74
Studio Microphone Channel Operation 74
Line Channel Operation 77
Phone Channel Operation 79
Codec Channel Operation 79

Chapter Five: Show Profiles 85
Creating A Show Profile 85
Build A Show 85
Capture It! 85
Show Profile Options 86

©2008 Axia Audio — Rev. 3.0

The Channel Description Screen 86
Channel Description Section 87
Individual Headphones Section 89
The Auxiliary Send / Return Description Screen . . 89
Monitor Section Screen. 90
General Monitor Options 90
Sources for External 1 & 2 Section 91
Control Room Monitor (Monitor 1) Options . . 91
Control Room (Monitor 1) Headphone Options 91
Studio Monitor (Monitor 2) Options 92
Master Module Control Lock Map 93
Save and Exit 93
Record Mode Screen 93
Record Mode Configuration. 93
Group Start Screen 94
Phone Screen 94

Chapter 6: Virtual Mixer 95
Constructing Virtual Mixes with VMIX 95
StudioEngine VMIX Controls. 96
Element VMIX Controls 97
Enabling Element control of VMIX 97
VMIX Control Description 97
What can you do with a VMIX? 99
GPIO control of your VMIX. 99
PathfinderPC assisted VMIX 99
Manipulating Streams With VMODE 100
What does VMODE do? 100
VMODE Controls. 100
What can you do with VMODE? 102

Chapter 7: Advanced Element Controls 107
Element Configuration Screens 107
The Element Control Center 107
The Setup Menu 107
The Customize Menu 108
The Log and Log History Menu 109
The Screenshot Menu. 109
The Module Information Menu. 109
The Firmware Update Menu 110
CAN Bus Information Manu 110
Brightness Control Menu. 110
Phone Channels Menu 111
User Modules Menu 111
GPIO Configuration Menu 111
Script Information Menu 111
V-Mixer Menu 111
NTP Menu 111
Engine Menu 111
Sources & Profiles Configuration Menu 111

Chapter 8: StudioEngine Controls 113
StudioEngine Configuration Screens 113
The Main Menu. 113
The Fader Channels Menu 113
The Return & Monitor Inputs Menu 113
The Program & Monitor Outputs Menu 114
The VMIX & VMODE Menu 115
Stream Statistics Menu 115
The Engine ID Menu 115
Console ID Menu. 116
System Menu 116
Diagnostics Menu 117

Appendix A: Table of Inputs and Outputs 121

Appendix B: Block Diagrams 123

Appendix C: FAQ / Diagnostics / Maintenance. 135
Installing A New Module 137
Module Diagnostics 140
Fader Cleaning Procedures 141

Appendix D: Working With Phone Hybrids 143
Telos 2101 Multi-Studio Talkshow System 148
Telos Two ISDN Hybrid 152
Telos One and Delta Hybrids 153
Telos ONE-x-Six, Direct Interface and 1A2 Systems 154

Appendix E: Accessory User Panels. 155
Installing a CANBus Accessory Panel 155
Installing a GPIO Accessory Module 156

Appendix F: Channel / IP Worksheets 159

Appendix G: CE Declaration of Conformity. 163

Warranty 165

©2008 Axia Audio — Rev. 3.0

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

©2008 Axia Audio — Rev. 3.0

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...

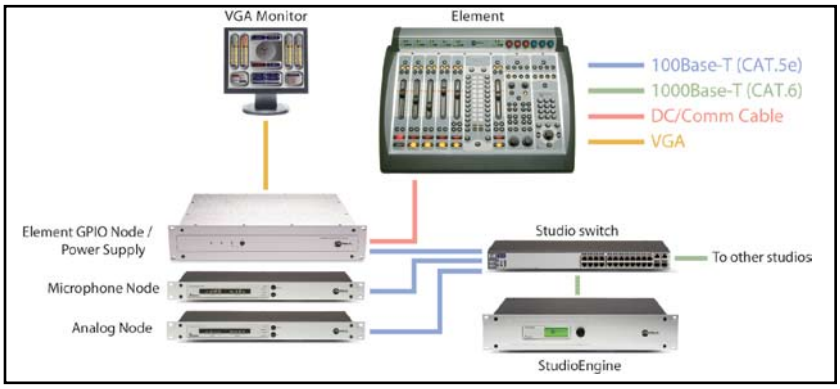
Michael "Catfish" Dosch
February 2004

©2008 Axia Audio — Rev. 3.0

Quickstart

The following chapters of this manual will give you an in-depth understanding of the capabilities of your new Element, from installation to advanced functions.

But maybe you're the sort who really doesn't want to read a manual – you want to open up the boxes and play! This Quickstart section isn't meant to take the place of the following manual chapters, but it will help you get everything connected fast, and point you to the parts of the manual you'll need to get up and running with a minimum of fuss.



Making Connections

You've probably unpacked your boxes and are sitting next to a pile of Axia gear, wondereing what to do first.

Take a look at the diagram above: it represents a typical Livewire-connected radio studio. Here's what you should do to get going:

1. Using CAT.5e or CAT.6 Ethernet cable, connect all of your Axia Audio Nodes (Analog, AES/EBU, Microphone and Router Selector) to 100Base-T ports on your studio's Ethernet switch.
2. Use CAT.6 Ethernet cable to connect your StudioEngine to a Gigabit (1000Base-T) port on the switch.
3. Use another CAT.5e or CAT.6 cable to connect the Ethernet port on the back of the Element GPIO Node/Power Supply to a 100Base-T port on the switch.
4. Packed with your Element is a cable with 6-pin Molex™ connectors. This is the power/communications cable. Lift the Element meter panel and connect the end of the cable *without* the strain relief to the corresponding port on the Element connection panel. Connect the other end of the cable to one of the jacks labeled "48VDC OUTPUT & COMM" on the back of your Element GPIO Node/Power Supply unit.
5. Connect all Audio Node power cords and plug them into your studio's power receptacles.
6. "Capture" your console using the instructions in Chapter 1, Page 3 of this manual.
7. Follow the instructions found in Chapter 1 to configure IP address settings for Element and the StudioEngine.
8. Refer to the User's Manuals packed with your other Axia Audio Nodes for information on how to configure IP addresses for these units.

Software Setup

When you've completed physically connecting your Axia hardware, read through the remainder of this manual to begin configuring the software options that determine how your new Element will operate.

- Chapter 2, "Configuring Inputs," explains how to generate Source Profiles and construct backfeeds for selected sources.
- Chapter 3, "Configuring GPIO," tells how to associate routable logic commands with any audio source.
- Chapter 4, "Element Operations," takes you under the hood to explore in-depth software options.
- Chapter 5, "Show Profiles," illustrates how to set and save unique "snapshots" tailored for specific operational situations that can be recalled whenever the need arises.

Enjoy your new Element!

©2008 Axia Audio — Rev. 3.0

Light shines, metal gleams

the soul of a new machine

Element awaits.

©2008 Axia Audio — Rev. 3.0

Chapter One:

Setup and Connections

Introduction

There's nothing quite as much fun as taking something new out of its box. That sense of excitement as protective packaging is stripped away; that "new gear" smell… it's exhilarating!

If you're reading this, there's a 99.9836% chance that you've done just that, and are now sitting in the middle of a room surrounded by packing material and a large wooden crate, admiring your new Element, its GPIO Node/Power Supply and StudioEngine — and wondering what to do next.

But before you dig in, be advised that *you need to assemble the rest of your Livewire network first.* Because so much of the Element configuration process depends upon having access to audio sources and destinations, you *must* connect and configure your other Axia peripherals (Audio Nodes, GPIO Nodes, Router Selectors, etc.) prior to configuring Element, according to the instructions supplied with each. Once you've done that, come on back and we'll get started.

Be Prepared

This chapter will show you how to get you up and running. We'll install the Element console and its power supply first, followed by the StudioEngine.

Before you dive into the hardware, you'll need to determine a range of IP addresses to assign to each studio and log each device's IP as you assign it; Appendix F contains an IP Assignment Worksheet you can use for this purpose. IP addresses used in an Axia network must be valid Unicast IP addresses. Determine your network's subnet mask settings at the same time (typically this value is set at 255.255.255.0 for intranet applications).

Unicast vs. Multicast IP Addresses: Data is routed over IP networks in one of two ways: point-to-point unicast or multicast.

Livewire devices use common TCP/IP unicast IP addresses for control and web browser access. These are numbers you assign within the range used by your network. Since most Livewire networks are not intended to be accessed via the Internet, we recommend you use the non-routable IP addresses in the range of 192.168.0.0 to 192.168.255.255. These IP addresses have been set aside specifically for use with local networks.

Multicast allows efficient one-to-many connections, so Livewire uses that for its audio streams, source advertising, and synchronization signals. You do not need to assign these multicast addresses because the system does this automatically.

For more information on network construction, please refer to our companion Introduction to Livewire: System Design Reference and Primer available at www.AxiaAudio.com/downloads/ .

To install Element in your studio furniture, you'll need to prepare a cutout in the countertop. A dimensional drawing with all pertinent measurements is found just prior to this manual's Appendix A. Make certain that there's adequate clearance inside the furniture for the required power cable entry into the console; knockout plugs are located at both bottom rear corners of Element for cable routing access.

To configure Element, you'll need a computer. Any Windows desktop or notebook PC with an Ethernet port and a standard Internet browser is acceptable.

For cable connections, CAT. 6 Ethernet cable is required. Axia strongly recommends shielded CAT. 6 if your studio design calls for cable runs through areas containing high levels of RF. (For greater detail on cable selections, we again recommend you read *The Axia Guide to Choosing Category Cable* available from www.AxiaAudio.com/tech/ .)

Always make certain that power has been applied to your studio's Ethernet switch before powering any other Axia components.

©2008 Axia Audio — Rev. 3.0

Element Preliminary Setup

If you haven't done so yet, it's time to get your Element out of its packing crate. This is a two-person job, so grab a partner.

Important: Like any electronic device, Element can be affected by static electricity. Use of a personal grounding device is strongly recommended during transport.

Position one person at each end of Element and, grasping the metal enclosure at the bottom, simultaneously lift upward out of the shipping crate. Remove packing materials and store them in the crate for future use. Carefully lower Element into your counter top cutout, taking care to lower both sides at an equal rate to avoid becoming jammed in the cabinetry.

Important: Element is a drop-in style control surface and requires a countertop cutout. For cutout dimensions, please refer to Appendix A.

Unpack the GPIO Node/Power Supply that came with your Element and place it in its rack.

Connections and IP Configuration

Connections

Cable connections for Element are entirely different from any other mixing surface you may be familiar with. There are no audio inputs; Element needs only a single data / power connection to operate.

Important: Your Element MUST be grounded. Grounding reduces the risk of electric shock by providing a "path of least resistance" for electric current. Improper grounding can result in a risk of electric shock.

Check with a qualified electrician if you are in doubt about how to properly ground this equipment. If your local electrical code prohibits the use of a Station Ground for this purpose, as described above, use the specific "Safety Ground" your local regulations mandate.

1. Rack-mount the GPIO Node/Power Supply unit and connect it to a 100Base-T port on your studio's Ethernet switch using CAT. 6 cable. Do not power up the Power Supply yet.

Figure 1-2: Accessing Element's connection panel.

2. Element **must** be connected to a grounded metal permanent wiring system or other equipment grounding conductor using the threaded grounding stud located inside the connections bay. Open the bay by flipping open the overbridge above the faders, as shown in Figure 1-2. For ground sources, we recommend, in order of preference:
   - » "Station Ground," the heavy copper strap found in the walls and floors of many radio studios.

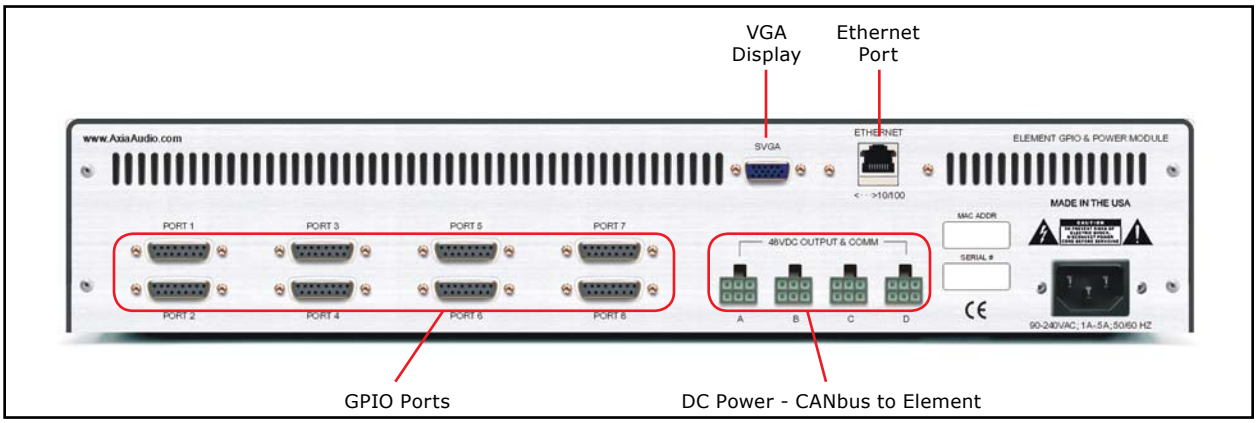


Figure 1-1: Rear panel of Element GPIO/Power Supply showing connections.

©2008 Axia Audio — Rev. 3.0

» AC Safety Ground — the "3rd prong" of a nearby outlet.

» A bonded electrical conduit.

#12 AWG GREEN stranded wire is the *minimum* wire gauge acceptable for grounding Element.

Figure 1-3: Locating the grounding stud.

3. One end of the Element power / communication cable is fitted with a strain relief. Connect this end to any one of the four Molex connectors on the back of the Element GPIO Node/Power Supply. Connect the other end to the Molex socket on the Element connection panel.

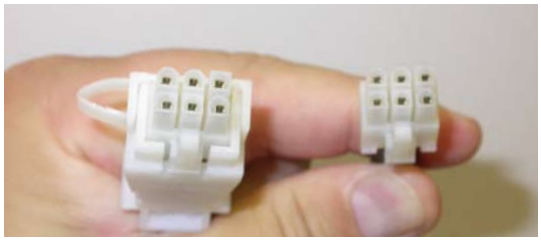


Figure 1-4: Detail of data/communication cable ends.

4. Connect a VGA monitor to the VGA connector on the back of the Element GPIO/Power Supply.

IMPORTANT: Element's power cable is also host to all data communication between the actual surface and its CPU (housed in the Element Power Supply/GPIO Node) by **CANbus** technology; it does not require a separate Ethernet connection. **Never connect any of the RJ-45 ports on the Element connection panel to an Ethernet switch.** These ports are for connection of Element fader and accessory modules only; connecting them to a powered Ethernet switch could damage the board's electronics and **void your warranty.**

Figure 1-5: Element connection panel. Circled connector is for power/data cable; RJ-45's connect to fader and accessory modules.

What is CANbus? Controller Area Network (CAN) is a multicast shared serial bus standard, originally developed in the 1980s by Robert Bosch GmbH, for connecting electronic control units (ECUs). Although initially created for automotive purposes, nowadays it is used in many embedded control applications.

The CAN data link layer protocol is standardized in ISO 11898-1 (2003).

5. Whew, that was hard work. Better have a beer. (We'll have one for you, too.)

"Capturing" Your Console

Upon power-up of your Element, some or all of the alphanumeric displays above the faders may read INACTIVE. This means that the Element CPU must be initialized. To do this, you'll need to enter Capture Mode. The way you'll do this depends on whether the standard Monitor Module or the Monitor + 2-Fader Module is installed in your Element.

Figure 1-6: Identifying the Standard Monitor Module (left) and Monitor + 2-Fader Module (right).

"Capturing" using the Standard Monitor Module

Press and hold the **Help** key on the Monitor Module for 5 seconds. The alphanumeric displays will change

©2008 Axia Audio — Rev. 3.0

to read Capture Mode; this enumerates all of the installed faders in the control surface.

Now, make sure that LED displays above the faders display the channel numbers in the order you desire. If not, you can adjust them individually using the **PGM1** and **PGM2** buttons above each fader to change the channel number to the one you desire. Note that even though the hardware will let you assign channel numbers as high as 64, numbers greater than 28 should not be used.

When you're done adjusting displays, press the ↵ **(enter)** key to exit Capture Mode. Your Element is ready to use.

'Capturing" using the Monitor + 2-Fader Module

Press and hold both the *** (star)** and **"2"** keys on the Monitor Module numeric pad for 5 seconds. The alphanumeric displays will change to read Capture Mode; this enumerates all of the installed faders in the control surface.

Now, make sure that LED displays above the faders display the channel numbers in the order you desire. If not, you can adjust them individually using the **PGM1** and **PGM2** buttons above each fader to change the channel number to the one you desire. Note that even though the hardware will let you assign channel numbers as high as 64, numbers greater than 28 should not be used.

When you're done adjusting displays, press the **# (pound)** key to exit Capture Mode. Your Element is ready to use.

Hello Moto: If you have motorized faders, you can "mirror" other motorized faders while you are in Capture Mode. A motorized fader that is "slaved" to a master fader will move automatically to mirror the movements of the master. Mirrored faders both control gain for the same source, but the master can override the setting of the slave — useful for "split" Element setups where talent has a local set of faders that may be used to control certain sources, but you still want the Board Operator to have master control.

To mirror a fader, press the **Options** key above the fader you wish to make the "slave". An apostrophe (') will be displayed next to its numeric indicator. Pressing the fader's **PGM1** and **PGM2** keys will change the numeric indicator, indicating the "Master" to which this fader is now slaved; a display reading 2' means that the fader is slaved to fader 2.

Pressing the **Options** key again returns the fader to non-mirrored mode.

Short Navigation Tutorial

There are several different ways to change on-screen options. The first way is with Element's SoftKnobs, located directly above the Monitor Module and shown in Figure 1-7.

Figure 1-7: Element SoftKnobs.

These knobs are context-sensitive. When any of Element's setup modes are activated using the options keys on the faders or Master Module, you'll see a corresponding row of "knobs" represented onscreen, as shown in Figure 1-8.

Figure 1-8: On-screen SoftKnobs.

Note that there are sometimes functions shown both above and below the on-screen SoftKnobs. When a function is presented above the knob, turning that knob accomplishes that function. When a function is presented below the knob, pushing in gently on the corresponding SoftKnob activates that function. The example shown in Figure 1-8 is from the IP Address Book screen.

If your Element is equipped with the standard Monitor Module, you can also navigate using the control knob, directional keys and Enter key located at the lower right-hand corner of the module, as shown in Figure 1-9.

©2008 Axia Audio — Rev. 3.0

Additionally, whenever numeric input is required (like entering IP addresses), you can use the numeric keypad located just above the Navigation Keys (Figure 1-10) to enter numbers directly.

Figure 1-10: Master Module numeric keypad.

IP Address Configuration

Using your IP Assignment Worksheet, you'll now choose an IP address for both your Element and StudioEngine. You must make sure that IP addresses for both Engine and Surface are in the same IP network, and are both valid Unicast addresses. For example:

» 192.168.2.101: Engine
» 192.168.2.201: Surface

Both addresses belong to 192.168.2.x IP network, and so will work for our application.

Note: "Gateway" settings on Livewire equipment are optional. They may be left blank unless you intend to access the StudioEngine's configuration utility remotely, from outside your network. Should you desire to do so, enter the IP address of your master router (the one with external network connections) whenever you're given the option to enter a gateway IP address.

Note: Element v2.x software supports redundant NTP servers. Use the second field in the IP Address Book to enter the address of a backup server.

Once you've determined the IP addresses you'll assign, turn on the Element power supply and the attached VGA monitor.

Setting IP using the Standard Monitor Module

When Element has completed its start cycle (meter screen appears), find the section marked **Global Options** on the Monitor Module. Press and hold the key marked **Fader Options** located in the top right corner of the module for five seconds to enter the IP Address Book (Figure 1-11).

Figure 1-11: IP Address Book, Standard Monitor Module

To set the IP address, rotate the leftmost SoftKnob to highlight the line titled "Element IP Address" and push the knob to select that line. Now use the numeric pad to enter the IP address. Press the leftmost SoftKey again to "take" the new IP value. Repeat this process to set the subnet mask value.

Continue down the screen and, in the same way, enter the other fields' values as appropriate.

- 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:
  » **Element IP Address:** Every piece of Livewire gear must have its own unique IP Address. Set the IP Address for each Element in this field.

©2008 Axia Audio — Rev. 3.0

» **Subnet Mask:** Enter your network's subnet mask value here, usually 255.255.255.0

» **Gateway IP Address:** To enable remote administration of your Livewire network, you'll need a network gateway. If you have one installed, enter its IP Address value here.

» **Engine IP Address:** Each Axia control surface controls a unique Studio Engine. Enter the IP of the controlled Engine here.

» **1st NTP Server IP Address:** Element supports NTP (Network Time Protocol) timekeeping to synchronize time-of-day clocks across your facility. If you have an NTP server, enter its IP Address here.

» **2nd NTP Server IP Address:** Element v2.x software supports redundant NTP servers. Enter the IP Address of your backup server here.

» **Web Page User Password:** You can password-protect your Element's webserver connection by entering a numeric password here.

Note: If you set a password for HTTP access, you will not see your password displayed in this field upon re-entering the IP Address book. This is a security feature intended to prevent "over-the-shoulder" password theft.

- If you want to reverse changes you've made and haven't saved, press the **Undo** knob to erase them.
- Press **Reboot** to immediately reboot the Element.
- Press **Save** to save changes you've made.
- Press **Exit** to leave the IP Address Book.

Note: After making changes to any Element IP address settings, you must reboot the console. Use the "REBOOT" soft key to begin an immediate reboot sequence.

Remember, you can even reboot Element while on-the-air, since the audio mixing is not done in the console but in the Studio Engine; your audio will continue to play without interruption.

Setting IP using the Monitor+ 2-Fader Module

When Element has completed its start cycle (meter screen appears), find the numeric keypad at the top right of the Monitor Module. Press and hold the *** (star)**, **"7"** & **"4"** keys simultaneously for five seconds to enter the IP Address Book (Figure 1-12).

Figure 1-12: IP Address Book, Monitor Module

To set the IP address, rotate the leftmost SoftKnob to highlight the line titled "Element IP Address" and push the knob to select that line. Now use the numeric pad to enter the IP address. Press the leftmost SoftKey again to "take" the new IP value. Repeat this process to set the subnet mask value.

Continue down the screen and, in the same way, enter the other fields' values as appropriate.

- 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:
  - » **Element IP Address:** Every piece of Livewire gear must have its own unique IP Address. Set the IP Address for each Element in this field.

©2008 Axia Audio — Rev. 3.0

» **Subnet Mask:** Enter your network's subnet mask value here, usually 255.255.255.0

» **Gateway IP Address:** To enable remote administration of your Livewire network, you'll need a network gateway. If you have one installed, enter its IP Address value here.

» **Engine IP Address:** Each Axia control surface controls a unique Studio Engine. Enter the IP of the controlled Engine here.

» **1st NTP Server IP Address:** Element supports NTP (Network Time Protocol) timekeeping to synchronize time-of-day clocks across your facility. If you have an NTP server, enter its IP Address here.

» **2nd NTP Server IP Address:** Element v2.x software supports redundant NTP servers. Enter the IP Address of your backup server here.

» **Web Page User Password:** You can password-protect your Element's webserver connection by entering a numeric password here.

Note: If you set a password for HTTP access, you will not see your password displayed in this field upon re-entering the IP Address book. This is a security feature intended to prevent "over-the-shoulder" password theft.

- Rotating the **Action** knob choosees between the following options:
  » **Save** stores the changes you've made to the IP Address Book.
  » **Undo** reverses any changes you've made if you haven't yet saved them.
  » **Reboot** immediately reboots the Element.

- Press the **Exit** knob to leave the IP Address Book.

Note: After making changes to any Element IP address settings, you must reboot the console. Use the "REBOOT" soft key to begin an immediate reboot sequence.

Remember, you can even reboot Element while on-the-air, since the audio mixing is not done in the console but in the Studio Engine; your audio will continue to play without interruption.

Clock Configuration

Locate the Timer section of the Monitor Module; press and hold the key marked **Options** to enter the Clock & Timer Options screen (Figure 1-13).

Figure 1-13: Clock & Timer Options menu screen.

- Rotate the **Select/Set** SoftKnob to scroll through the list of available options. To select an option to modify, push this knob. Rotate the knob to change an option's value, and push the knob when you're done to move to other fields. 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

©2008 Axia Audio — Rev. 3.0

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.

» **Show Tenths On:** Allows you to set the countdown and elapsed-time counters to show or hide tenths-of-a-second displays.

A Note About Metering: Element's bargraph meters feature true VU meter ballistics, though with an expanded scale. The red line at -20dBFS corresponds to the 0dB mark on a traditional analog meter. Just as with a traditional analog meter, the 300ms time-constant filter specified in the VU standard means that peaks are much higher than the meter indicates. Depending upon the nature of the program material, transient peaks will be 9-15dB higher than the VU-filtered average. This is why modern consoles, including Element, have 20dB headroom above the nominal operating level. Some of this headroom is used to cover the peak transients the VU meter can't see, and some is used for operator error or surprise margin. (ie, the well-known "excited sportscaster scream".)

By default, Element meters are set to display both average and peak levels, the average being represented by the main bargraph display, and the peaks represented by the small floating line that appears above them.

» **Show Meter Peaks:** Allows you to turn Element's Peak meter display on or off. **Yes** displays peaks; **No** does not.

- Push the **Save** knob to commit to the changes you've made. You may do this at any time, after changing any option to apply the change immediately.
- Push the **Exit** knob to leave the Clock & Timer Options screen.

The next section will walk you through setting up your StudioEngine.

StudioEngine Quick Setup

Connections and IP Configuration

Connections

Note: We recommend **leaving 1RU of "breathing space"** for ventilation above and below the StudioEngine when you rack-mount the unit.

Using the supplied AC power cord, connect the StudioEngine to the mains. Route a length of CAT- 6 cable from StudioEngine to a **Gigabit** port on your studio's Ethernet switch, but don't connect it yet.

IP Configuration

1. Power on your StudioEngine; the front-panel display will indicate the boot-up sequence. When the screen displays IP address and network connection information, push the control knob to call the Main Menu.
2. Turn the control knob to highlight Option #4, `IP Settings` (Figure 1-14), and push to select.

Figure 1-14: StudioEngine IP configuration screen.

3. Push the control knob and you'll see that the first line of the display (marked `Net Addr`) is underlined. Push the knob again to select this line.
4. You'll see a flashing underline under the first digit

©2008 Axia Audio — Rev. 3.0

of the IP address. Push and rotate the control knob to select the value for this digit. Push again to "take" the value.

5. Continue this sequence until your IP address is properly entered. When finished, turn the control knob until the check mark at the end of the line is highlighted, and push the knob to apply the new IP address value.
6. When you've set the subnet mask and gateway IP address, choose OK at the bottom of the screen.
7. You'll be asked if you'd like to reboot for changes to take effect. select YES and connect the Ethernet cable to your StudioEngine.

Secret Button Function: Here it is, the first of many "secret" controls hidden about Element and meant for your Engineering eyes only.

If, during bootup, you press the StudioEngine's control knob within 6 secs of the initial screen display, you'll be asked if you want to reset the front panel and HTTP passwords to factory default. This is your "backdoor" into the system should you forget the login password to the front panel. Changing these passwords forces a reboot.

StudioEngine Channel Settings

For the next steps we'll be using the StudioEngine's HTTP interface, so connect a computer loaded with an Internet browser to the local switch. This computer should have an IP address in the same range assigned to the Element and StudioEngine.

Note: Axia web interfaces have been tested with Microsoft Internet Explorer, version 5 or later, but may also work with other browsers.

Whatever browser you choose, **Java must be enabled and your pop-up blocker, if any, must be disabled** in order to work with Axia equipment.

Note that Windows Vista versions *do not* have Java installed by default as in prior versions. You must install the Sun Java package found at http://www.java.com .

In a Livewire network, individual devices (audio nodes, studio controllers, etc.) are identified by unique IP addresses. But what about the audio streams these devices generate? Think about how many audio sources and destinations there are in even a small studio — there are a lot to keep track of. In the analog days, we'd affix a numbered label to each cable that entered the terminal room to identify each audio circuit. Obviously, we can't attach pieces of paper to digital packet streams, but we can give each one a "label."

We refer to these "labels" as Network Channel Numbers. Each Livewire system can support 32,766 channels of audio, which enables us to give each audio source or destination its own unique numeric channel number.

Note: 32,000+ channel numbers exceeds the average human's RAM storage capacity. We recommend using a spreadsheet program — just to make sure you don't generate painful read-memory register errors.

As soon as you connected your StudioEngine to the network (following the steps in the previous section), it began to generate audio using **factory default channel numbers.** To avoid conflict with other studios, you *must* change those default values to new channel numbers.

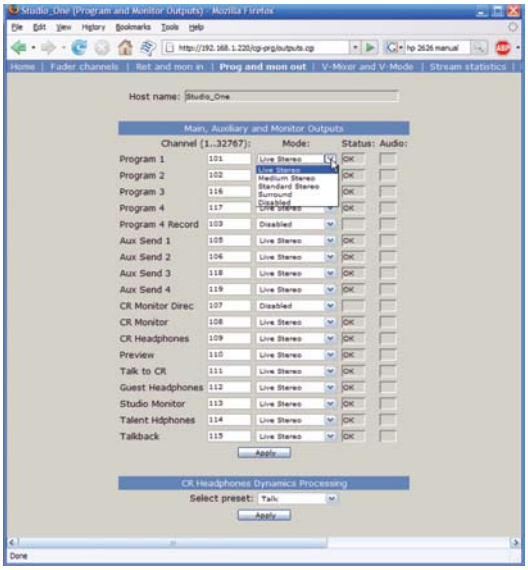


Figure 1-13: StudioEngine Output configuration

1. Open the browser on your computer and point it to the IP address you previously set for your StudioEngine. Choose the **Program And Monitor Outputs** menu option to configure the output buses of the Element.

©2008 Axia Audio — Rev. 3.0

2. You'll be prompted to enter a user name and password. The default user name is "user". Leave the password field blank.
3. You'll see the screen shown in Figure 1-13, This screen allows you to set channel numbers for all of the audio outputs generated by Element; program and record busses, monitor feeds, and talkback destinations. Work down the list, assign each one a unique channel number and record them in your channel number log.
4. Find the column entitled **Mode**. Each Studio Engine is shipped with the most common streams enabled with **Live Stereo** mode. Make sure all of your desired streams are enabled.
5. Choose **Apply**.

A word about stream types: There are several different types of streams available in the **Mode** dropdown box: **Live Stereo, Medium Stereo, Standard Stereo** and **Surround.**

Live Stereo was previously referred to as **Livestreams**. Use Live Stereo streams whenever you need the lowest latency for on-air feeds, mic-to-headphone paths, etc.

Medium Stereo is our recommended default for everything else. This setting balances low latency with high compatibility for systems containing PC playout systems.

Standard Stereo is intended to provide the best compatiblity with older PC systems, but the tradeoff is that these streams have a higher amount of latency.

Surround Streams: Element can mix both Stereo and Surround (5.1) audio. If you're feeding the system with a 5.1 audio source, choose the **Surround** option.

Set Your Console Type

Axia StudioEngines will work with different types of consoles, so you need to tell it which type you're pairing it with.

1. Navigate to the "Engine ID" menu and find the "Application Interface Options" section. Be sure that **Element** is checked for the "Console Type".
2. Click **Apply**.

Note: Manual setting of console type is only necessary on Element consoles running software versions prior to v2.0. In Elements running v2.0 or higher, this setting will auto-detect.

Advanced Settings

You've no doubt noticed that there are many more options for your Element and StudioEngine than we've covered here. That's because Chapter One was meant to get you up and running as quickly as possible. For an in-depth explanation of the other available settings, please refer to Chapters 7 & 8 of this manual.

What's Next

Take a break! You're done with initial set-up. When you're ready, continue to the next chapter to walk through one of the most important Element concepts: Source Profiles.

©2008 Axia Audio — Rev. 3.0

Chapter Two:

Configuring Inputs

In this chapter we'll discuss how to set up inputs for your Element. You'll be using the HTTP servers built into the Element and StudioEngine to do this.

> ***Note:*** Axia web interfaces have been tested with Microsoft Internet Explorer, version 5 or later, but also work with other browsers.
>
> We also like Mozilla's FireFox browser, especially with regards to its "tabbed browsing" capabilities. With tabbed browsing, it's possible to bookmark the IP addressees of entire rooms worth of Audio Nodes, then recall them in tabbed groups with a single click.
>
> Whatever browser you choose, Java must be installed and enabled, and your pop-up blocker must be disabled. The latest Java package is available at http://www.java.com .

Understanding Source Profiles

One of the most repetitive and tedious parts of studio building has always been tying "data" to audio sources. Consider the wiring of line selector modules: you need a cable for the remote audio source, a set of wires for a "start" closure, another set for "stop" — even with TDM routers, an operation as seemingly simple as logic-follows-audio often turns out to be not so simple.

Mix-minus and IFB generation is also a problem. Too often it's a complicated process that requires air talent to correctly determine source and bus assignments at a moment's notice. How many times has unwanted audio made air thanks to this error-prone procedure?

Element eliminates these headaches by letting you merge audio, logic and program data into a single, routable information stream. Since audio in a Livewire network is transported as packetized data, it's easy for us to "piggyback" other data along with audio. This is accomplished through the use of Source Profiles.

Element Source Profiles allow you to assign modifiers to an audio input. Characteristics like EQ curves, Pan trim, voice dynamics, even mix-minus source selections, can all be set in the Source Profile and are then automatically loaded whenever that source is assigned to an Element fader.

The result of all this is that Element is always presented with complete, consistent information about how you want a given audio source handled.

> ***Note:*** Source Profile settings do not transfer between studios. In other words, a mic in Studio A that has EQ and dynamics applied through a Source Profile on the Studio A Element will not have those settings automatically applied if that mic is loaded on the Element in Studio B. We are, however, planning on this capability in future versions.

Source Profile Setup

Now that you understand what Source Profiles do, let's set one up. You'll use the Element and StudioEngine HTTP interfaces to do this, so connect a computer with an Internet browser to the local switch.

> ***Note:*** You can't construct Source Profiles unless you've already configured the inputs and outputs of your studio's Audio Nodes. Be sure you've assigned Livewire channel numbers and enabled all sources in your Nodes, Engine, and PC drivers. This will make your sources visible on the selection list in the source profile setup screen. Make sure you've done this before proceeding further through this chapter.

Your computer must be assigned a valid network IP address in order to "see" Livewire web pages. Livewire networks do not contain DHCP servers, so you must manually assign your computer an IP address within the range you previously determined for your network. Different operating systems use different methods to do this; please consult your OS's documentation for specific instructions.

Source Profile Options

1. Enter the IP address of your Element in your browser;

©2008 Axia Audio — Rev. 3.0

you'll see the "Element Control Center" displayed onscreen. Using the sidebar menu, choose the "Configuration" menu item under the "Sources & Profiles" heading. Enter your password in the box (default is "user") and click **OK**.

2. Choose the **Source profiles** menu item from the main screen, and click the **New Source Profile** link at the bottom of the screen. You're now in the Source Profile creation screen.

> ***Reminder:*** Did you define your audio sources from the Engine and Nodes as outlined in Chapter One? If not, go no further – you've got to do that before you can continue with Source Profile setup.

- **PRIMARY SOURCE:** All of the audio sources that you previously defined (during Audio Node setup) are available in this drop-down box; pick one.

- **SOURCE PROFILE NAME:** Type a 10-character descriptive name for your source.

- **SOURCE NAME OVERRIDE:** If the source you're defining can supply Program Associated Data (for instance, a computer playout system that sends song/artist information), you can choose to have that text dynamically substituted in the alphanumeric display of the fader the source is assigned to. Instead of reading `PROPHET_1`, for example, it could scroll `FERGIE`.
  - » Choose **Show Sourcename** to have the channel alpha display the name you typed into the Source Profile Name field above.
  - » Choose **Allow Application to Override** if you'd like the PAD text displayed instead.
  - » Choose **Show Both** to scroll the Source Name followed by the PAD text across the alpha display.

- **SOURCE AVAILABILITY:** These check boxes determine which faders (or other inputs) you'll allow this source to be assigned to. For example, if your source is a Control Room mic, and you only want it to appear on the left-most fader, you would check only the box marked **Channel 1**. You might want to make a CD player available on any fader, so you'd check the boxes for all faders. Some sources – especially air monitors – you might wish to prevent from ever being assigned to a fader, so you would uncheck the boxes for all faders. The number of fader checkboxes show will vary depending upon the number of faders installed in your Element.

The four final check boxes, **CR monitor input**, **CR headphone input, Studio monitor input** and **V-Mixer input** determine whether the board op will be able to assign this source to the monitor and headphone buses or the Virtual Mixer (please see Chapter 6: Virtual Mixer for details on this feature).

To prevent a source from appearing on these inputs' selection lists, uncheck the desired box.

> ***About Mics and Headphone Feeds:*** Element provides you with several headphone options, all selectable from Element's Monitor Module. The *Monitor 1* (control room) headphone output provides a feed intended for the Host (board-op's) mic position and includes the ability to monitor the Preview bus in headphones. Guests can talk to the host directly using the **Talkback** keys on their remote control panels.
>
> The *Monitor 2* headphone output sends audio to talent and guest mic positions. Talent positions include a talkback channel that the host or board op can use to communicate using the **Talk To Monitor 2** function (guest positions do not have this Talkback channel).
>
> An *individual headphone feed* can also be constructed to associate a dedicated headphone channel with a specific mic. This allows the board op to send individual, targeted talkback to a specific individual (using the **Talkback** keys) without other talent or guests hearing.

- **SOURCE TYPE:** Define your source as a microphone, a line input, a telephone input, a codec or a computer source.
  - » **Operator** is the board operator's mic. It is the source mic for Element's **TALK TO** functions. Its **On** button serves as a **COUGH/MUTE** button. Its associated logic mutes the CR monitors when **On**.
  - » **CR Producer** is used for in-studio Producer's mic positions. It has associated GPIO logic

©2008 Axia Audio — Rev. 3.0

which can operate **Talk to...** functions from a remote producer's panel. It also mutes CR monitors when **On.**

» **CR Guest** is used for any other guest mic in the control room. Its associated logic port mutes the CR monitors when **On**. It can have an individual headphone feed.

» **Studio Guest** is used for any mic located in a separate studio. Its logic mutes the Studio monitors when **On**. It can have an individual headphone feed.

» **Line** is used for any line input audio source, analog or digital. A GPIO logic port can be used to provide machine start/stop pulses if desired.

» **Phone** defines this source as a hybrid or broadcast phone system input. Summed mono mix-minus provided.

» **Codec** marks this source as a codec. Dual-mono mix-minus provided; one PA feed + one talent feed with talkback.

» **Computer Player** defines this source as originating from a PC. Associated GPIO logic can be used to send start/stop signals to a playout system.

About Telos Phone System Modes: Telos TWOx12, Nx12 and Series 2101 broadast phone systems work in two different modes: **US Mode** and **Euro Mode.**

In US Mode, one of the phone system's hybrids is assigned to a console faders, and taking a caller to air is accomplished by using a switching control to select a line on that hybrid and turning its fader **ON.**

In Euro Mode, the swiching control is not used; instead, each line from the phone system is permanently assigned to a separate fader, which may be turned **ON** or **OFF** at will.

Element supports both US and Euro modes. The next few settings are used for setting up the desired style of phone operation.

The next 4 options are used for EURO mode only:

- **PHONE HYBRID IP:** If this source is a Telos Nx12, TWOx12, Series 2101 or Telos TWO and you will be operating in Euro mode, enter the IP address of the phone system.

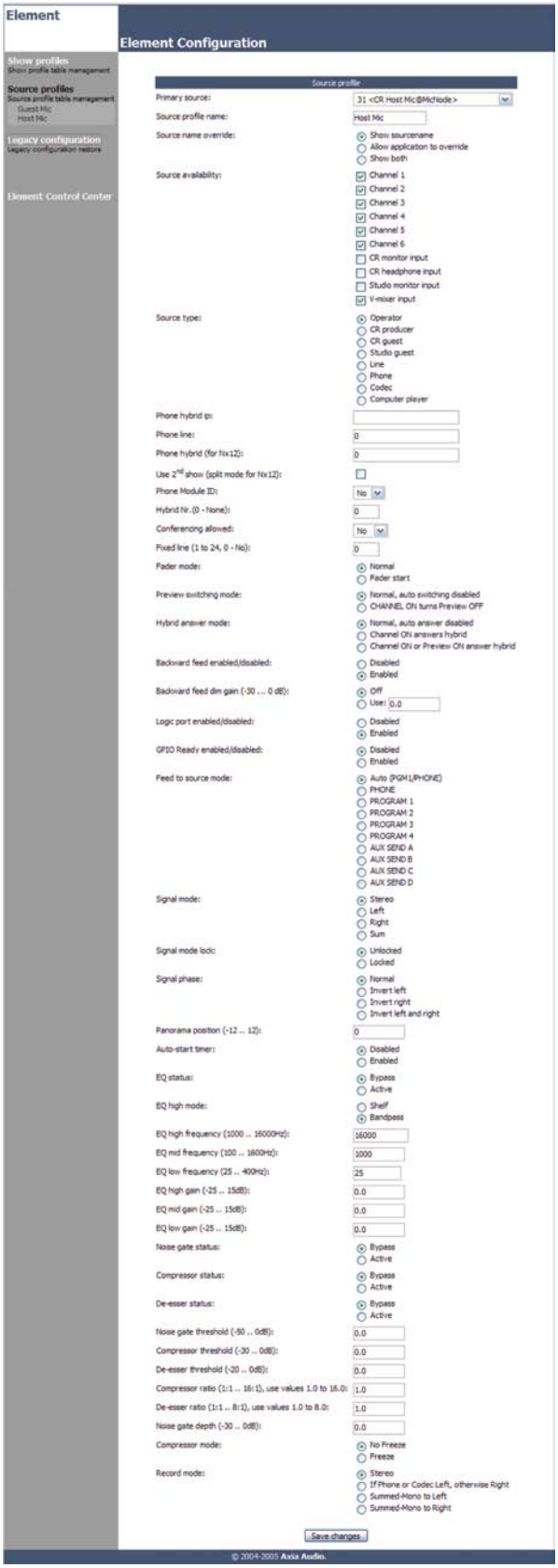


Figure 2-1: Source Profile options (yes, there are a lot!)

©2008 Axia Audio — Rev. 3.0

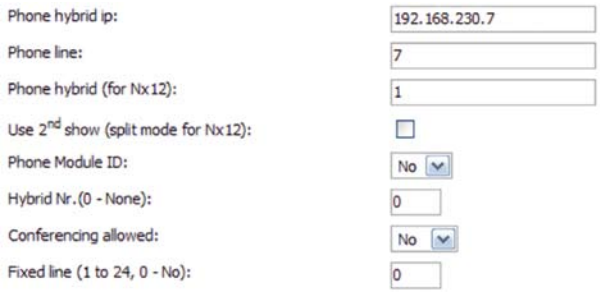


Figure 2-2: Settings for configuring Element for operation with Telos phone systems

- **PHONE LINE:** If this source is a Telos Nx12, TWOx12, Series 2101 or Telos TWO and you will be operating in Euro mode and wish to dedicate a phone line to this fader, enter it here.

- **PHONE HYBRID (FOR NX12):** Even though the option title says "for Nx12", if this source is a Telos Nx12, TWOx12 or Series 2101 and you will be operating in Euro mode, you may specify hybrid 1 - 4 for this line assignment.

- **USE 2ND SHOW (SPLIT MODE FOR NX12):** Even though the option title says "for Nx12", if this source is a Telos Nx12, TWOx12 or Series 2101 and you will be operating in Euro mode, you may check this to designate Split Studio Mode, where two hybrids are assigned to one show, and the other 2 assigned to a second show. Element will connect to the second Show in the list.

The next 4 options are used for US mode only:

- **PHONE MODULE ID:** For Telos Nx12, TWOx12 or Series 2101 operating in US mode. If you are constructing a Source Profile for a telephone hybrid and have multiple Phone Modules installed, this allows you to determine which module will control this hybrid. For details on setting up a phone system with Element, please refer to Appendix D: Working With Phone Hybrids.

- **HYBRID NR.:** For Telos Nx12, TWOx12 or Series 2101 operating in US mode. Specify which hybrid this Source Profile will control.

Nx12 Users Note: For Nx12's second show, you would STILL enter "1" for Hybrid 3 and "2" for Hybrid 4. Why? Because, to Nx12's internal logic table, Hybrid 3 is really the first hybrid in Show 2 and Hybrid 4 is the second hybrid in Show 2.

- **CONFERENCING ALLOWED:** For Telos Nx12, TWOx12 or Series 2101 operating in US mode. Specifies whether or not "button mash" conferencing is permitted. Select **Yes** to allow, **No** to prohibit.

- **FIXED LINE:** For Telos Nx12, TWOx12 or Series 2101 operating in US mode. When working with a Telos hybrid that supports it, this command lets you assign a line permanently to a specific fader, emulating a dedicated hybrid.
  - » **0** means there is no fixed line assigned.
  - » **1 - 24** slaves the specific line to this source's fader.

Note: For complete instructions on setting up a phone system with Element, please refer to Appendix D: Working With Phone Hybrids.

- **FADER MODE:** Allows you to set the channel activation method of the fader this source is assigned to.
  - » **Normal** conforms to the US method of requiring talent to manually turn the channel on and off.
  - » **Fader Start** follows the common European standard of activating the channel and associated machine logic when the fader is raised from ∞.

- **PREVIEW SWITCHING MODE:** Determines what will happen to a source's off-air **Preview** assignment when that source is taken to air.
  - » **Normal, Auto-Switching Disabled:** The **Preview** bus assignment will be retained even after the fader is turned **ON.**
  - » **CHANNEL ON turns Preview OFF:** If this source is assigned to **Preview** for off-air auditioning, turning its channel **ON** will cancel its **Preview** assignment.

- **HYBRID ANSWER MODE:** This setting allows you to tailor Element's Hybrid Answer mode to suit your facility's operating style.

©2008 Axia Audio — Rev. 3.0

- » **Normal, Auto Answer Disabled:** SmartSurface provides no auto-answer logic when a fader with a hybrid assigned is turned **On.**
- » **Channel ON Answers Hybrid:** When a hybrid has an incoming call, turning its assigned fader **ON** will answer the call.
- » **Channel ON or Preview ON Answers Hybrid:** When a hybrid has an incoming call, turning its assigned fader **ON** or pressing that channel's **Preview** key will answer the call.

- **BACKWARD FEED ENABLED/DISABLED:** Allows you to enable or disable backfeeds to this audio source.

- **BACKWARD FEED DIM GAIN:** This is a feature unique to Element: Talkback is sent over dimmed backfeed audio. If you want a hard interruption (Talkback interrupts Program audio), set it to maximum dim, which mutes the audio. But maybe your jocks would like to just have their program audio dimmed, and the Talkback come over it so they can still monitor air while they're being talked to. In this case, input a value by which the program audio will be dimmed. Adjustment range is 0dB to -30dB.

- **LOGIC PORT ENABLED/DISABLED:** Allows you to enable or disable GPIO machine logic to this audio source device.

- **GPIO READY ENABLED/DISABLED:** Allows devices such as CD players or tape machines to signal their readiness to the operator through the OFF button LED. If a device is cueing, its STOP lamp will not be lit. Tying the device's stop lamp to the Ready command means that a machine must be cued and ready before the OFF lamp illuminates.
  If **Enabled**, the OFF lamp illuminates only when the fader is OFF *and* the Ready command is active on the GPIO. If **Disabled**, the OFF lamp illuminates normally whenever the fader is turned OFF.

- **FEED TO SOURCE MODE:** If there is a backfeed associated with this source, this option determines what content is fed back from the board.
  - » **Auto (Pgm-1/Phone)** feeds the Program 1 bus (minus the source) when this source's fader is ON, and feeds the Phone bus (minus the source) when the fader is OFF. (Program 1 and 2 feeds are post-fader; Phone feed is pre-fader to allow speaker-phone style operation thru the Operator's mic.
  - » **Phone** feeds the Phone bus, minus the source.
  - » **PGM1** feeds the Program 1 bus, minus the source.
  - » **PGM2** feeds the Program 2 bus, minus the source.
  - » **PGM3** feeds the Program 3 bus, minus the source.
  - » **PGM4** feeds the Program 4 bus, minus the source.
  - » **AUX SEND A** the Aux Send A bus, minus the source.
  - » **AUX SEND B** feeds the Aux Send B bus, minus the source.
  - » **AUX SEND C** feeds the Aux Send C bus, minus the source.
  - » **AUX SEND D** feeds the Aux Send D bus, minus the source.

About the PHONE, RECORD and PROGRAM 4 buses: Element's PGM-4 mix bus is a special multi-purpose bus that does several things at once to make life easier for the operator.

Sources assigned to PGM-4 are also sent to the RECORD bus, a special post-fader, pre-on/off output that can be used to feed recording devices.

At the same time, sources assigned to PGM-4 are mixed and sent to the PHONE bus. PHONE is an internal bus which does not feed a separate output; instead, it picks up PGM-4 audio pre-fader and pre-On/Off and backfeeds it callers (minus themselves, of course) when the channel is **OFF** and **AUTO** is selected as the **Feed To Source Mode**.

- **SIGNAL MODE:** Determines whether source will be treated as mono or stereo, and how.
  - » **Stereo** feeds incoming L/R signal to left and right channels of assigned bus(es).
  - » **Left** feeds incoming left channel to both channels of assigned bus(es).
  - » **Right** feeds incoming right channel to both channels of assigned bus(es).

©2008 Axia Audio — Rev. 3.0

» **Sum** creates L+R mono mix of incoming stereo source and feeds it to both channels of assigned bus(es).

- **SIGNAL MODE LOCK:** Allows or disallows user's ability to change the source's signal mode (as set above) using controls on Element's surface.
  - » **Unlocked** allows the user to change modes if needed.
  - » **Locked** prevents changes to the signal mode.

- **SIGNAL PHASE:** Allows for correction of out-of-phase program material.
  - » **Normal** assumes correct phase polarity of source input material.
  - » **Invert left** reverses phase on the left channel input.
  - » **Invert right** reverses phase on the right channel input.
  - » **Invert left and right** reverses phase on both channel inputs.

- **PANORAMA POSITION:** Lets you preset pan settings for this input. Pan is variable in 25 steps, center being **0**, far left **-12** and far right **12**. This setting can be adjusted on the fly by the operator.

- **AUTO-START TIMER:** Determines whether the event timer found in Element's right-hand meter display will reset to zero when this source's fader is turned on.
  - » **Disabled:** timer will not reset when this source is activated.
  - » **Enabled** timer resets when the source is activated.

About Element EQ: Equalization is available for all audio sources, and can be pre-defined or adjusted on-the-fly. Our EQ model is three-band quasi-parametric. Controls are provided for center frequency and boost/cut, with a unique SmartQ™ automatic bandwidth system.

SmartQ works by varying the Q of the selected parametric band proportional to the amount of boost or cut you specify. A small amount of boost or cut will affect a broader range of frequencies for a warm, musically-pleasing effect. As gain is increased the Q sharpens, affecting a narrower range of frequencies for tighter control of target bands. At aggressive cut levels, the EQ becomes a tunable notch filter. SmartQ keeps the EQ sounding natural at virtually all settings.

Pre-defined EQ can be entered during Source Profile construction, and will be automatically applied whenever that source is assigned to a Element fader; i.e., a -2 dB 12 kHz high shelf filter pre-set in a microphone's Source Profile will be applied when any surface, in any studio, loads that mic for use.

At-will EQ is set by talent using Element's **Channel Options** controls, and can be employed to trim EQ on-the-fly .

- **EQ STATUS: Bypass** or make **Active** EQ for this source.

- **EQ HIGH MODE:** Choose between **shelf** or **band pass** filter for high-frequency EQ.

- **EQ HIGH FREQUENCY:** Sets the active frequency for high-band EQ. If **EQ HIGH MODE** is set to **Shelf**, this setting determines the top of the shelf. If **EQ HIGH MODE** is set to **Band pass**, this sets the center frequency of the BP filter (SmartQ determines the width of the BP filter, as described above).

- **EQ MID FREQUENCY:** Sets the center frequency of the midrange parametric band.

- **EQ LOW FREQUENCY:** Sets the center frequency of the low parametric band.

- **EQ HIGH, MID & LOW GAIN:** Provides 40 dB of adjustment range for boost or cut of the three parametric bands, from -25 dB to +15 dB.

The following options dealing with dynamics processor settings apply to Mic, Codec and Phone sources only. If applied to other source types, these settings will be ignored.

- **NOISE GATE STATUS:** Turns the voice processor's noise gate on and off. Choose **Bypass** or **Active.**

- **COMPRESSOR STATUS:** Turns the voice processor's compressor functions on and off. Choose **Bypass** or **Active.**

©2008 Axia Audio — Rev. 3.0

- **DE-ESSER STATUS:** Turns the voice processor's de-esser module on and off. Choose **Bypass** or **Active.**

- **NOISE GATE THRESHOLD:** Determines the point at which the Noise Gate will operate (if set to **Active**). Adjustment range is between 0dB and -50dB.

- **COMPRESSOR THRESHOLD:** Determines the point at which the Compressor will operate (if set to **Active**). Adjustment range is between 0dB and -30dB.
- **DE-ESSER THRESHOLD:** Determines the point at which the De-Esser will operate (if set to **Active**). Adjustment range is between 0dB and -20dB.

- **COMPRESSOR RATIO:** Sets the amount of compression that will be applied to the mic audio. Adjustment range is between 1:1 and 16:1.

- **DE-ESSER RATIO:** Sets how aggressively the de-esser will remove sibilance from the mic audio. Adjustment range is between 1:1 and 8:1.

- **NOISE GATE DEPTH:** Sets the amount of noise reduction that will be applied to the mic audio if the noise gate is set to **Active** and the Noise Gate Threshold is reached. Adjustment range is between 0dB and -30 dB.

- **COMPRESSOR MODE:** Toggles the voice compressor between **Freeze** and **No Freeze** modes.

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.

- **RECORD MODE:** This option allows you to define how this source is sent to the **PGM-4/Record** bus. (**Record** is a special pre-on/off, post-fader bus with a special output for a dedicated recording device.)

  - » **Stereo** sends both sides of the source to both sides of the recording device, as normal.
  - » **If Phone or Codec Left, otherwise Right** sends the source to the recorder's left channel ONLY if the Source Type is defined as **Phone** or **Codec.** Any other Source Type will be fed to the recorder's right channel.
  - » **Summed-Mono to Left** sums the left and right

©2008 Axia Audio — Rev. 3.0

sides of the audio source and sends the summed signal to the recorder's left channel.

» **Summed-Mono to Right:** Same as above, but sends summed audio to the recorder's right channel.

That's the last option. Click on the **Save Changes** button at the bottom of the page.

Congratulations — you've just created your first Source Profile! You can now bring up your new source on any Element fader.

Adding Backfeeds and GPIO to a Source Profile

Since making a mix-minus and mapping contact closures is often done at the same time a new Source Profile is constructed, let's briefly recap those procedures. (Detailed setup and operation instructions for your GPIO Nodes and Audio Nodes are found in their respective User Manuals.)

Let's say you've connected a hybrid to your Analog Node, named it "Hybrid", and assigned it a channel number of 77. You've constructed a Source Profile for it, and you now want to give it a mix-minus backfeed and set up a GPIO contact closure for the "take" and "drop" functions. Here's how:

1. Make sure that, in your hybrid's Source Profile, the **Source Type** option is set to "Phone" and the **Feed to Source Mode** is set to "Auto". Also set the **Backfeed Enabled/Disabled** option to "Enabled".

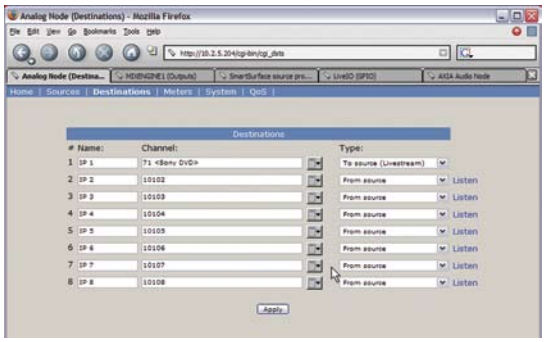


Figure 2-3: Analog Node Destinations screen

2. Use your Web browser to connect with the Audio Node you'll be sending your backfeed to. Choose `Destinations` from the main menu to enter the Destinations screen (Figure 2-3); we'll set up our mix-minus for output #7.
3. Just to the right of each **Channel** name box , you'll see a "list" icon. Click on the list icon for output #7. A list of available sources will pop up (Figure 2-4).

Figure 2-4: Picking a source from the list box

Notice that the names and channel numbers of the sources are both displayed for easy identification. We'll choose **Hybrid** from the list; the list window closes and you're returned to the **Destinations** screen. The line for output #7 now displays the channel number and name of the source.

4. Now comes the step that makes this output a mix-minus. Each **Destination** entry line has a drop box at the end; click on the drop-down and choose **To source**.

Automagic: If the hybrid source is loaded to an Engine and the engine is generating a mix minus source already, the list box will show TWO identical channels for the source — but one will have a description of "TO:" in front of it. Selecting this source automatically selects TO SOURCE in the TYPE box.

For a detailed description of these options and how they operate, please refer to the *8x8 Analog Node / 8x8 AES Node User's Manual* available at **www.AxiaAudio.com/manuals/**.

5. Click **Apply**.
6. Now point your browser back to your Element Control Center page and choose `GPIO Configuration` from the menu sidebar. (You may also use a GPIO Node other than the Power Supply/GPIO that shipped with your Element, just point your browser to that node's IP address instead.)

©2008 Axia Audio — Rev. 3.0

7. The GPIO setup screen uses the same method of assigning devices to ports as the Analog Node. Pick a port to map your hybrid's contact closures to, using the list tool, and click **Apply**.

The process is complete: you've made a Source Profile for your hybrid, created a mix-minus, mapped it to an Analog Node audio output, and associated a GPIO contact closure with the source.

To confirm, choose the **Options** key at the top of any Element fader channel and load your new Phone source. Now press the **Talk** key just below the fader; you'll see the Status Symbol shown in Figure 2-5 appear in the window above the channel to indicate that the operator's mic is talking to this telephone hybrid through it's back-feed channel. (Status Symbols are described in detail at the end of Chapter Four, "Element Operations".)

Figure 2-5: Status Symbol indicates mix-minus/ talkback feed to phone source.

Also, take another look at the GPIO screen on your browser. Note the green indicators that have appeared on the port representing your hybrid; switch the Element channel on and off and you'll observe the active tally change pins (as shown in Figure 2-6).

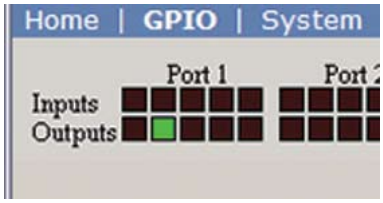


Figure 2-6: GPIO status indicators

What's Next

When you're ready, join us in Chapter Three for more detail about setting up logic commands and GPIO ports for your audio sources.

©2008 Axia Audio — Rev. 3.0

CAT.6, small and still,

yet data and sound pour forth.

Electronic bliss.

©2008 Axia Audio — Rev. 3.0

Chapter Three:

Configuring GPIO

Since the days of rotary-switch source selectors, it's been a challenge to properly route machine logic along with audio. Some complicated schemes have been devised over the years, but broadcasters wanted a simple, fast way to associate logic with audio in a routable environment.

Since the Axia IP-Audio system was designed as a true computer network, we were able to start with a clean sheet of paper to design the first truly routable machine logic/audio interface. Unlike conventional logic connections which require each command circuit to be wired individually, Axia sends machine controls over the same Ethernet your audio travels on, further reducing infrastructure, cost and tedium.

Along with controlling external audio devices, Element's GPIO interface also includes logic commands for routine studio/control room operations such as tally lights, monitor muting, On-Air lights and more.

This chapter explains how to set up Element GPIO to handle these functions. (We assume you've already read the GPIO Node User's Manual, familiarized yourself with its controls, and assigned it an IP. If not, **we strongly recommend you do so**, as this chapter is not designed to be a comprehensive primer.)

> ***Note:*** Your Element ships with a combination GPIO Node/Power Supply that also contains the Element CPU. These units are convection cooled, and we strongly recommend that, when rack-mounting these units, you leave at least 1RU of space **above and below** the GPIO Node/Power Supply to ensure adequate ventilation.

GPIO Port Definitions

Each Axia GPIO has eight DB-15 connectors on its back panel. Each connector can be associated with a device in your studio, and provides five opto-isolated inputs and 5 opto-isolated outputs per device for machine control, lamp drives and remote channel controls.

GPIO ports are programmed to support several different types of devices. How does a GPIO port "know" which type of device is assigned to it? Well, there's a little bit of magic involved.

Back in Chapter Two, when you constructed a Source Profile for a telephone hybrid, you defined the source type (see "Adding Backfeeds and GPIO to a Source Profile," p. 12). This is important, because when that source is assigned to a console fader, Element uses this Source Profile selection to tell the GPIO Node what sort of command to send to the attached device..

If Element "sees" in the Source Profile that the assigned device is a microphone, it tells the GPIO Node to send logic for **On, Off, Remote Mute** and Remote **Talk** commands on the appropriate pins. If it "sees" a line input, it tells the GPIO Node to send **Start, Stop** and **Reset** commands, plus closures for **Ready** lights, etc.

Axia GPIO ports can deliver command sets for the following devices:

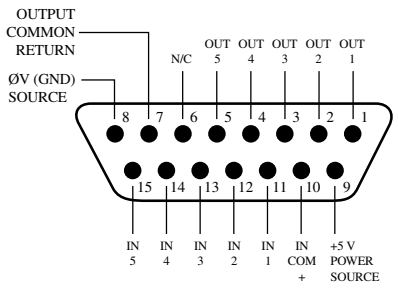
1. Microphone (Operator, Guest or Producer)
2. Line Input
3. Codec
4. Telephone Hybrid
5. Computer Playback Device
6. Control Room Monitor
7. Studio Monitor
8. Profanity Delay Device
9. Accessory Button Panel Device

The next few pages contain tables that explain what function the pins provide in each different device mode.

©2008 Axia Audio — Rev. 3.0

GPIO Operator's Microphone Logic

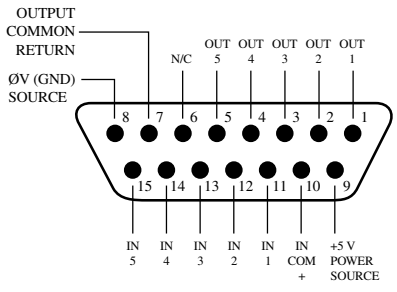
Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turn channel OFF
TALK (to Monitor 2) Command	13	Active Low Input	Activates the Element TALK TO MON2 function and routes mic audio to the Talkback bus.
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (to PREVIEWED SOURCE) Command	15	Active Low Input	Activates the TALK button on every source currently in preview and routes mic audio to the Talkback bus.
OUTPUTS			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON unless TALK or MUTE is active
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
TALK (to Monitor 2) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK TO MON2 is active
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
TALK (to PREVIEWED SOURCE) Lamp	5	Open Collector to Logic Common Return	Illuminates when TALK to PREVIEWED SOURCE is active.
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic +5 Volt Supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Input Common	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



©2008 Axia Audio — Rev. 3.0

GPIO Control Room Guest Microphone Logic

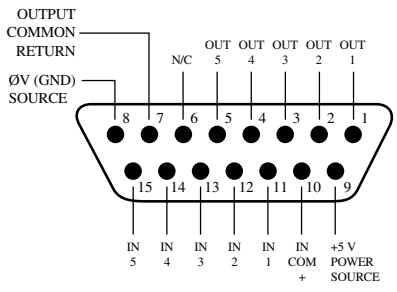
Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turn channel OFF
TALK (to CR) Command	13	Active Low Input	Mutes channel outputs and routes source audio to PVW speakers
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (to SOURCE) Command	15	Active Low Input	Allows an external button to activate channel TALK TO SOURCE function. (Element only; SmartSurface not used)
OUTPUTS			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON unless TALK or MUTE is active
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
TALK (to CR) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK is active
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
TALK (to SOURCE) Lamp	5	Open Collector to Logic Common Return	Illuminates when the channel TALK TO SOURCE function is active. (Element only; SmartSurface not used)
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



©2008 Axia Audio — Rev. 3.0

GPIO Studio (Monitor 2) Guest Microphone Logic

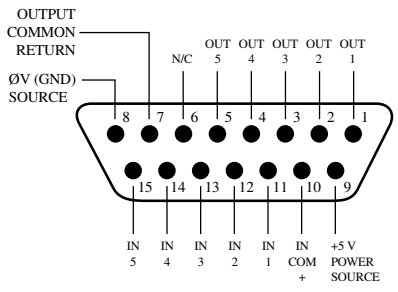
Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turn channel OFF
TALK (to CR) Command	13	Active Low Input	Mutes channel outputs and routes source audio to PVW speakers
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (to SOURCE) Command	15	Active Low Input	Allows an external button to activate channel TALK TO SOURCE function. (Element only; SmartSurface not used)
OUTPUTS			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON unless TALK or MUTE is active
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
TALK (to CR) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK is active
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
TALK (to SOURCE) Lamp	5	Open Collector to Logic Common Return	Illuminates when the channel TALK TO SOURCE function is active. (Element only; SmartSurface not used)
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



©2008 Axia Audio — Rev. 3.0

GPIO Producer's Microphone Logic

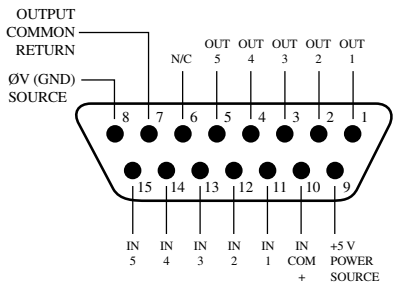
Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turn channel OFF
TALK (to MONITOR 2) Command	13	Active Low Input	Activates the Element TALK to MON2 function and routes mic audio to the Talkback bus.
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (to PREVIEWED SOURCE) Command	15	Active Low Input	Activates the TALK button on every source currently in Preview and routes mic audio to the Talkback bus.
OUTPUTS			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON unless TALK or MUTE is active
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
TALK (to MONITOR 2) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK to MON2 is active.
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
TALK (to PREVIEWED SOURCE) Lamp	5	Open Collector to Logic Common Return	Illuminates when TALK to PREVIEWED SOURCE is active.
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



©2008 Axia Audio — Rev. 3.0

GPIO Line Input Logic

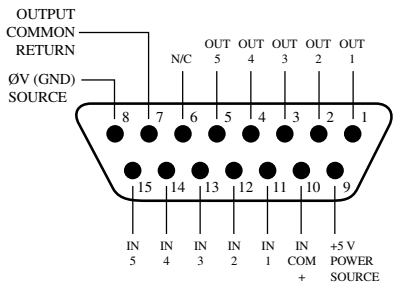
Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turns channel OFF & sends 100 msec STOP pulse
PREVIEW Command	13	Active Low Input	Turns preview ON
RESET Command	14	Active Low Input	Turns channel OFF, while not sending a STOP pulse
READY Command	15	Active Low Input	Illuminates OFF lamp to indicate source's readiness
OUTPUTS			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF and READY is active
PREVIEW Lamp	3	Open Collector to Logic Common Return	Illuminates when PREVIEW is ON
START Pulse	4	Open Collector to Logic Common Return	A 100 msec pulse when the channel status changes from OFF to ON
STOP Pulse	5	Open Collector to Logic Common Return	A 100 msec pulse when the channel status changes from ON to OFF
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



©2008 Axia Audio — Rev. 3.0

GPIO Codec Logic

Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turns channel OFF
TALK (to CR) Command	13	Active Low Input	Mutes channel outputs and routes source audio to PVW speakers
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (to SOURCE) Command	15	Active Low Input	Allows an external button to activate channel TALK TO SOURCE function.
OUTPUTS			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON unless TALK or MUTE are active
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF.
TALK (to CR) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK is active
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
TALK (to SOURCE) Lamp	5	Open Collector to Logic Common Return	Illuminates when the channel TALK TO SOURCE function is active.
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		

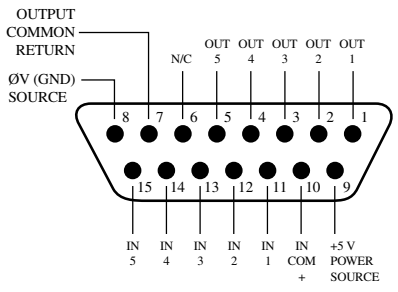


GPIO v."Z" 12/2007

©2008 Axia Audio — Rev. 3.0

GPIO Telephone Hybrid Logic

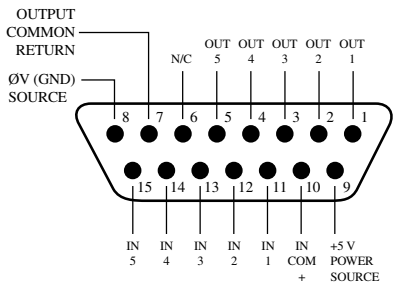
Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turns channel OFF
PREVIEW Command	13	Active Low Input	Turns preview ON
RESET Command	14	Active Low Input	Turns channel off while not sending a STOP pulse
READY Command	15	Active Low Input	Illuminates OFF lamp to indicate source's readiness
OUTPUTS			
ON Lamp	1	Open Collector to Logic Common Return	Illuminates when channel is ON
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
PREVIEW Lamp	3	Open Collector to Logic Common Return	Illuminates when PREVIEW is ON
START Pulse	4	Open Collector to Logic Common Return	A 100 mS PULSE is sent when channel is first turned ON or when PVW is first selected (if "PVW answers" option is selected).
STOP Pulse	5	Open Collector to Logic Common Return	A 100 mS PULSE sent when channel is turned OFF.
POWER & COMMAND			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



©2008 Axia Audio — Rev. 3.0

GPIO Control Room Monitor Logic

Name	Pin	Type	Notes
INPUTS			
MUTE CR Command	11	Active Low Input	Mutes CR monitors and Preview speakers
DIM CR Command	12	Active Low Input	Allows external dimming of CR monitor speakers.
Enable EXT PREVIEW Command	13	Active Low Input	Feeds External Audio Input to PREVIEW
Not Used	14	Active Low Input	
TALK TO EXT Command (Element only).	15	Active Low Input	Turns on Talk to External Audio on Element.
OUTPUTS			
CR ON AIR Lamp	1	Open Collector to Logic Common Return	Illuminates whenever CR monitors are muted
DIM CR Lamp	2	Open Collector to Logic Common Return	Illuminates whenever control room monitors are DIMMED
PREVIEW Lamp	3	Open Collector to Logic Common Return	Illuminates when PREVIEW is active.
Not Used	4	Open Collector to Logic Common Return	
TALK (to CR) Active Lamp	5	Open Collector to Logic Common Return	Active whenever a source has activated its TALK (to CR) function
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		

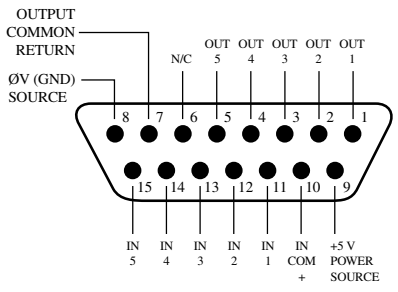


GPIO v."Z" 12/2007

©2008 Axia Audio — Rev. 3.0

GPIO Studio Monitor Logic

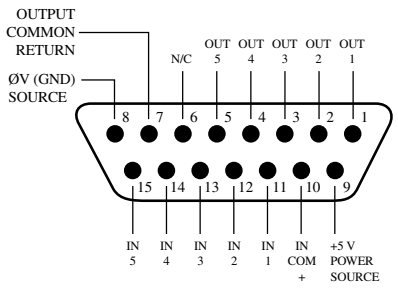
Name	Pin	Type	Notes
INPUTS			
MUTE Studio Command	11	Active Low Input	Allows external muting of Studio monitor speakers.
DIM Studio/Monitor 2 Command	12	Active Low Input	Allows external dimming of Studio monitor speakers.
Remote Event Timer Trigger Command	13	Active Low Input	Resets Countup Event Timer to zero and starts timer.
Remote Countdown Timer Trigger Command	14	Active Low Input	Resets Countdown timer to preset max time and starts timer (Element only).
Not Used	15	Active Low Input	
OUTPUTS			
Studio ON AIR Lamp	1	Open Collector to Logic Common Return	Illuminates whenever Studio monitors are muted
DIM Studio/Monitor 2 Lamp	2	Open Collector to Logic Common Return	Illuminates whenever Studio monitors are DIMMED
Event Timer Trigger Output	3	Open Collector to Logic Common Return	A 100 ms PULSE sent when event timer is started from zero.
Countdown Timer Trigger Output	4	Open Collector to Logic Common Return	A 100 mS PULSE sent when countdown timer is started from preset max (Element only).
TALK TO STUDIO/MONITOR 2 Active Lamp	5	Open Collector to Logic Common Return	Illuminates whenever the TALK TO STUDIO/MON2 function is active.
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



©2008 Axia Audio — Rev. 3.0

GPIO Computer Playback Device Logic

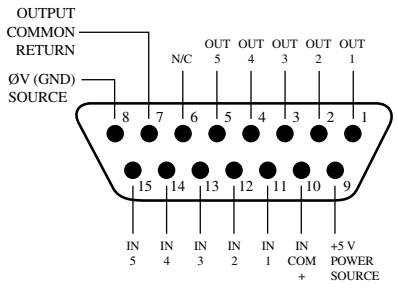
Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turns channel OFF & sends 100 msec STOP pulse
PREVIEW Command	13	Active Low Input	Turns preview ON
Not Used	14	Active Low Input	
READY Command	15	Active Low Input	Illuminates OFF lamp to indicate source's readiness
OUTPUTS			
NEXT Pulse	1	Open Collector to Logic Common Return	A 100 mS PULSE sent when ON button is depressed, except when initially turned ON.
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF and READY is active
PREVIEW Lamp	3	Open Collector to Logic Common Return	Illuminates when PREVIEW is ON
START Pulse	4	Open Collector to Logic Common Return	A 100 mS PULSE sent when channel is first turned ON.
STOP Pulse	5	Open Collector to Logic Common Return	A 100 mS PULSE sent when channel is turned OFF.
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



©2008 Axia Audio — Rev. 3.0

GPIO External Profanity Delay Logic

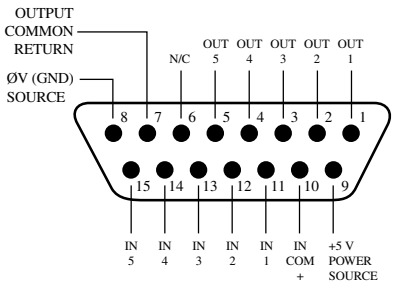
Name	Pin	Type	Notes
INPUTS			
DUMP Lamp	11	Active Low Input	Illuminates DUMP lamp (located on Element Monitor Module).
EXIT Lamp	12	Active Low Input	Illuminates EXIT lamp.
PAUSE Lamp	13	Active Low Input	Illuminates PAUSE lamp.
ENGAGE Lamp	14	Active Low Input	Illuminates ENGAGE lamp.
DELAY ANNUNCIATOR Command	15	Active Low Input	Illuminates the DELAY annunciator on the Element main screen.
OUTPUTS			
DUMP Switch	1	Open Collector to Logic Common Return	Active when DUMP button is depressed.
EXIT Switch	2	Open Collector to Logic Common Return	Active when EXIT button is depressed.
PAUSE Switch	3	Open Collector to Logic Common Return	Active when PAUSE button is depressed.
ENGAGE Switch	4	Open Collector to Logic Common Return	Active when ENGAGE button is depressed.
Not Used	5	Open Collector to Logic Common Return	
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



©2008 Axia Audio — Rev. 3.0

GPIO Film Legendable User Button Module / Accessory Panel / Rack Panel

Name	Pin	Type	Notes
INPUTS			
User Button 1 Lamp	11	Active Low Input	Illuminates User Button 1 Lamp
User Button 2 Lamp	12	Active Low Input	Illuminates User Button 2 Lamp
User Button 3 Lamp	13	Active Low Input	Illuminates User Button 3 Lamp
User Button 4 Lamp	14	Active Low Input	Illuminates User Button 4 Lamp
User Button 5 Lamp	15	Active Low Input	Illuminates User Button 5 Lamp
OUTPUTS			
User Button 1 Switch	1	Open Collector to Logic Common Return	Active when User Button 1 is pushed
User Button 2 Switch	2	Open Collector to Logic Common Return	Active when User Button 2 is pushed
User Button 3 Switch	3	Open Collector to Logic Common Return	Active when User Button 3 is pushed
User Button 4 Switch	4	Open Collector to Logic Common Return	Active when User Button 4 is pushed
User Button5 Switch	5	Open Collector to Logic Common Return	Active when User Button 5 is pushed
POWER & COMMON			
Source Common	7	Logic Common	Connect to ground of source device or to Pin 8
Logic Common	8	Internal 5 Volt return	Can be connected to Pin 7 if source is not providing common
Logic + 5 Volt supply	9	Logic Supply, Individually Fused	Can be connected to Pin 10 if source is not providing voltage; active only when source has been assigned to channel.
Source Supply	10	Common for all 5 inputs	Connect to power supply of source device or to Pin 9
NOT CONNECTED	6		



©2008 Axia Audio — Rev. 3.0

Assigning GPIO to a Source

As you've seen by studying the previous pages, a lot of the work of assigning logic to a source is done for you; once a GPIO port is linked with a Source Profile, all that remains to do is to solder cables connecting the GPIO's DB-15 connectors to the device's control interface.

So, how *do* you link a GPIO port with a Source Profile? It's very easy; let's do it step-by-step.

> ***Note:*** This procedure assumes that you have already defined at least one Source Profile using the instructions outlined in Chapter Two.

1. Open your Web browser and enter the IP address of your Element console. Choose **GPIO Configuration** from the Element Control Center menu. Enter your password if prompted (default login is "user", leave the password field blank).
2. If you haven't previously assigned any GPIO ports, the GPIO definitions screen will be blank. Notice the status indicators at the top of the page, showing the state of the input and output pins of each port. Click on the list icon to the right of an unused line.
3. When you click on any list icon, a small popup window will open, enumerating all of the audio sources available on the Livewire network (Figure 3-3). Choose the source you wish to associate with a GPIO port by clicking on it; the window will close and the source's name and channel number will appear in the **Channel** box.
4. Type a descriptive name in the **Name** field, and click on the **Save** button at the bottom of the page.
5. Look at the page-top status indicator for Port 1. You'll see that one of the pin status indicators is lit in green; this means that the port is now sending a GPIO logic state of "true" through this pin. Assign the source for which you just created a GPIO link to a Element channel; operate the **On** and **Off** keys for the channel and watch the pin status indicators change as you do so.

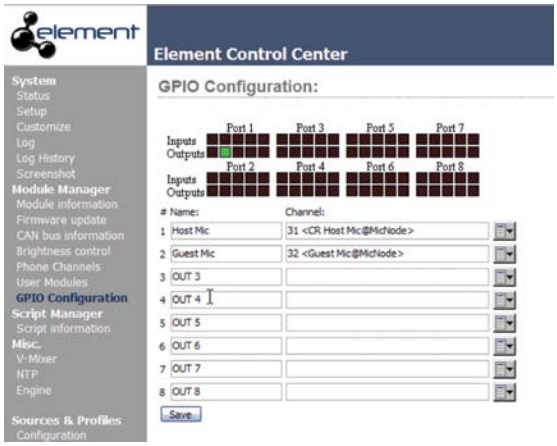


Figure 3-2: GPIO definitions page

Figure 3-3: GPIO Select Source popup list

The source we've been using for this demonstration is a telephone hybrid; we can now observe the pin status indicators change as we turn the channel on and off, as shown in Figure 3-4.

Figure 3-4: Pin status indicators showing GPIO port activity

Referring to the Hybrid Logic Chart on Page 28, we can see that when the Element channel is **Off**, the indicator representing Output Pin 2 – the **Off Lamp** logic command – is high. Turning the channel **On**, we see Pin 4 pulse briefly before Pin 1 goes high: the GPIO has just sent a **Start** pulse, then lit the **On** Lamp. If you turn the channel **Off** again, you'll observe a **Stop** pulse, and the **Off Lamp** command

©2008 Axia Audio — Rev. 3.0

once again become active.

See how easy that was? Simply assigning an existing audio source to a GPIO port automagically configures the port for the type of device supplying the audio, and send the appropriate logic commands to that port when the source is assigned to a Element channel.

A word about the GPIO Ports

Input Connections

Current on these inputs must be limited to 20 mA or less, through the use of a current-limiting resistor. An external power source (24 volts DC maximum) is recommended for all inputs and outputs, to prevent ground loops between equipment. However, if customer equipment is completely isolated, using power from the GPIO Port connectors is acceptable. Figure 3-5 shows details for both types of connections:

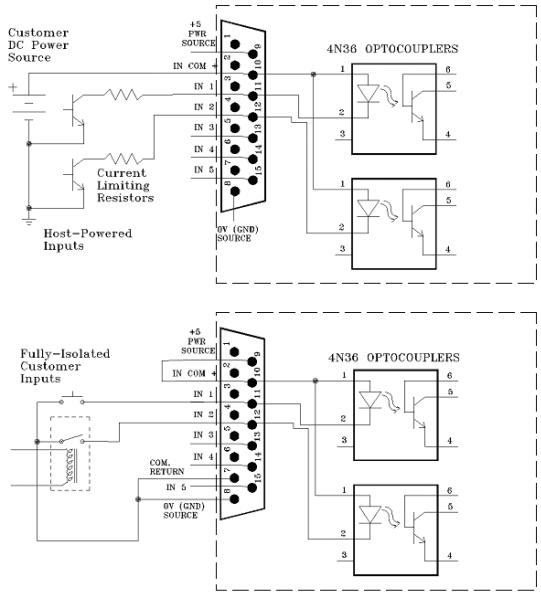


Figure 3-5: GPIO input connections

Output Connections

The GPIO node's outputs are opto-isolated. Current should be limited 100 mA through each output, with the total current draw from the +5 Volt supply not to exceed 3 amps. Figure 3-6 shows the recommended connections for outputs:

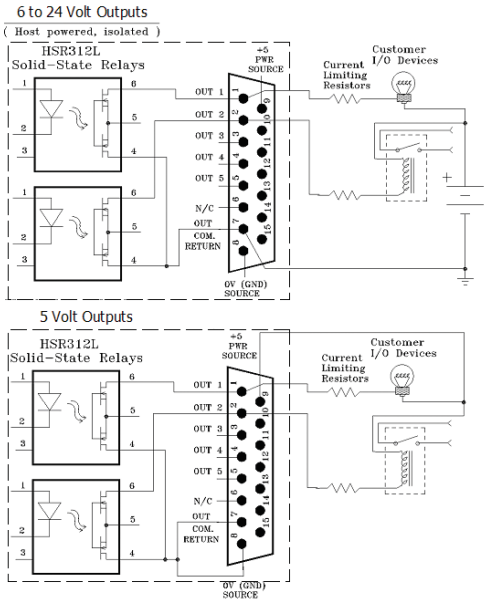


Figure 3-6: GPIO output connections

Note: Some external devices will allow a logic control input "sink to ground" to activate. Thus, some devices may not work with an Axia GPO control, because they may not fully achieve ground through the output transistor. The voltage drop between the collector and emitter may not be low enough to activate the device, so an external relay controlled by the GPO may need to be used to provide a "dry" contact closure to the external device.

Please note that this section is provided only as a "jump-start" introduction to Axia GPIO nodes. For a full understanding of the GPIO node's options and requirements, please read the GPIO Node User's Manual included with your GPIO node, or available for download at www.AxiaAudio.com/downloads/ .

What's Next

When you're ready, join us in Chapter Four for a comprehensive walk-through of Element operations.

©2008 Axia Audio — Rev. 3.0

The power is off

yet your broadcasts continue.

The UPS works!

©2008 Axia Audio — Rev. 3.0

Chapter Four:

Element Operations

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, Channels and Faders

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" would normally be used when referring to Element's linear pots and associated on/off controls, alpha displays, bus assignment keys, etc. But "Channel" is also used when discussing numbered Livewire network streams. To avoid confusion, we'll call Element channel strips "Faders".

©2008 Axia Audio — Rev. 3.0

Sources are assigned to faders 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 fader, source logic follows.

Fader settings are instantly reconfigured whenever a source is selected, to accommodate the unique requirements of the source. For example, a fader controlling a microphone source also controls appropriate monitor mutes. A fader 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 fader.

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 fader 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.

Monitor Module and SoftKnobs

The Monitor Module and SoftKnobs contain Element's operating controls. This is where you'll find volume and selection controls for monitors and heaphones, as well as keys that provide fast access to show profiles, EQ & dynamics, Aux feeds and more. There are two different type of Monitor Modules: one that contains only monitor and option controls, and one that subtracts some monitor controls but adds two faders. We'll outline how each type works.

Software

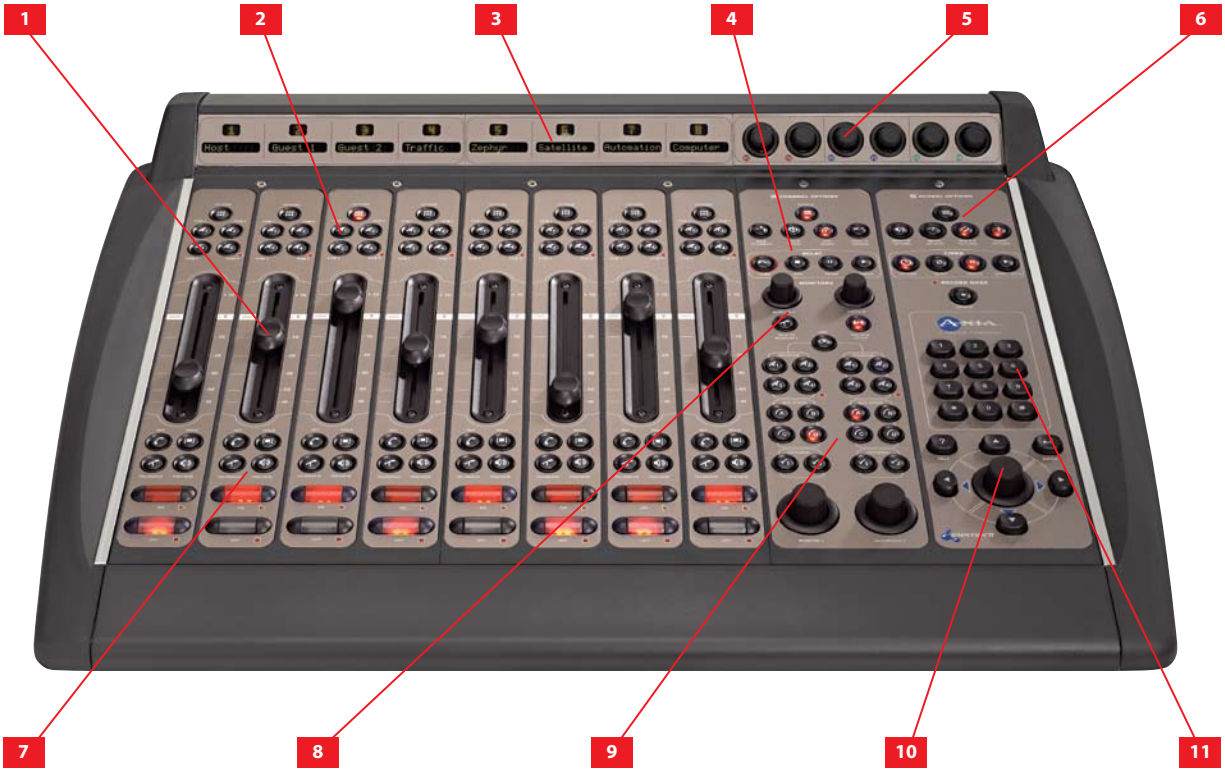
The Element operating system uses a high-performance embedded Linux kernel to help ensure bulletproof, 24/7 operation. Software updates can be obtained from Axia via the Internet and applied by the station engineer. Axia Support recommends performing backups or making printouts of Show Profiles and other configuration data before applying new software updates.

Control Callout

On the following pages are a "bird's-eye view" of Element, with major controls located and their functions explained.

©2008 Axia Audio — Rev. 3.0

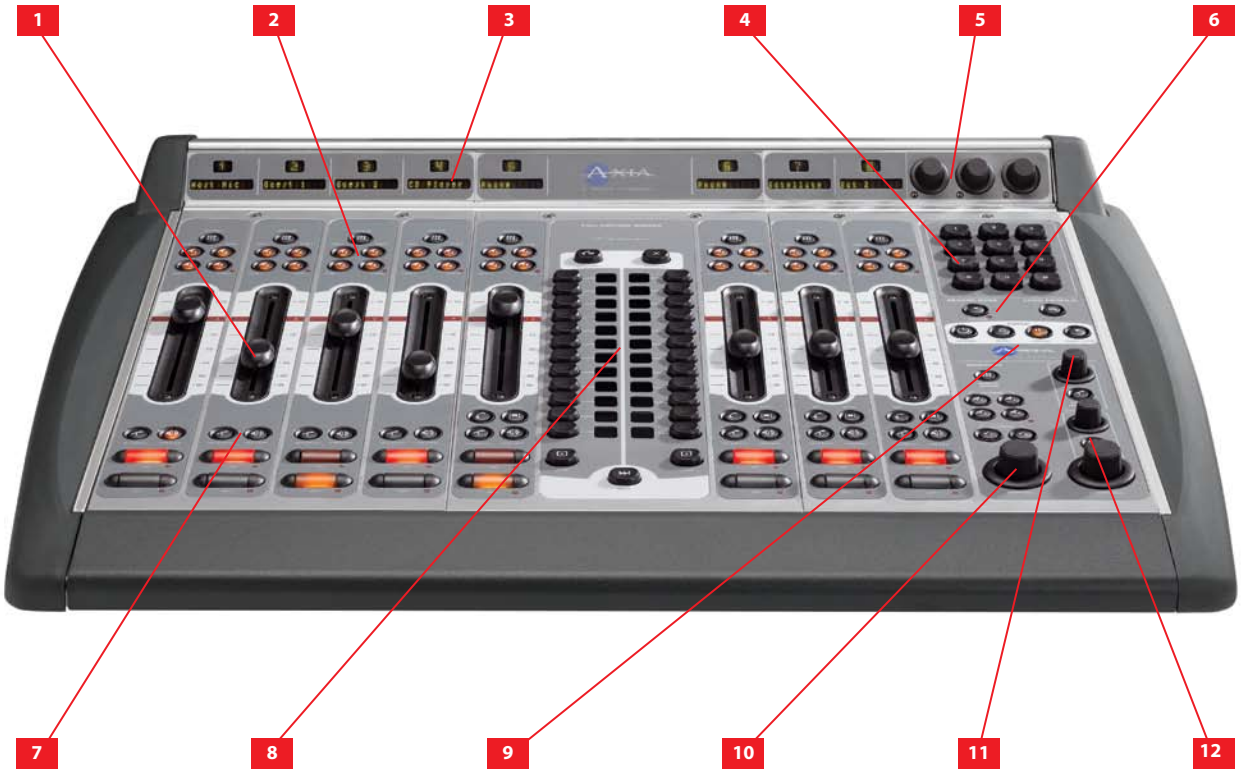
Element with Standard Monitor Module



1. **Faders** can accommodate any input source. Each channel's **Options** button (at the top of the fader strip) lets operators change sources and call 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 fader strip.
3. **Channel Displays** contain a 10-character text readout and a window for fader position numbers and Status Symbol icons. 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. **Monitor Module 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 profanity-delay units.
5. **Six 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. **Monitor 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. **Monitor 2 Talkback & Volume Controls** let operators easily talk to hosts and guests in a second studio, and control their monitor volume.
9. **Monitor Assignment Section** contains volume and selection controls for monitors, headphones, Aux Send buses and Preview.
10. **Control Knob/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.

©2008 Axia Audio — Rev. 3.0

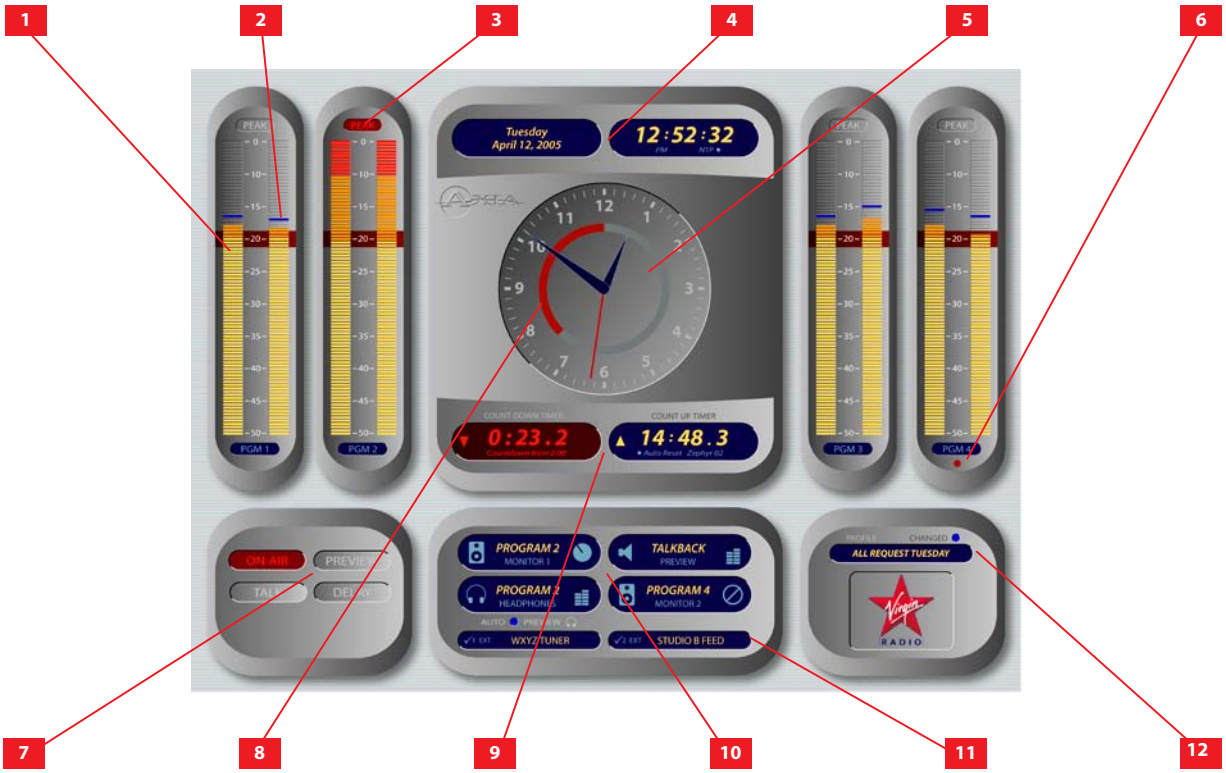
Element with 2-Fader Phone Module and 2-Fader + Monitor Module



1. **Faders** can accommodate any input source. Each channel's **Options** button (at the top of the fader strip) lets operators change sources and call 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 fader strip.
3. **Channel Displays** contain a 10-character text readout and a window for fader position numbers and Status Symbol icons. 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. **Numeric Keypad** lets operators dial phone systems or codecs directly. Also provides numeric input during surface setup.
5. **Three 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. **Record Mode and Load Profile Keys** let operators recall customized Show Profiles and activate Record Mode to quickly capture off-air bits to an external recording device.
7. **Talkback & Preview Keys** let operators preview (cue) selected sources off-air, or quickly talk to any source(s) with associated backfeed(s).
8. **2-Fader Phone Module** provides direct control of all hybrid and line control functions for Telos Nx12, TWOx12 or Series 2101 Broadcast Talkshow Systems.
9. **Timer Controls** help keep track of on-screen elapsed and remaining time displays.
10. **Monitor 1 Section** contains volume and selection controls for monitors, headphones, Aux Send buses and Preview.
11. **Monitor 2 Talkback & Volume Controls** let operators easily talk to hosts and guests in a second studio, and control their monitor volume.
12. **Headphone and Preview Volume Controls** let operators adjust Host headphone volume and control Preview (cue) volume and options.

©2008 Axia Audio — Rev. 3.0

Element Stereo Main Screen Overview



1. **Program Meters** for each of Element's four stereo Program buses feature true VU meter ballistics with an expanded scale. Solid bargraphcs indicate average level; the red line at -20dBFS corresponds to the 0dB mark on a traditional analog meter.
2. **Peak Indicator** indicates absolute peaks to show talent where the loudest portions of program material are landing on the VU scale. (This indicator may be switched off if desired.)
3. **Peak Indicator Warning** illuminates when peaks reach or exceen 0dBFS.
4. **Day/Date and Digital Clock** keep operators informed of what day it is (in case they forget, which is entirely normal). Clock can be set for 12- or 24-hour operation, and can be slaved to a networked NTP server if desired.
5. **Analog Clock** presents a traditional clockface with second hand for fast orientation.
6. **Utility Meter** normally shows the volume of sources assigned to the **PGM-4/Phone/Record** multifunction bus, but can optionally be slaved to meter whatever source is auditioned in the Control Room Monitors.
7. **Annunciator Panel** informs operators when Mic channels are open, when audio is being auditioned in the Preview (cue) channel, when Talkback is active in either direction, and when external profanity delay units (if any) are active.
8. **Countdown Ring** appears when Element's Countdown Timer is active, and decreases during the last :60 seconds as a visual prompt.
9. **Count Up and Countdown Timers** provide an automatic event timer with manual elapsed-time function, and a countdown timer to assist in back-timing live program material.
10. **Monitor Selection Indicators** show talent what source is being presented in the Control Room and Studio Monitor channels, the CR Headphone channel and the Preview channel, and shows the position of each volume control when adjusted.
11. **External Monitor Indicators** show what sources are assigned to each of the two programmable External Monitor channels.
12. **Show Profile Indicator** tells talent what Show Profile is currently loaded, and whether the console has been changed from Profile settings.

©2008 Axia Audio — Rev. 3.0

Control Details

General Fader Controls

Although any Element faders can handle any type of audio source, 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.

- **Fader Assignment Display**

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

If a channel is ON and the operator changes its source, the display will flash between the source currently on-air and the newly selected source. The fader will not "take" the new source until the operator has turned the channel off, preventing on-air errors. If a Show Profile is loaded that specifies an unavailable source, the display reads `Requested source not found`.

Just above the alpha display is a Status Symbol display. This display indicates the fader's channel number or presents a Status Symbol to indicate when backfeed or talkback is active.

- **Channel Options Key**

Pressing this key activates the **Channel Options** keys located on Element's Master Module. Refer to the sections on Monitor Module controls later in this chapter for detailed functions.

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

Each channel is assigned to any of the three main output buses by selecting any (or all) of these program keys. Generally, PGM-1 is the main air bus and other buses are used for production or other programming requirements. All program outputs are post-fader and post on-off function.

Figure 4-1: Line Module Fader Channel

- **PGM-4 / Record Bus / Phone Bus Key**

The fourth bus functions as a regular output bus, but any sources feeding PGM-4 are also sent to a special **Record** bus that feeds a dedicated recording device, as well as to the **Phone** bus that sends mix-minus audio to any phone callers. These buses operate independently but simultaneously, allowing jocks plenty of operational flexibility.

PGM-4 bus audio output is post-fader, post-on/off; **Record** bus audio output is post-fader, pre-on/off, and **Phone** bus audio is sent to callers pre-fader, pre-on/off.

- **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 is has an associated backfeed, pressing this key will allow the board operator to communicate with the person at the other end of the backfeed; the fader's Status Symbol display will change to show that Talkback is active.

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 fader 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. *Momentarily* pressing any channel's **Preview** key as-

©2008 Axia Audio — Rev. 3.0

signs 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 interlock option for Preview can be disabled in any Show Profile's Monitor section so that users can gang preview sources without the need to press and hold the preview button.

The audio from the Preview bus feeds any speakers dedicated to Preview, and can also be routed to the operator's headphones if desired. Preview speakers will be muted whenever the board operator's microphone is **ON**, or is assigned to the Preview or Record buses.

New Feature: If an operator presses Talk on a fader to which a Phone source is assigned and, while holding it, presses the Preview button, both will latch on, makeing it easy to have a hands-free off-line chat using the CR Mic. Pushing either one clears both.

- **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. In **fader-start** mode, the **On** and **Off** keys act more as indicators of channel status; the **On** key will not turn the channel on if the fader is at the bottom of its travel.
2. In **fader-normal** mode, on/off status and start/stop commands follow the channel ON/OFF buttons independent of fader position.

Figure 4-2: Phone Module Fader Channel

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

- **On and Off Keys**

When the channel is in **fader-normal** mode the **On** key (▶) turns the channel on and the **Off** key (■) turns it off. The channel **On** and **Off** functions also provide logic (start and stop pulses, monitor muting, etc.) appropriate to the selected source.

If a control-room mic is assigned to a fader, the CR monitors and Preview speakers will mute when that fader is turned on. and when sources are Previewed. If a fader is hosting a microphone source, pressing the **On** button when the fader is already on mutes the channel, providing a "cough" function, until the button is released.

New Feature: In fader-start mode, holding the **Off** button while moving the fader up disables the usual GPIO start command. The **On** button may be later used to issue the command.

4-Phone Module Channel Controls

You can assign any type of source to any Element fader module. But 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 (described in the previous section). But whan a **Telos**

©2008 Axia Audio — Rev. 3.0

telephone 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 indicates an available line by displaying a single dot. A call ringing in shows a "ringing" icon. It is answered by pushing either the **On** or **Preview** button. If **Preview** is already on, toggling to **Off** and then **On** answers the call. When the call is answered, an "up arrow" Status Symbol icon displays. Also, a window on Element's screen will show call status and incoming caller ID (if available).

When the phone channel is off and in Preview, press **Off** to drop the call. When the channel is on, press **Off** *once* to put the caller on hold; press *twice* to drop.

Making Calls

To dial out from the console, press any control key (**Off**, **On**, **Hold**, **Set**, or **Preview**) on a fader that hosts a Phone source. This assigns the Keypad on the Monitor Module to that fader. Dialing works like a mobile phone: enter all digits, then press the **Enter** (↵) or **Pound** (#) key to place the call. The Status Symbol icon above the fader indicates first that a line is active and then that the call is proceeding. The Element display screen will also indicate the numbers dialed and call status.

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

- **Set Key**

If you're using certain Telos broadcast phone systems, pressing the **Set** key transfers the phone line assigned to that fader to a handset. These buttons are mutually exclusive across all faders; touching a **Set** key on any Phone Fader switches the handset to the line assigned to that fader and cancels any previous line selection. For details on Telos phone systems that utilize the **Set** key, please refer to Appendix D, "Working With Phone Hybrids".

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-Fader Phone Module Controls

This module (Figure 4-3) works with Telos TWOx12, Series 2101 and Nx12 multiline talk show systems. Its features the hybrid controls found on Telos talk show controllers, flanked by two faders. Typically, the left fader is linked to the left hybrid control, and the right fader to the right hybrid control. For details on hybrid control operation and description of Status Symbols used, please refer to your Telos TWOx12, Series 2101 or Nx12 Operator's Manual.

©2008 Axia Audio — Rev. 3.0

Production Module Controls

The Element Production Module gives production-room gurus quick access to often-used EQ and positioning controls for each loaded source. Its controls provide direct access to the options presented in the **EQ & DYNAMICS** options screens detailed later in this chapter.

> ***About Element EQ:*** Our EQ model is three-band quasi-parametric. Controls are provided for center frequency and boost/cut, with a unique SmartQ™ automatic bandwidth system.
>
> SmartQ works by varying the Q of the selected parametric band proportional to the amount of boost or cut you specify. A small amount of boost or cut will affect a broader range of frequencies for a warm, musically-pleasing effect. As gain is increased the Q sharpens, affecting a narrower range of frequencies for tighter control of target bands. At aggressive cut levels, the EQ becomes a tunable notch filter. SmartQ keeps the EQ sounding natural at virtually all settings.

To activate the Production Module, press the **Options** key on any fader strip. The Production Module's overbridge display will immediately mirror the number of that fader and the name of the source you're adjusting EQ for.

Once the Production Module is active, you can quickly adjust EQ using the controls described below. To quickly set EQ for multiple channels, rotate the **SELECT** knob on the overbridge panel to choose a different fader source, and press it to **TAKE** that source.

Figure 4-4:Production Module + overbridge display

- **AUX SEND A/B/C/D:** These controls allow you to send the output of the active fader to any or all of the four Aux Send buses for ourboard audio effects processing, custom IFB mixes, etc. Each Aux Send control has 50 dB of adjustment range, from -∞ to +10dB. Rotating the knobs adjusts the send level; pushing the knob toggles the send between **PRE-** and **POST-FADER.**

- The three central pairs of knobs are **EQ Controls: HI GAIN/HI FREQ**, **MID GAIN/MID FREQ** and **LO GAIN/LO FREQ**.

The **GAIN** knobs provide 40 dB of adjustment range for boost or cut of the three parametric bands, from -25 dB to +15 dB.

The **FREQ** knobs each act slightly differently:

» **HIGH FREQ:** Sets the active frequency for high-band EQ. Pushing this knob toggles the EQ model between **Shelf** and **Bandpass** modes. If set to **Shelf**, the **HIGH FREQ** knob determines the top of the shelf. If set to **Bandpass**, it sets the center frequency of the BP filter (SmartQ determines the width of the BP filter, as described above).

» **MID FREQ:** Sets the center frequency of the midrange parametric band.

» **LOW FREQ:** Sets the center frequency of the low parametric band. Pushing this knob toggles the EQ model between **Shelf** and **Bandpass** modes. If set to **Shelf**, the **LOW FREQ** knob determines the top of the shelf. If set to **Bandpass**, it sets the center frequency of the BP filter

- The four knobs at the bottom of the module are the **PAN** controls.

» The upper left knob **PANs** the sound field left-to-right.

» The **FRONT-BACK** knob moves the soundfield front-to-rear (in 5.1 mode).

» The **SURROUND** knob pans the rear speakers left-to-right (in 5.1 mode).

» The **LFE** knob adjusts the gain of the Low Frequency Effects channel (in 5.1 mode).

©2008 Axia Audio — Rev. 3.0

Standard Monitor Module Controls

The standard Monitor Module (Figure 4-5) contains direct-access controls for all of Element's options and features. 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 & Control Knob

Located at the bottom right of the module is a **Control Knob** 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 **Control Knob** 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. The operator first presses the **Options** key atop the fader s/he wishes to modify; the Channel Options box is displayed on Element's screen (Figure 4-6), and the Channel Options keys then become active.

Figure 4-5:Standard Monitor Module + SoftKnobs

Channel Jumping: If any fader is in **Options** mode, you can "quick jump" to set options for other fader channels by pressing the Options key at the top of their channel strips.

For example, if you need to change the **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 onscreen. 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, and 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 4-6) 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. They can also preview a source before selecting it.

©2008 Axia Audio — Rev. 3.0

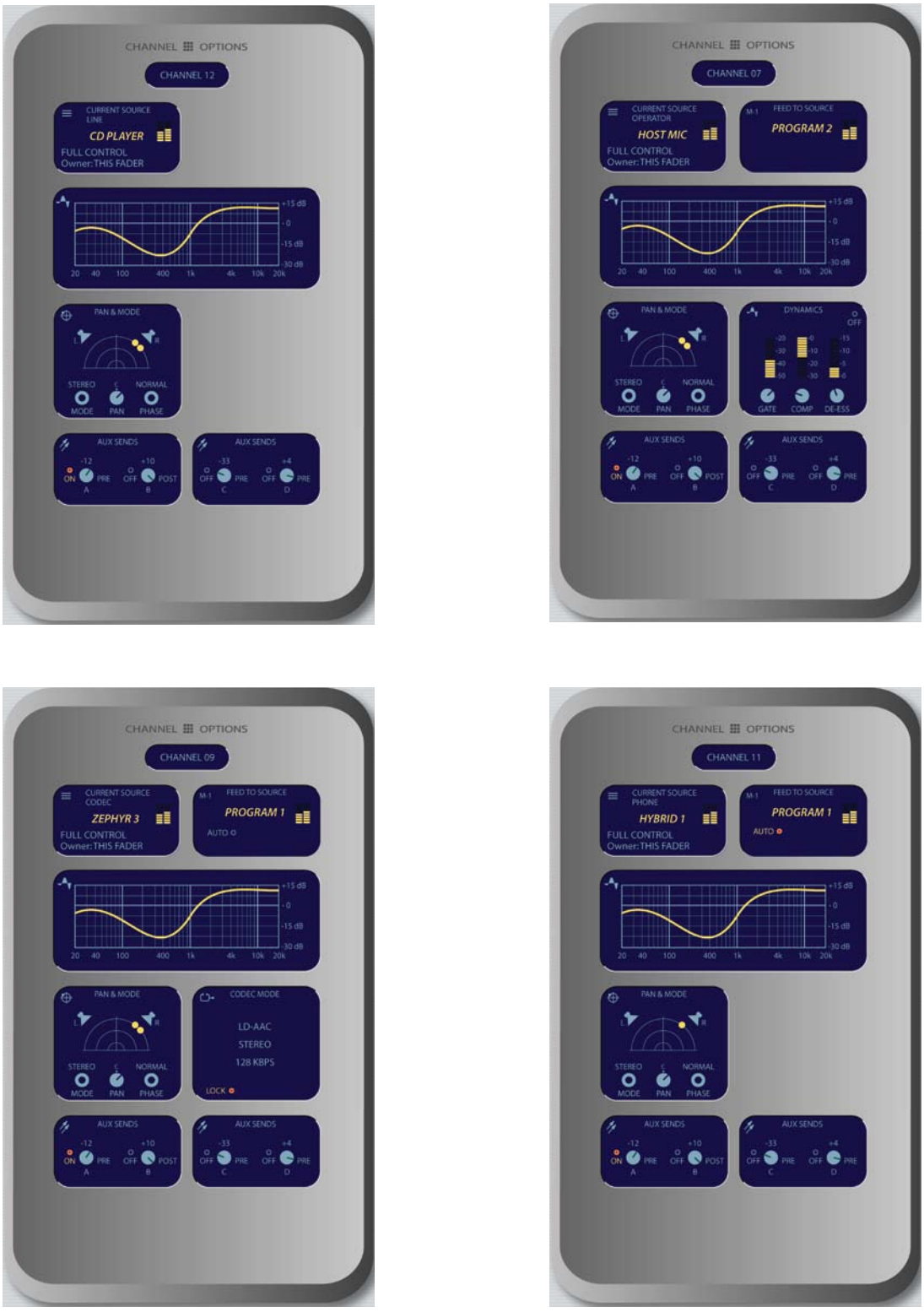


Figure 4-6: Channel Options screens. Information displayed differs based on the type of source currently assigned to the active fader. Shown clockwise from top left: Line Source Channel Options screen, Mic Source Channel Options screen, Phone Source Channel Options screen, Codec Source Channel Options screen.

©2008 Axia Audio — Rev. 3.0

A Note About Source Sharing: Element's source sharing protocol keeps sources from being controlled by more than one console at a time.

If an operator tries to assign a source that's already in use on another console and has a backfeed or logic control enabled, the ID box in the Channel Options screen informs him that he has "Listen Only" rights to the source (as seen in the righthand box above) - no backfeed will be sent from his console, and no machine logic is tied to the fader's **ON/OFF** keys.

If the source is not in use anywhere else, the operator is informed that they have "Full Control" of the source (as shown in the lefthand box above), and backfeed and logic operate normally.

Figure 4-7: Source menu screen.

» **Source/Take**: rotating this SoftKnob scrolls through the available sources. Push to take.

» **Previous**: Whenever you "take" a new source, your last source selection is held in memory. Should you change your mind about the new source and wish to return to your previous source selection, press the **Previous** knob. the prior source will be immediately reloaded to the fader.

» **Preview** allows you to hear the highlighted source through the Preview speakers without haveing to "take" it. Press and hold to listen; release to end the preview.

» Pressing **Source** again returns to the Channel Options screen.

- **EQ & Dynamics Key**

This command allows three-band parametric equalization to be applied to the selected audio source and, if 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 amount of cut/boost applied, and the center frequency of each band.

» **Lo Gain/Bypass/Active:** Rotating this Soft-

Figure 4-8: EQ menu screen.

©2008 Axia Audio — Rev. 3.0

Knob adjusts the amount of gain or cut applied to the low EQ band; note that the Shelf/Peak indicator next to the onscreen control automatically shifts to Peak mode when boost is applied, and shifts to Shelf mode when cut is applied. Pressing this knob toggles EQ between **Active** and **Bypass.**

» Rotating the **Lo Freq** knob adjusts the center frequency of the low EQ band (20 - 320 Hz).

» Rotating the **Mid Gain** and **Mid Freq** SoftKnobs adjust gain and center frequencies of the mid-range EQ band (125 Hz - 2 kHz).

» The **Hi Gain** knob adjusts the boost/cut of the high frequency band (1.25 kHz - 20 kHz).

» **Hi Freq/Shelf/Peak:** 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 **Peak** 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 44 of this manual.

Dynamics controls are available if the selected source is a Mic, Codec or Phone input. Pressing the **Dynamics & EQ** key twice invokes the **Dynamics** menu. 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

Figure 4-9: Dynamics menu screen.

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

©2008 Axia Audio — Rev. 3.0

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. When the Freeze function is active, a red "lamp" illuminates just above the Compression Ratio bargraph meter.

» **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.

Figure 4-10: Pan / Mode menu screen.

- **Pan & Mode Key**

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

It's a Lock: If a "padlock" icon appears in any options screen, that function is "locked out" in the Source Profile and adjustments cannot be made. In Figure 4-10, the padlock appears just below the soundfield representation, above the Mode indicator. The onscreen Mode SoftKnob is also "grayed out" to show that this function is unavailable.

Operators cannot "unlock" locked functions; if you wish to make locked functions available, simply modify the source profile to allow them and when the source is reloaded to the console, the padlock will have disappeared.

» **Pan/Balance:** Rotating this SoftKnob adjusts the the Pan setting for mono sources and the Balance setting for stereo sources.

» **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

©2008 Axia Audio — Rev. 3.0

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). Push to **Take** the new setting.

» **Phase**: If the selected source is *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 Both** reverses phase of both stereo channels. If the selected source is in *mono*, the choices are **Normal** and **Invert** only. Press to **Take** the new setting.

» 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.

» **Send A/B/C/D:** Rotate these knobs to adjust the level of the audio you'll be sending to any or all of the four Aux Send buses.

» **Pre/Post Fader:** Pushing any of the four **Send** SoftKnobs toggles the activity state of that particular **Send.** The state is indicated in the display above.

* **On Pre** sends the source assigned to the fader to the selected Aux Send bus *prior* to having its gain adjusted by the fader (fixed level).
* **On Post** sends the source assigned to the fader to the selected Aux Send bus *after* having its gain adjusted by the fader (variable level).
* **Off** sends no audio to the Aux Send bus.

» Pressing **Aux Sends** again returns to the Channel Options screen.

• **Feed To Source Key** M-1

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 audio supplied through the backfeed. This selection is normally pre-set when a Source Profile is constructed (see Chapter Two, "Configuring Inputs," for information on Source Profiles), but can be modified on-the-fly.

Figure 4-11: Aux Sends menu screen.

Figure 4-12: Feed to Source menu screen.

©2008 Axia Audio — Rev. 3.0

» The **Select/Take** knob, when rotated, scrolls through the list of available audio sources that can supply backfeed or mix-minus.

* **Automatic** backfeeds audio from the Phone bus when the channel is **Off**, and audio from the Program-1 bus when the channel is **On**, with mix-minus if needed.
* **Phone** backfeeds audio from the Phone bus, 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.

About the PHONE and PROGRAM 4 buses: Element's Program 4 mix bus is a special multi-purpose bus that does several things at once to make life easier for the operator.

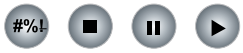
Any source assigned to Program 4 also be fed to the PHONE bus and sent to callers when the channel is **OFF** and **AUTO** is selected as the **Feed To Source Mode**. PHONE is an internal bus which does not feed a separate output; instead, it picks up the Program 4 audio pre-fader and pre-On/Off and backfeeds them to an PHONE sources.

Sources assigned to Program 4 are also sent to the RECORD bus, a special fixed-level output that can be used to feed recording devices.

» 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 used to provide from-the-board control of an external profanity delay unit using one of Element's GPIO ports (refer to Chapter 3: Configuring GPIO for instructions on GPIO setup).

The keys provide for remote activation of **Dump, Exit, Pause** and **Engage** functions, respectively.

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.

Programming the External Monitor Selector keys: 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.

You can also pre-program the source assigned to these keys when you construct or capture Show Profiles, using the options available on the **Monitor** page of the Show Profile setup menu.

- **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

©2008 Axia Audio — Rev. 3.0

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 automagically 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 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.

There are three different Preview to HP modes; the current mode is indicated by an annunciator on the Element display screen just below the headphone selection tally.

» **Off:** When Preview to HP is off, the selector key is unlit. No preview audio will be heard in the board-op's HP path.

» **On:** Pressing the **Preview to HP** key once from the Off state sends preview audio, when present, into both sides of the operator's headphone feed, overriding the Monitor audio. The **Preview to HP** key is now lit.

» **Auto:** Pressing the **Preview to HP** key a second time toggles Auto mode. In this mode, 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.

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.

LOADING A NEW SOURCE WHILE ANOTHER SOURCE IS STILL ACTIVE: Element will never interrupt an active source in order to load a new one.

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

©2008 Axia Audio — Rev. 3.0

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 previously loaded Profile until you exit the Show Profile screen. The **Previous** command functions as a "speed key," highlighting the previously assigned source in the source list and allowing the operator to return to it should they change their mind. Press to "take."

DON'T PANIC! 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.

Pressing and holding the **Profile** key for five seconds instantly clears changes made to the current Show Profile, resetting 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 4-14.

» Pushing the **TALK LEV** SoftKnob selects between adjusting the levels of the **Monitor 1 Talk To** and **Dim** levels and the **Monitor 2 Dim** level. The active function is highlighted in yellow on the screen. Turn the knob to adjust the values of

Figure 4-13: Show Profile selection screen.

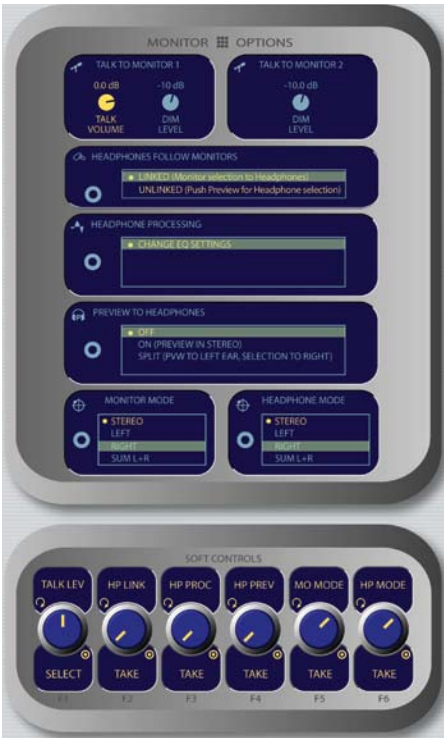


Figure 4-14: Monitor Options menu screen.

©2008 Axia Audio — Rev. 3.0

these settings.

» Rotate the **HP LINK** knob to choose whether the headphone source will follow the Monitor 1 source selection. Push the knob to **Take** your selection.

  * **Linked** means that Headphone and Monitor 1 selections will mirror each other; making a selecton from one bank of keys on the Monitor Module will automatically choose the same source on the other bank.
  * **Unlinked** allows Monitor 1 and Headphone sources to be selected independently of each other.

» The **H/P PROC** knob allows the board op to turn Headphone Processing on and off, define how it operates if active, and change Headphone Processing settings. Pushing the **HP Proc** knob selects **Change Settings** onscreen and takes the operator to the **H/P Process** screen shown in Figure 4-15.

In this screen, the operator is presented with a graphical representation of the EQ curve applied to the headphone channel. Talent can then 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.

Figure 4-15: H/P Process settings.

» **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.** Note that the EQ mode for this band automatically changes to Peak mode when boost is applied, and Shelf mode when cut is applied.

  * Rotating the **Lo Freq** knob adjusts the center frequency of the low EQ band (20 - 320 Hz).
  * Rotating the **Mid Gain** and **Mid Freq** Soft-Knobs adjust gain and center frequencies of the midrange EQ band (125 Hz - 2 kHz).
  * The **Hi Gain** knob adjusts the boost/cut of the high frequency band (1.25 kHz - 20 kHz).
  * **Hi Freq/Shelf/Peak:** Rotating this Soft-Knob adjusts the center frequency of the high EQ band (1 kHz - 16 kHz). Pushing this knob toggles the high-band EQ method between **Peak** and **Shelf** modes.
  * Press the **Monitor Options** key to return to the previous 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 44 of this manual.

» The **H/P PREV** knob controls whether audio from the **Preview** bus can be fed to the board op's headphones, and if so, in what manner. Note that operators can also press the **Preview to HP** key on Element's Monitor Module to toggle through these settings. Rotating this knob selects from the following options:

  * **Off:** The operator will not hear Preview audio in the headphone channel.
  * **On:** audio from the Preview channel over-

©2008 Axia Audio — Rev. 3.0

rides the **Headphone Monitor** selection and is sent to both the left and right channels of the operator's headphones.

* **Split:** 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.

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

» **MO MODE** knob: Rotate to select from the following options.

* **Stereo:** The selected source will be presented in stereo on the Monitor speakers.
* **Left:** The left channel of the selected source will be presented in both Monitor speakers.
* **Right:** The right channel of the selected 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.

» **H/P MODE** knob: Works like the **MO MODE** knob described above, but affects the board operator's **Headphone** channel. Pressing **Monitor Options** or **Options** exits this menu.

- **Meter Options Key**

Pressing this key opens the Meters screen, where you can choose whether the far-right on-screen meter bars will display levels for the **PGM-4** bus or for the source feeding the **Monitor 1.**

» **Dedicated** means the meter will be slaved to the **PGM-4** bus.

» **Follow Monitors** means the meter will show levels for any source assigned to **Monitor 1.**

Figure 4-16: Aux Sends menu screen.

- **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 4-16); pressing it twice displays the **Aux Returns Master** screen (Figure 4-17), and pressing a third time exits the menu.

Aux Sends Master screen options:

» The **Send A/B/C/D** SoftKnobs, when rotated, set the Master Send Gain for each **Aux Send** bus. Pushing any of these knobs toggles that individual **Aux Send Master On** or **Off.**

» The **Select/Preview** knob allows you to hear any of the four **Aux Send** mixes in the **Preview** channel. Push to preview; releasing to end the preview.

©2008 Axia Audio — Rev. 3.0

Figure 4-17: Aux Returns menu screen.

Figure 4-18: Fader Options menu screen.

Aux Returns Master screen options:

» 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 assginment.

» 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 assginment.

» 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. Push the knob to preview; release it to end the preview.

Press the **Sends & Returns** key one more time to exit the menu.

- **Fader Options Key**

Element supports fader-normal and European-style fader-start channel activation methods. Fader style can be set either globally, or on a per-fader basis.

Each input's Source Profile contains a Fader Mode choice, which lets you set the behavior of *individual faders*. The Global Fader Options menu allows you to set the behavior of the *entire console*.

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 ∞.

Rotate the **St. Mode/Take** SoftKnob to choose fader modes:

» **Fader Normal:** Selects the US convention.

» **Fader Start:** Selects the Euro convention.

» **Retain Source Setting:** Each individual fader uses the settings specified in its Source Profile.

Press the **Fader Options** key again to exit.

©2008 Axia Audio — Rev. 3.0

Figure 4-19: Timer menu.

Note: Pressing and holding the **Fader Options** key for five seconds enters the **IP Address Book** discussed in Chapter 1.

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 4-19). This menu allows users to adjust how the onscreen timers behave. Note that the SoftKnobs are divided into two sections: *Countdown Timer* and *Count Up Timer.*

» The *Countdown timer* is useful for counting down to network joins, top-of-hour IDs, and other events which require exact timing. The Countdown timer is started, stopped and reset manually using the controls displayed when the Timer Options key is pressed. To use:

* Rotate the SoftKnob marked **Begin Time/Reset** to set the interval from which the countdown will begin. As you rotate, small numbers below the red counter onscreen will inform you of the new countdown interval.
* Press the **Begin Time/Reset** knob to set the new time you've dialed in.
* Press the **Start/Stop** knob to begin or end the countdown.
* Pressing **Freeze** holds the on-screen Timer display so you may take note of it; the timer continues to count while the display is frozen, and a second press of the **Freeze** again displays the running elapsed time counter.

"Look at me!!" During the last 60 seconds of the countdown, the inner ring of Element's onscreen analog clock will turn red and diminish as time passes, providing the operator with a visual cue to how much time is left until zero.

- The *Count Up Timer* is an event timer with three modes, which can be set to start/stop/reset either automatically or manually. Using Source Profiles, you can determine whether individual sources will start or stop this timer automatically; the onscreen controls described below allow the operator to override or modify those settings.

» Rotate the **Timer Mode** SoftKnob to choose how the event timer will operate.

* Choose **Manual** to start, stop and reset the timer by hand using the controls displayed when the **Timer Options** key is pressed.
* Choose **Auto-Reset** and the timer will automatically reset to zero and begin counting whenever a fader whose source starts the timer is turned **On**.
* Choose **Auto-Add** to have the timer count up when a designated source's fader is turned **On** and stop counting when that source is turned **Off.** In this mode, the timer will not reset to zero when it is started.

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

Note: Pressing and holding the **Timer Options** Key opens the **Clock & Timer Options Menu**

©2008 Axia Audio — Rev. 3.0

discussed in Chapter 1.

- **Reset Key:**

Pressing this key resets Element's Count Up Timer to zero.

- **Freeze Key:**

Pressing **Freeze** holds the on-screen Timer display so you may take note of it; the timer continues to count while the display is frozen, and a second press of the **Freeze** again displays the running elapsed time counter.

- **Start/Stop Key:**

If the Count Up Timer is not running, pressing this 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 so that talent can record a phone call or remote interview with a single button-press. **Record Mode** can be enabled or disabled for each **Show Profile** (see Chapter 5, "Show Profiles", for details).

Figure 4-20:Monitor + 2-Fader Module + SoftKnobs

- **Engage Key:**

Press this key to put Element into **Record Mode**. Pressing this button activates a pre-determined recording device. Pressing the key a second time deactivates **Record Mode** and stops the recorder.

"Magic Key" Sequences for Standard Monitor Module

The list of "Magic Key" sequences for the Standard Monitor Module are:

- **IP Address Book:** Press and hold the **Fader Options** key for 5 seconds.
- **Clock & Timer Settings:** Press and hold the **Timer Options** key for 5 seconds.
- **Capture Mode:** Press and hold the **Help** key for 5 seconds.
- **Test Mode:** Press and hold the **Help** and **Right Arrow** keys for 5 seconds.

Monitor + 2-Fader Module Controls

The Monitor + 2-Fader Module (Figure 4-20) allows users to customize their console design, adding two full-fuction faders at the cost of some direct-access control keys in the Monitor section. No flexibility has been sacrificed, however; all the functions provided by the Standard Monitor Module are present, but are accessed differently. Controls for all of Element's options and features are accessed via on-screen menus with the aid of SoftKnobs; constantly-used controls like headphone and monitor selection and volume, plus command keys for the on-screen timers and a numeric keypad for convenient dialing of telephone systems and codecs are included. Since there are less physical controls present on the Monitor + 2-Fader Module than on the Standard Monitor Module, changes are accomplished using the module's SoftKnobs.

©2008 Axia Audio — Rev. 3.0

Figure 4-21: Channel Options Screen

Channel Options Controls

These controls allow the operator to customize the behaviour of individual faders. The operator first presses the **Options** key atop the fader s/he wishes to modify; the Channel Options box is displayed on Element's screen (Figure 4-21), and the SoftKnobs become active.

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 4-6) 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.

Figure 4-21 shows the Channel Options screen that's displayed when any fader's **Options** key is pressed, with the functions of the SoftKnobs shown below:

» Rotating the **Function/Select** knob selects the options you can modify, highlighting the current function in red. Pressing this knob allows you to modify the selected item; i.e., pressing when the Source box is highlighted (as in Figure 4-21) takes you to the Source Selection screen.

» Pressing the **Exit** knob returns you to Element's main display screen.

Channel Jumping: If any fader is in **Options** mode, you can "quick jump" to set options for other fader channels by pressing the Options key at the top of their channel strips.

For example, if you need to change the **Source** on several channels, select the first channel using its **Options** key, and then choose the **Channel Options Source** box. 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, and you're now adjusting settings for the new channel. The channel number is displayed onscreen as confirmation.

- **Source Selection Controls**

Pressing **Select** when the **Source** box is highlighted displays the Source Selection menu, which displays the currently selected source alongside a list of available sources operators may select using the SoftKnobs. They can also preview a source before selecting it.

Figure 4-22: Source menu screen.

©2008 Axia Audio — Rev. 3.0

Figure 4-23: Channel Options screens for the Monitor + 2-Fader Module. Information displayed differs based on the type of source currently assigned to the active fader. Clockwise from top left: Line Source Channel Options screen, Mic Source Channel Options screen, Phone Source Channel Options screen, Codec Source Channel Options screen.

©2008 Axia Audio — Rev. 3.0

A Note About Source Sharing: Element's source sharing protocol keeps sources from being controlled by more than one console at a time.

If an operator tries to assign a source that's already in use on another console and has a backfeed or logic control enabled, the ID box in the Channel Options screen informs him that he has "Listen Only" rights to the source (as seen in the righthand box above) - no backfeed will be sent from his console, and no machine logic is tied to the fader's **ON/OFF** keys.

If the source is not in use anywhere else, the operator is informed that they have "Full Control" of the source (as shown in the lefthand box above), and backfeed and logic operate normally.

» **Select/Take**: rotating this SoftKnob scrolls through the available sources. Push to take.

» **Preview** allows you to hear the highlighted source through the Preview speakers without haveing to "take" it. Press and hold to listen; release to end the preview.

» Pressing the **Exit** Soft Knob returns you to the Channel Options screen.

- **EQ Controls**

Press the **Select** SoftKnob when the **EQ Curve** box is highlighted to adjust three-band parametric equalization for the selected audio source. 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 amount of cut/boost applied, and the center frequency of each band.

» Rotate the **Function/Exit** SoftKnob to highlight the **Lo, Mid** or **Hi** band knobs shown onscreen. Notice that the left and center SoftKnob labels change to display **Lo Gain/Lo Freq, Mid Gain/ Mid Freq** and **Hi Gain/Hi Freq** as you rotate the **Function** knob.

» The leftmost SoftKnob is the **Gain** control; rotate to adjust the amount of gain or cut applied to the selected band.

* Note that when you adjust **Lo Gain**, the Shelf/ Peak indicator next to the onscreen control

Figure 4-22: Source menu screen.

Figure 4-23: EQ menu screen.

©2008 Axia Audio — Rev. 3.0

automatically shifts to Peak mode when boost is applied, and shifts to Shelf mode when cut is applied.

* Note that when you select the **Hi** band, the label below the **Gain** control displays **Hi Peak** or **Hi Shelf**, depending on which mode has been selected. Push the knob to toggle between the two modes.

» The center SoftKnob is the **Freq** control and adjusts the center frequency of the selected EQ band.

* In **Lo Freq** mode, rotating the knob adjusts the center frequency of the low EQ band (25 - 400 Hz).
* In **Mid Freq** mode, rotating the knob adjusts the center frequency of the midrange EQ band (100 Hz - 1 kHz).
* In **Hi Gain** mode, the knob adjusts the boost/cut of the high frequency band.

Pressing the center SoftKnob toggles EQ between **Active** and **Bypass.**

» Pressing the **Function/Exit** SoftKnob returns you to the Channel Options 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 44 of this manual.

- **Dynamics Controls**

Dynamics controls are available if the selected source is a Mic, Codec or Phone input. Pressing the **Select** SoftKnob when the **Dynamics** box is highlighted allows you to adjust **Gating, Compression** and **De-essing** for 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.

Figure 4-24: Dynamics menu screen.

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!

©2008 Axia Audio — Rev. 3.0

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.

» Rotate the **Function/Exit** SoftKnob to highlight the **Exp** (Expansion), **Comp** (Compression) or **De-Ess** sections of the onscreen display. Note that the labels above and below the left and center SoftKnobs change based upon which section is highlighted.

» When the **Expansion** controls are active:

  * Rotating the **Exp Thresh/Active/Bypass** 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 Depth** knob sets the amount of gain reduction when audio level falls below the **Exp Gate Threshold** value (adjustable between -3 and 0dB).

» When the **Compression** controls are active:

  * Rotating the **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. When the Freeze function is active, a red "lamp" illuminates just above the Compression Ratio bargraph meter.

» When the **De-Ess** controls are active:

  * Rotating the **De-Ess Thresh/Bypass/Active** knob 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.
  * Rotating the **De-Ess Ratio** knob sets the amount of sibilance reduction, in dB, that will be applied (adjustable between 1:1 and 8:1).

» Pressing the **Function/Exit** SoftKnob returns you to the Channel Options screen.

- **Pan & Mode Controls**

Pressing the **Select** SoftKnob when the **Pan & Mode** box is highlighted allows the operator to adjust the pan or balance of the channel source and correct signal phase errors.

It's a Lock: If a "padlock" icon appears in any options screen, that function is "locked out" in

Figure 4-25: Pan & Mode menu screen.

©2008 Axia Audio — Rev. 3.0

the Source Profile and adjustments cannot be made. In Figure 4-10, the padlock appears just below the soundfield representation, above the Mode indicator. The onscreen Mode SoftKnob is also "grayed out" to show that this function is unavailable.

Operators cannot "unlock" locked functions; if you wish to make locked functions available, simply modify the source profile to allow them and when the source is reloaded to the console, the padlock will have disappeared.

» Rotating the **Function/Exit** knob selects between **Pan, Mode** and **Phase** controls. The active control is highlighted onscreen, and the labels of the SoftKnobs change depending on which function is active.
  * When the **Pan/Balance** mode is active, rotating the SoftKnob adjusts the the Pan setting for mono sources and the Balance setting for stereo sources. Push the knob to **Take** your new settings.
  * When the **Mode** mode is active, rotating the SoftKnob 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). Push to **Take** the new setting.
» When the **Phase** mode is active:
  * If the selected source is *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 Both** reverses phase of both stereo channels. Press to **Take** the new setting.
  * If the selected source is in *mono*, the choices are **Normal** and **Invert** only. Press to **Take** the new setting.
» Pressing the **Function/Exit** SoftKnob returns you to the Channel Options screen.

- **Aux Sends Controls**

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 the **Select** SoftKnob when the **Aux Sends** boxes are highlighted takes the operator to the Aux Sends control screen.

Figure 4-26: Aux Sends menu screen.

» Rotating the **Select/Exit** knob selects between **Aux Send A, B, C** & **D**. The active control is highlighted onscreen, and the labels of the SoftKnobs change depending on which Send is active.
» **Send A/B/C/D / Pre/Post Fader:**
  * Rotate this knob to adjust the level of the audio you'll be sending to the selected Aux Send bus.
  * Pushing the Knob to toggle the activity state of the selected **Send.** The state is indicated in the onscreen display above.
    › **On Pre** sends the source assigned to the fader to the selected Aux Send bus *prior* to having its gain adjusted by the fader (fixed level).
    › **On Post** sends the source assigned to the fader to the selected Aux Send bus *after* having its gain adjusted by the fader (variable level).

©2008 Axia Audio — Rev. 3.0

› **Off** sends no audio to the Aux Send bus.

» Pressing the **Function/Exit** SoftKnob returns you to the Channel Options screen.

- **Feed To Source Controls**

If the assigned source (i.e., a codec or phone hybrid) has an associated backfeed or mix-minus, Pressing the **Select** SoftKnob when the **Feed To Source** box is highlighted takes the operator to the Feed To Source control screen. This selection is normally pre-set when a Source Profile is constructed (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 supply backfeed or mix-minus. Press to **Take** the highlighted source. Your choices are as follows:

* **Auto** backfeeds audio from the Phone bus when the channel is **Off**, and audio from the Program-1 bus when the channel is **On**, with mix-minus if needed.
* **Phone** backfeeds audio from the Phone bus, with mix-minus if needed.

Figure 4-27: Feed to Source menu screen.

* **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.

About the PHONE and PROGRAM 4 buses:
Element's Program 4 mix bus is a special multi-purpose bus that does several things at once to make life easier for the operator.

Any source assigned to Program 4 also be fed to the PHONE bus and sent to callers when the channel is **OFF** and **AUTO** is selected as the **Feed To Source Mode**. PHONE is an internal bus which does not feed a separate output; instead, it picks up the Program 4 audio pre-fader and pre-On/Off and backfeeds them to an PHONE sources.

Sources assigned to Program 4 are also sent to the RECORD bus, a special fixed-level output that can be used to feed recording devices.

» 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 the **Exit** SoftKnob returns you to the Channel Options screen.

Monitor Section

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

- **Monitor Options Key**

This key lets the operator access many options relating to Monitor and Headphone behavior. There are two different Monitor Options screen; the first is shown when the key is initially pressed, and a second is display when the key is pressed another time.

The first Monitor Options screen is shown in Figure 4-28:

» Rotating the **Function/Exit** SoftKnob highlights the different control boxes onscreen, making the highlighted control active. Labels on the other SoftKnobs change depending on context.

©2008 Axia Audio — Rev. 3.0

Figure 4-28: Monitor Options menu screen #1.

Figure 4-29: Headphone EQ menu screen.

» When the **Talk to Monitor 1 Talk** and **Dim** or **Talk to Monitor 2 Dim** control is highlighted, rotating the **TALK LEV** SoftKnob adjusts the levels of the **Monitor 1 Talk To** and **Dim** levels and the **Monitor 2 Dim** level.

» When the **Headphones Follow Monitors** box is highlighted, rotating the **HP LINK/Take** chooses whether the headphone source will follow the Monitor 1 source selection. Push the knob to **Take** your selection.

* **Linked** means that Headphone and Monitor 1 selections will mirror each other; making a selecton from one bank of keys on the Monitor Module will automatically choose the same source on the other bank.
* **Unlinked** allows Monitor 1 and Headphone sources to be selected independently of each other.

The strongest link: You've noticed that there are no Direct Selection Keys for the headphone source on the Monitor + 2-Fader Module. By default, CR Headphones and Monitor 1 sources are linked - the phones will play whatever source is assigned to the monitor speakers. So how can the operator select a headphone source if s/he choosees to **Unlink** the headphones?

When **Unlinked** is chosen in the Headphones Follow Monitors box, the **Preview** knob doubles as the headphone source selector. When the operator pushes the **Preview** knob, the **Headphone Source** menu is displayed. Voila!

» When the **Headphone Processing** box is highligted, pressing the **HP Proc/Take** knob selects **Change Settings** onscreen and takes the operator to the **H/P Process** screen shown in Figure 4-29. 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 amount of cut/boost applied, and the center frequency of each band.

* Rotate the **Function/Exit** SoftKnob to highlight the **Lo, Mid** or **Hi** band knobs shown onscreen. Notice that the left and center SoftKnob labels change to display **Lo Gain/Lo Freq, Mid Gain/Mid Freq** and **Hi Gain/Hi Freq** as you rotate the **Function** knob.
* The leftmost SoftKnob is the **Gain** control; rotate to adjust the amount of gain or cut ap-

©2008 Axia Audio — Rev. 3.0

plied to the selected band.

› Note that when you adjust **Lo Gain**, the Shelf/Peak indicator next to the onscreen control automatically shifts to Peak mode when boost is applied, and shifts to Shelf mode when cut is applied.

› Note that when you select the **Hi** band, the label below the **Gain** control displays **Hi Peak** or **Hi Shelf**, depending on which mode has been selected. Push the knob to toggle between the two modes.

* The center SoftKnob is the **Freq** control and adjusts the center frequency of the selected EQ band.

› In **Lo Freq** mode, rotating the knob adjusts the center frequency of the low EQ band (25 - 400 Hz).

› In **Mid Freq** mode, rotating the knob adjusts the center frequency of the midrange EQ band (100 Hz - 1 kHz).

› In **Hi Gain** mode, the knob adjusts the boost/cut of the high frequency band.

Pressing the center SoftKnob toggles EQ between **Active** and **Bypass.**

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 44 of this manual.

» When the **Preview to Headphones** box is highlighted, the **HP Prev/Take** knob controls whether audio from the **Preview** bus is fed to the board op's headphones, and if so, in what manner. Rotating this knob selects from the following options:

* **Off:** The operator will not hear Preview audio in the headphone channel.
* **On:** 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.
* **Split:** 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.
* After making a selection, push the knob to **Take** the new value.

» When the **Monitor Mode** box is highlighted, the operator can rotate the **Monitor/Take** knob to select from the following options.

* **Stereo:** The selected source will be presented in stereo on the Monitor speakers.
* **Left:** The left channel of the selected source will be presented in both Monitor speakers.
* **Right:** The right channel of the selected 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.

» When the **Headphone Mode** box is highlighted, the **HP Mode** knob is active. This control works the same as the **Monitor Mode** control described above, but affects the board operator's headphone channel.

» Pushing the **Function/Exit** knob returns you to the main Element screen.

The second Monitor Options screen (press the **Monitor Options** key a second time) is shown in Figure 4-30:

» Rotating the **Function/Exit** SoftKnob highlights the different control boxes onscreen, making the highlighted control active. Labels on the other SoftKnobs change depending on context.

» When the **Rightmost Meter** box is highlighted, rotating the **Met Mode/Take** knob selects how the far right meter on Element's main screen will behave. Pressing this knob will **Take** your selection.

* **Dedicated** means the meter will be slaved to

©2008 Axia Audio — Rev. 3.0

Figure 4-30: Monitor Options menu screen #2.

the **PGM-4** bus.

* **Follow Monitors** means the meter will show levels for any source assigned to **Monitor 1.**

» When the **Fader Options** box is highlighted, pressing the **St. Mode** SoftKnob selects between fader-normal and European-style fader-start channel activation methods.

* **Fader Normal:** Selects the US method of requiring talent to manually turn channels on and off.
* **Fader Start:** Selects the European convention of activating the channel and associated machine logic when the fader is raised from ∞.
* **Retain Source Setting:** Each individual fader uses the settings specified in its Source Profile.

» Element features four stereo **Aux Send** buses for use as utility, FX or IFB buses. When the knobs in the **Aux Send Masters** box are highlighted, the **SEND A/B/C/D** knob affects global gain settings for the master levels of the four **Aux Send** buses.

* The **Send A/B/C/D** SoftKnobs, when rotated, set the Master Send Gain for each **Aux Send** bus. Pushing the knob toggles the selected **Aux Send Master On** or **Off.**
* Pushing the **Preview** knob sends the highlighted **Aux Send** mix to the **Preview** channel. Push to preview; release to end preview.

» Element features two stereo **Aux Return** buses whose output can be applied to any of the four **PGM** buses. When the knobs in the **Aux Returns** box are highlighted, the **Return A/B** knob affects global gain settings for the master levels of the two **Aux Return** buses.

* The **Return A/B** SoftKnobs, when rotated, set the Master Send Gain for each **Aux Send** bus. Pushing the knob toggles the selected **Aux Return Master On** or **Off.**
* Rotate the adjacent **Assign/Take** knob to choose which of the four **Program** buses (or the **Preview** bus) to assign the output of the selected **Aux Return.** Push the knob to **Take** the assgnment.

» Pressing the **Function/Exit** SoftKnob at any time returns you to Element's main meter screen.

- **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** 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

©2008 Axia Audio — Rev. 3.0

External selector keys - like preset buttons on a car radio.

» Press and hold both **External** selector keys to open the **Monitor 1 Source** menu, which allows you to select from a list of Program and Aux Send buses.

Programming the External Monitor Selector keys: 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.

You can also pre-program the source assigned to these keys when you construct or capture Show Profiles, using the options available on the **Monitor** page of the Show Profile setup menu.

- **Headphone Knob**

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

» Rotate to vary the gain of the **Headphone** feed.

» By default, the **Headphone** feed is linked to and follows the **Monitor 1** source selection. To unlink the two and choose an independent headphone source, please see the description on Page 66 of this manual.

- **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.

There are three different Preview to HP modes; the current mode is indicated by an annunciator on the Element display screen just below the headphone selection tally.

» **Off:** When Preview to HP is off, the selector key is unlit. No preview audio will be heard in the board-op's HP path.

» **On:** Pressing the **Preview to HP** key once from the Off state sends preview audio, when present, into both sides of the operator's headphone feed, overriding the Monitor audio. The **Preview to HP** key is now lit.

» **Auto:** Pressing the **Preview to HP** key a second time toggles Auto mode. In this mode, 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 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.

The **Preview** knob is also used to select the source of the CR Headphone feeed when headphones are not linked to the CR Monitor selection. Please see Page 66 of this manual for more information on this function.

- **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 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.

Other Controls

Keys in this section are used to load Show Profiles,

©2008 Axia Audio — Rev. 3.0

control the onboard clock and timers, engage Record Mode and invoke hidden setup screens.

- **Profile Key**

Pressing this key allows the board operator to browse and select from a 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.

LOADING A NEW SOURCE WHILE ANOTHER SOURCE IS STILL ACTIVE: Element will never interrupt an active source in order to load a new one.

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 previously loaded Profile until you exit the Show Profile screen. The **Previous** command functions as a "speed key," highlighting the previously assigned source in the source list and allowing the operator to return to it should they change their mind. Press to "take."

Figure 4-31: Load Profile screen.

» **Exit** returns the operator to Element's main operating screen.

DON'T PANIC! 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.

Pressing and holding the **Profile** key for five seconds instantly clears changes made to the current Show Profile, resetting the Show Profile's default settings. All is well, and your jock once again knows where his towel is.

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 allows users to adjust how the onscreen timers behave. Pressing the **Timer Options** key *once* opens the *Count Up Timer* controls; pressing it *twice* opens the *Countdown Timer.*

©2008 Axia Audio — Rev. 3.0

Figure 4-32: Timer controls.

Note: Pressing and holding the **Timer Options** Key opens the **Clock & Timer Options Menu** discussed in Chapter 1.

The *Count Up Timer* is an event timer with three modes, which can be set to start/stop/reset either automatically or manually. Using Source Profiles, you can determine whether individual sources will start or stop this timer automatically; the onscreen controls described below allow the operator to override or modify those settings.

» Rotate the **Timer Mode/Reset** SoftKnob to choose how the event timer will operate. Your choice will be displayed onscreen just below the Count Up Timer readout.
  * Choose **Manual** to start, stop and reset the timer by hand using the controls displayed when the **Timer Options** key is pressed.
  * Choose **Auto-Reset** and the timer will automatically reset to zero and begin counting whenever a fader whose source starts the timer is turned **On**.
  * Choose **Auto-Add** to have the timer count up when a designated source's fader is turned **On** and stop counting when that source is turned **Off.** In this mode, the timer will not reset to zero when it is started.

» Press **Timer Mode/Reset** to return the Count Up Timer to zero.

» Press **Start/Stop** to begin and end the Timer's countdown.

» Pressing **Freeze** holds the on-screen Timer display so you may take note of it; the timer continues to count while the display is frozen, and a second press of the **Freeze** again displays the running elapsed time counter.

The *Countdown timer* is useful for counting down to network joins, top-of-hour IDs, and other events which require exact timing. The Countdown timer is started, stopped and reset manually using the controls displayed when the Timer Options key is pressed. To use:

» Rotate the SoftKnob marked **Begin Time/Reset** to set the interval from which the countdown will begin. As you rotate, small numbers below the red counter onscreen will inform you of the new countdown interval.

» Press the **Begin Time/Reset** knob to set the new time you've dialed in.

» Press the **Start/Stop** knob to begin or end the countdown.

» Pressing **Freeze** holds the on-screen Timer display so you may take note of it; the timer continues to count while the display is frozen, and a second press of the **Freeze** again displays the running elapsed time counter.

"Look at me!!" During the last 60 seconds of the countdown, the inner ring of Element's onscreen analog clock will turn red and diminish as time passes, providing the operator with a visual cue to how much time is left until zero.

- **Reset Key:**
  Pressing this key resets Element's Count Up Timer.

- **Freeze Key:**
  » Pressing **Freeze** holds the Count Up Timer display so you may take note of it; the timer con-

©2008 Axia Audio — Rev. 3.0

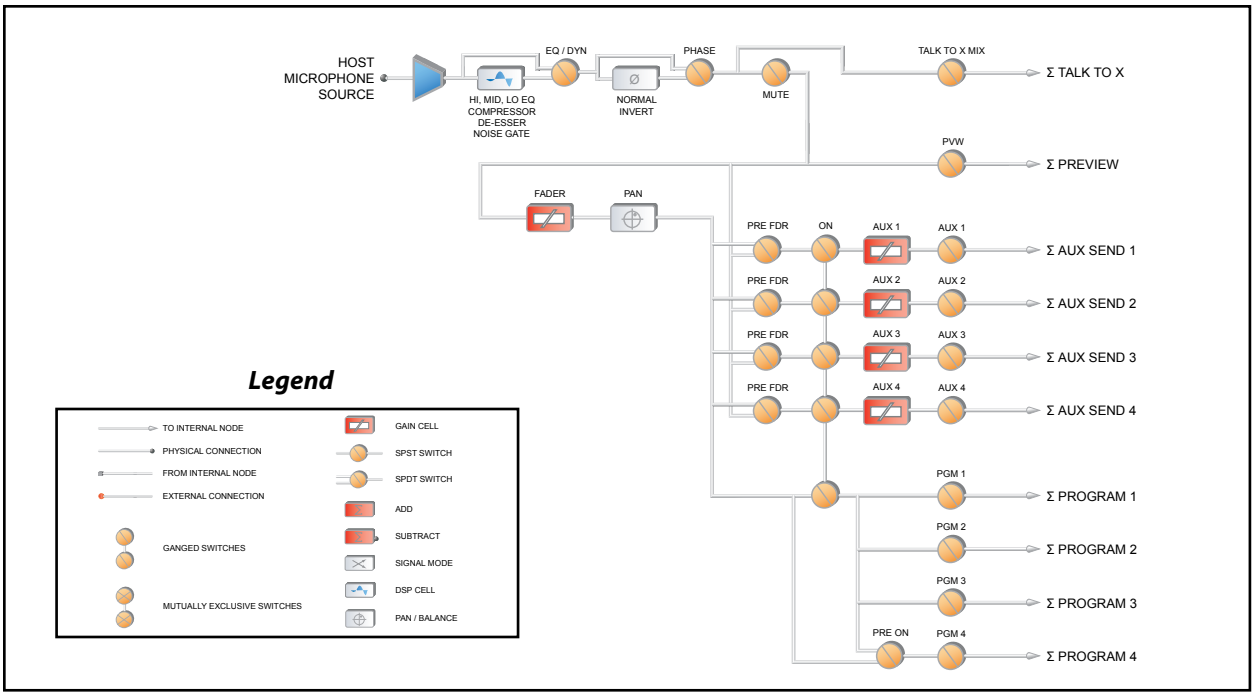


Figure 4-33: Control Room Operator Microphone channel block

tinues to count while the display is frozen, and a second press of the **Freeze** again displays the running elapsed time counter.

- **Start/Stop Key:**

If the Count Up Timer is not running, pressing this starts it. 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 so that talent can record a phone call or remote interview with a single button-press. **Record Mode** can be enabled or disabled for each **Show Profile** (see Chapter 5, "Show Profiles", for details).

- **Engage Key:**

Press this key to put Element into **Record Mode**. Pressing this button activates a pre-determined recording device. Pressing the key a second time deactivates **Record Mode** and stops the recorder.

"Magic Key" Sequences for Monitor + 2-Fader Module

The list of "Magic Key" sequences for the Standard Monitor Module are:

- **IP Address Book:** Press and hold the *** (star), 4** & **7** keys on the keypad for 5 seconds.
- **Clock & Timer Settings:** Press and hold the **Timer Options** key for 5 seconds.
- **Capture Mode:** Press and hold the *** (star)** & **2** keys on the keypad for 5 seconds.
- **Help Screen:** Press and hold the *** (star)** & **# (pound)** keys on the keypad for 5 seconds.
- **Test Mode:** Press and hold the **Help** and **Right Arrow** keys for 5 seconds.

Source-Specific Channel Controls

Some fader functions change their behavior to suit the type of source assigned to the fader. A Guest 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.

©2008 Axia Audio — Rev. 3.0

Control Room Operator Mic Channel Operation

The Control Room Operator microphone is the board operator's mic. It's always located in the Control Room, so activating a channel designated as the CR Operator mic affects Studio 1 monitor speaker and Preview speaker muting.

When the board operator pushes a **Talkback** key, all output bus assignments for the Control Room Operator 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 Operator 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 Operator microphone channel will mute both the Studio 1 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 4-33.

Producer Microphone Channel Operation

The Producer's microphone is meant to be used at position within the Control Room where a morning show or talk show producer would normally sit. This mic type mimics the operation of the Control Room mic described above, but does not include the "cough" function; also, a Producer mic can use Talkback to talk to the board op, but cannot talk directly to the Monitor 2 talkback channel.

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 an announcer located physically near the board op. These microphones control the muting of the Control Room monitor speakers.

The Studio 1monitor speakers and Preview speakers will mute when this channel is turned on, or when **Preview** or **Record Mode** are selected. 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 Status Symbol on the guest's fader strip displays a T icon. When the user stops talking, the channel returns to normal. The operator can then reply using the fader's **Talkback** key.

When the **Remote Mute** function is activated, the Status Symbol displays ⊠, 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 **Talkback** key on their fader strip.

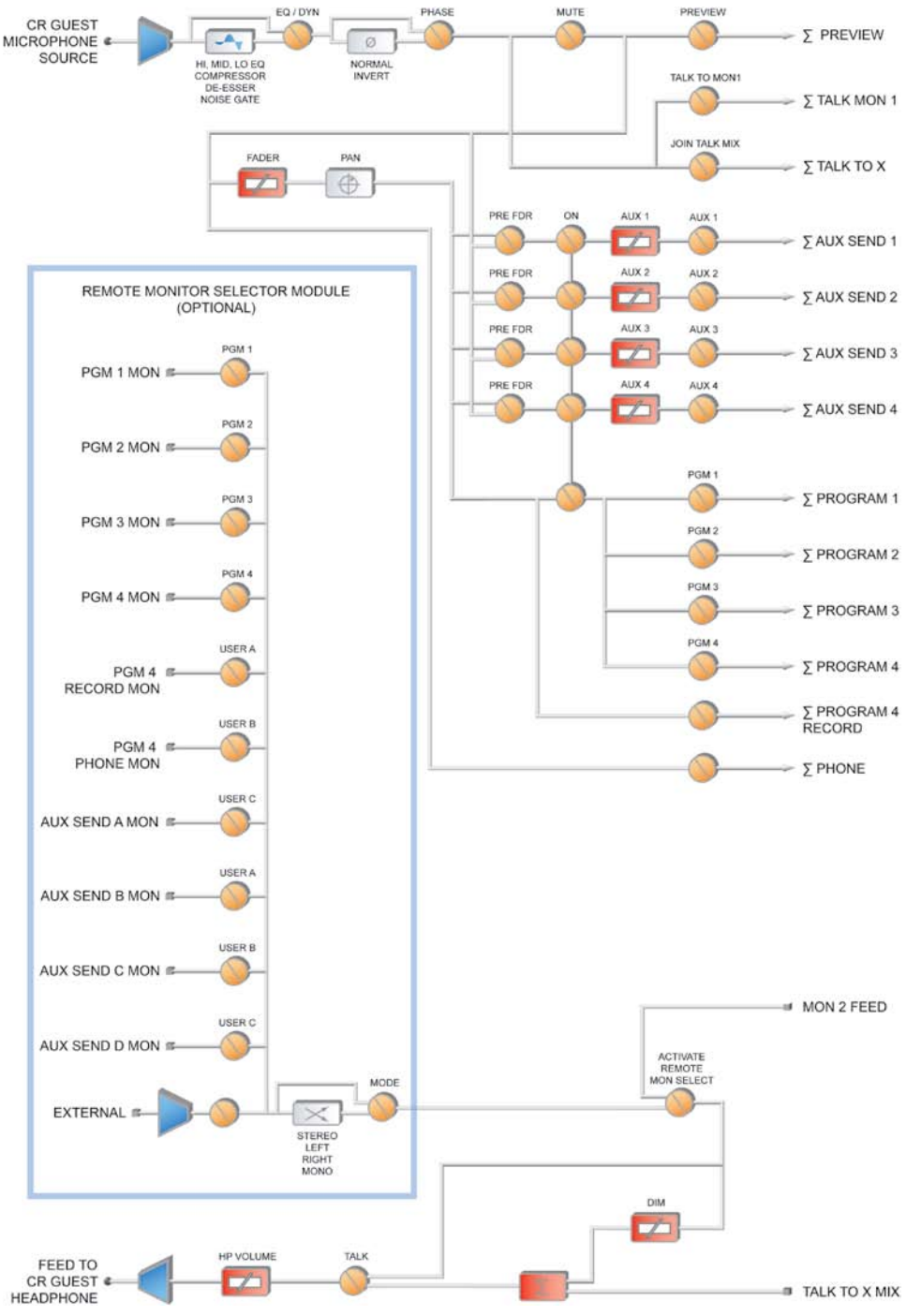
The block diagram for this channel selection is shown in Figure 4-34.

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

©2008 Axia Audio — Rev. 3.0



DRAWING UPDATED
2-11-08

Figure 4-34: Control Room Guest Microphone channel block

©2008 Axia Audio — Rev. 3.0

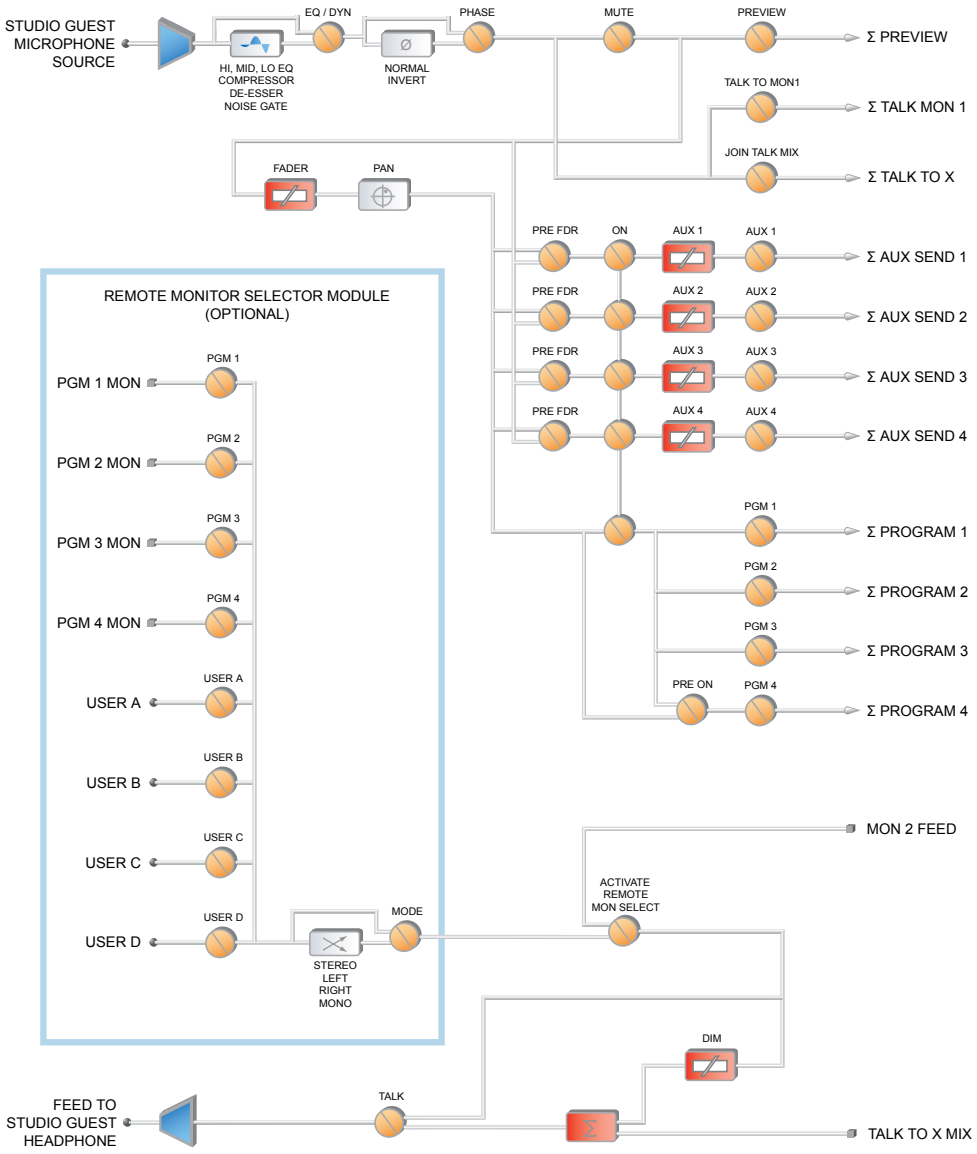


Figure 4-35: Studio Microphone channel block

©2008 Axia Audio — Rev. 3.0

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 device. **Remote Talk** and **Remote Mute** functions are providedfor 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 Microphone-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 4-35.

Line Channel Operation

Line-level devices like CD players, 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.

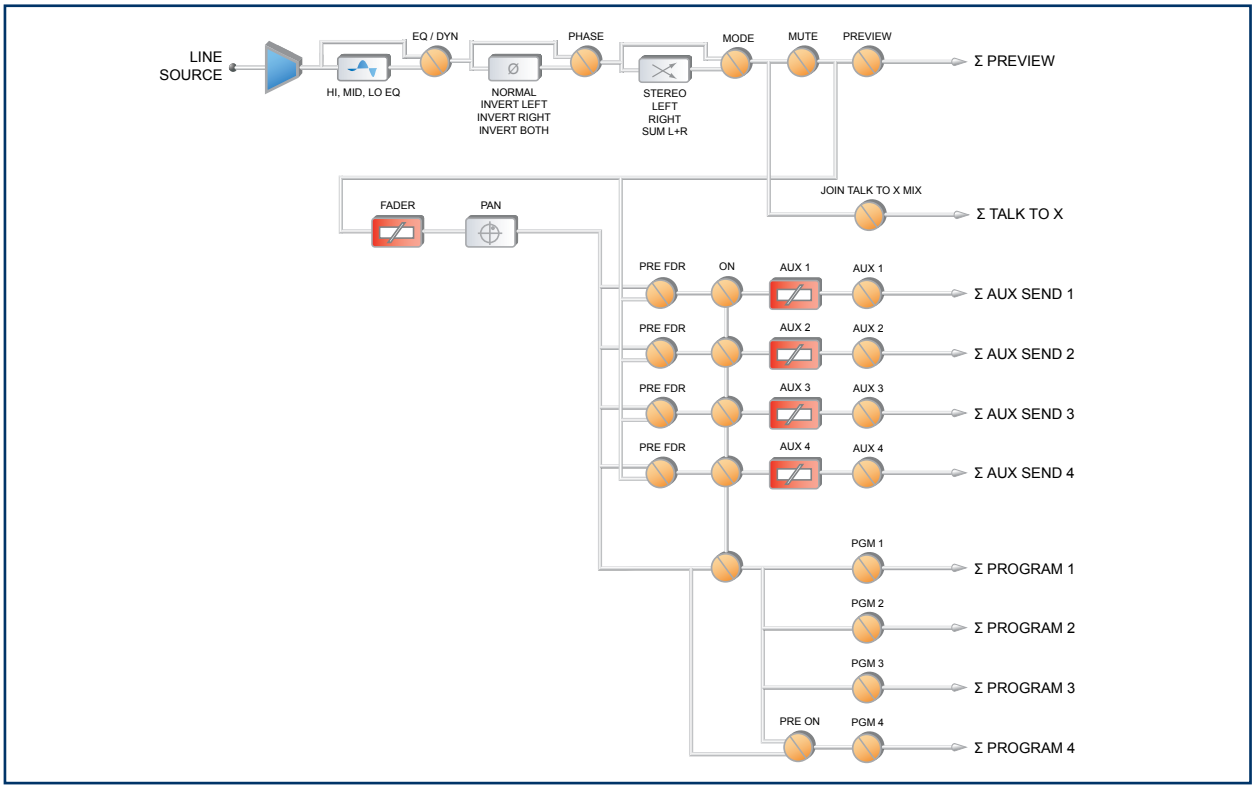
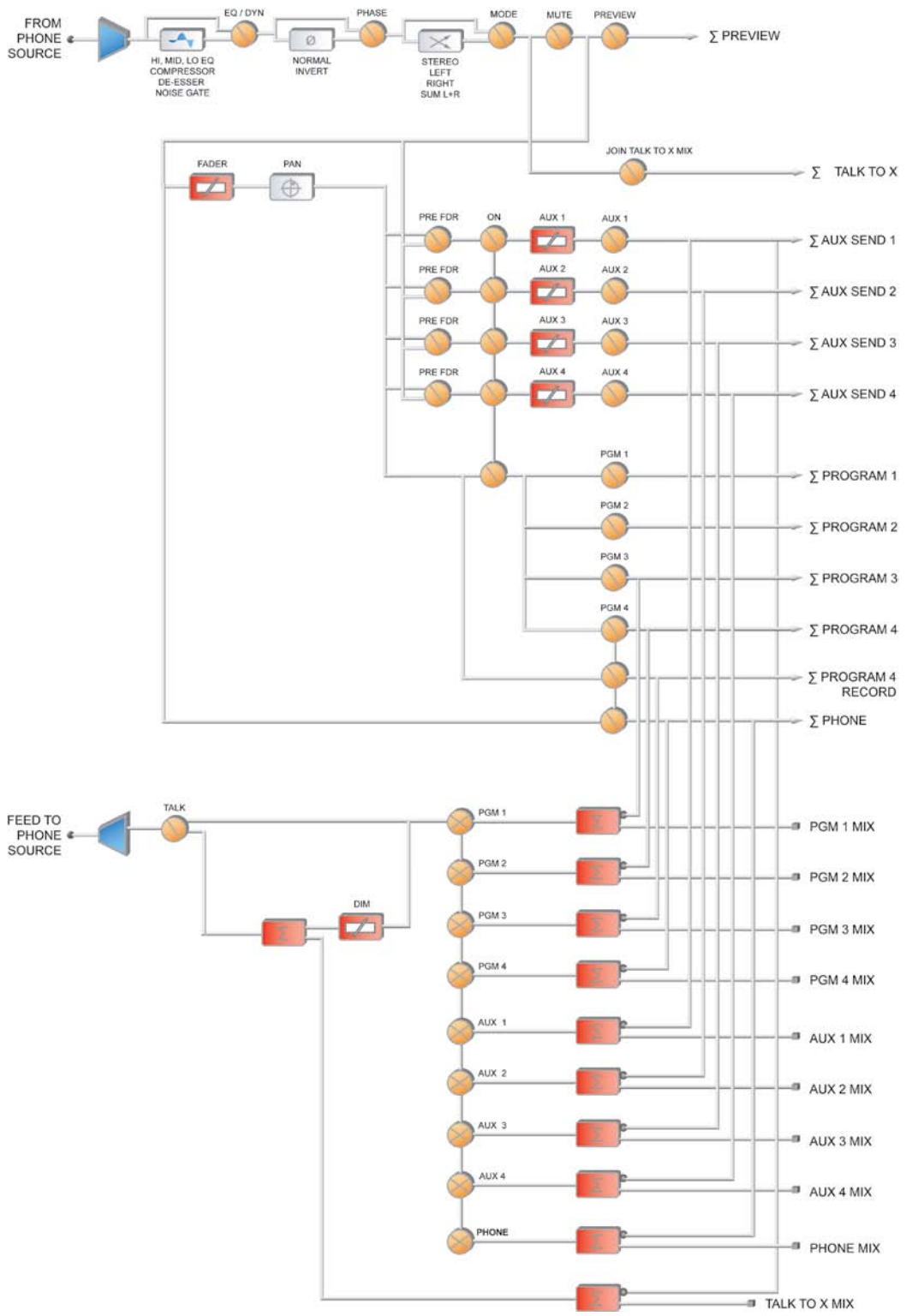


Figure 4-36: Line channel block

©2008 Axia Audio — Rev. 3.0



DRAWING UPDATED
2-11-08

Figure 4-37: Phone channel block

©2008 Axia Audio — Rev. 3.0

A **Reset** function is provided to remotely turn the channel **Off** while suppressing the **Stop** command, useful 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 4-36.

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 accomodate 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 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 outpus.

The block diagram for this channel selection is shown in Figure 4-37.

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. Codec Channels can be turned on and off remotely and provide on/off status to a remote logic device. **Remote Talk** and **Remote Mute** are also provided, allowing 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; the channel's **Talkback** key lights, and its Status Symbol displays T. 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 M, 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 4-38.

©2008 Axia Audio — Rev. 3.0

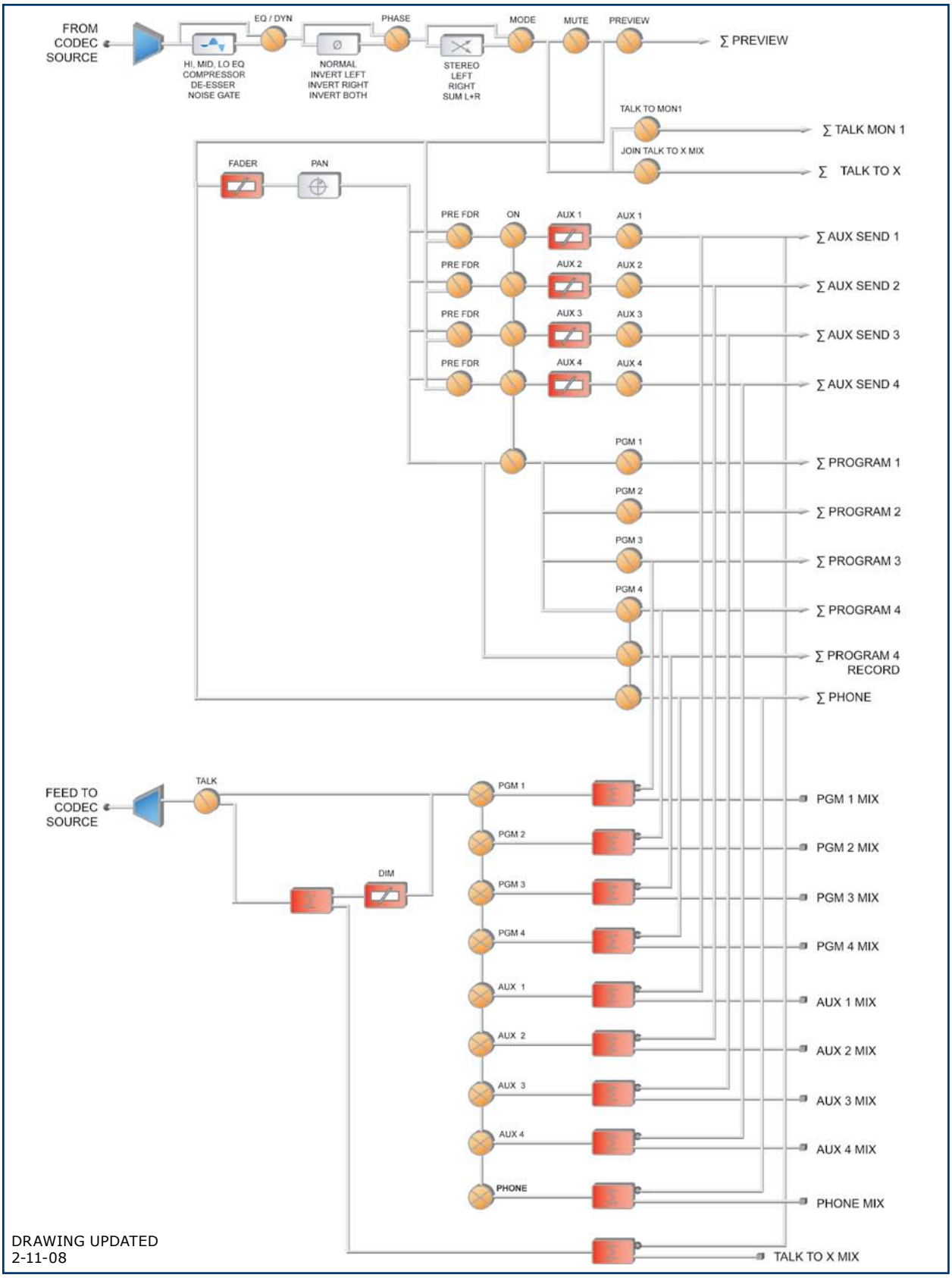
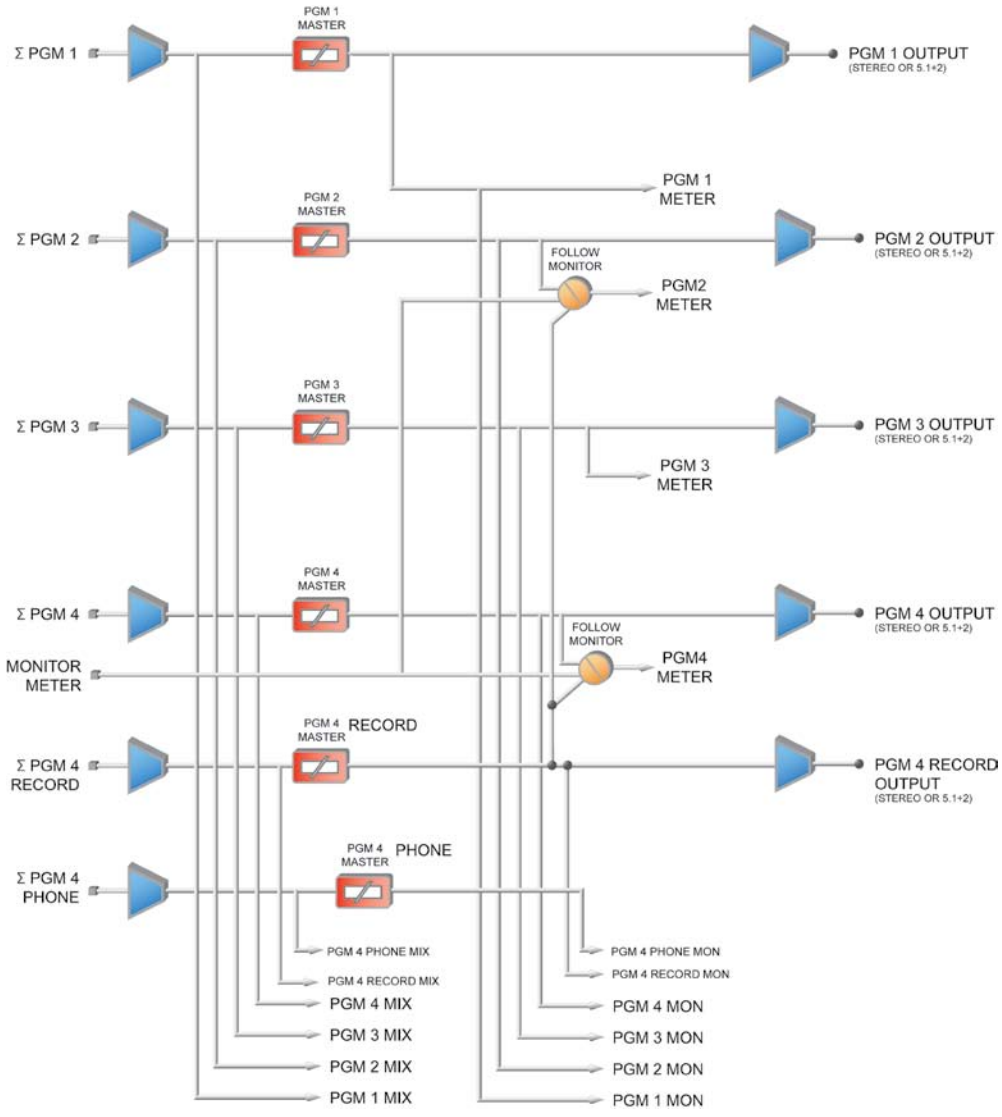


Figure 4-38: Codec channel block

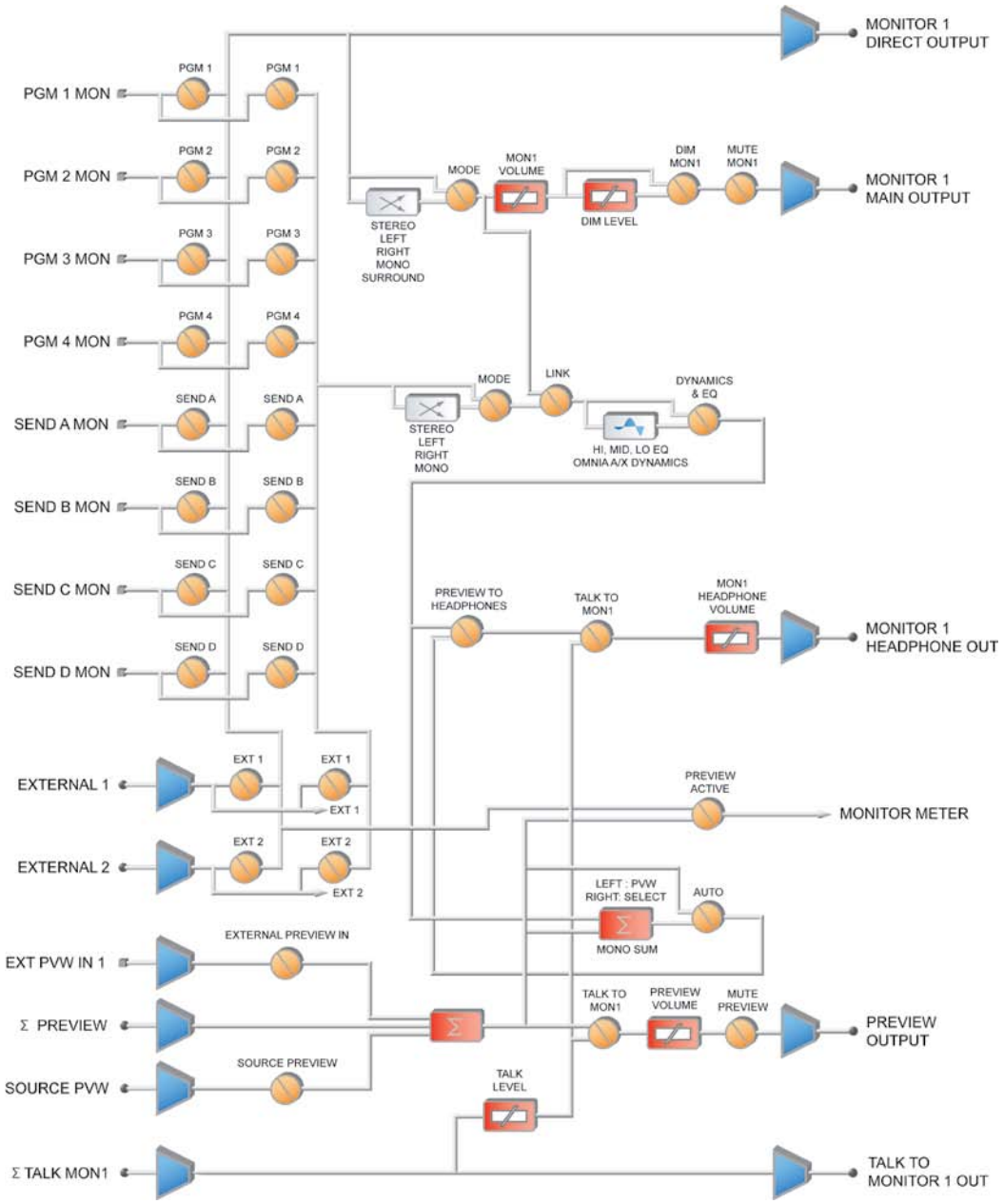
©2008 Axia Audio — Rev. 3.0



DRAWING UPDATED
2-11-08

Figure 4-39: Element Master Output block

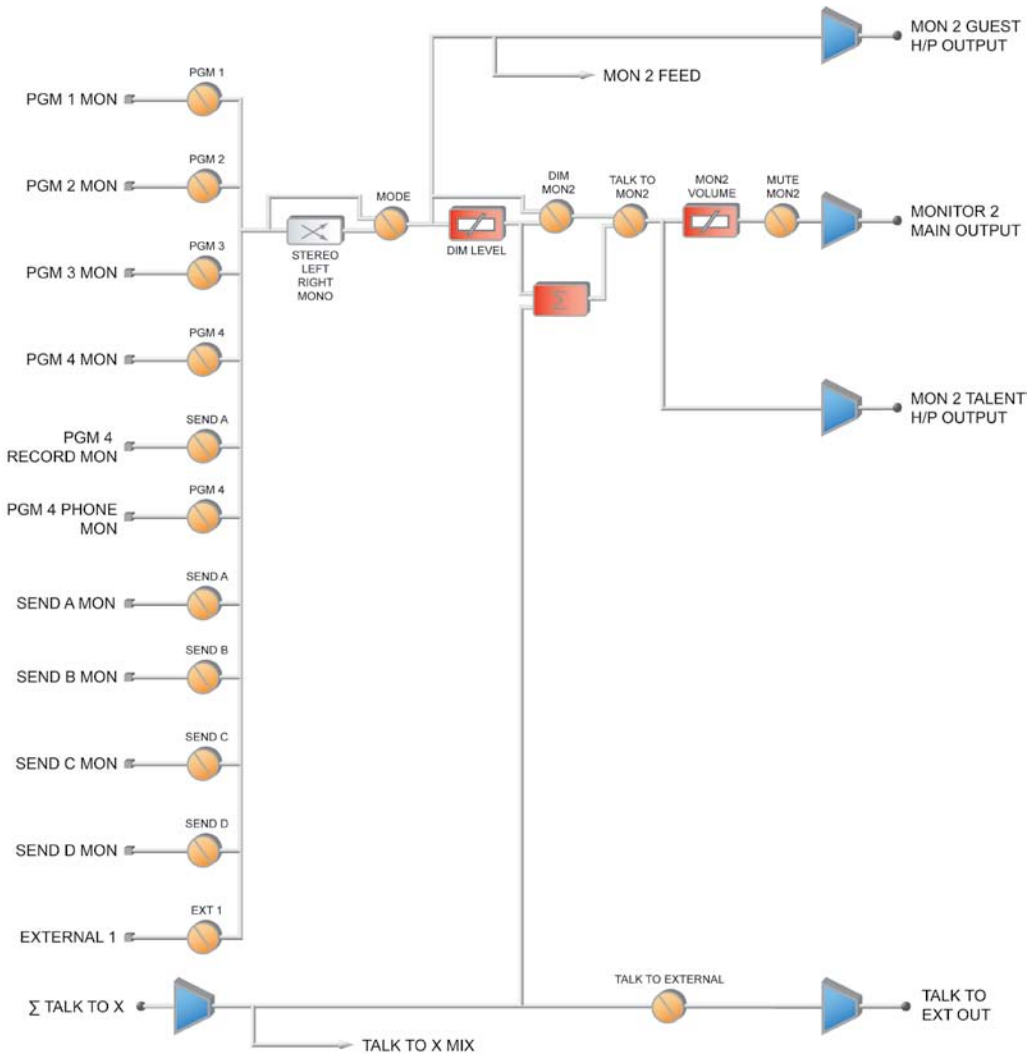
©2008 Axia Audio — Rev. 3.0



DRAWING UPDATED
2-11-08

Figure 4-40: Monitor 1 Output block

©2008 Axia Audio — Rev. 3.0



DRAWING UPDATED
2-11-08

Figure 4-41: Monitor 2 Output block

©2008 Axia Audio — Rev. 3.0

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 Five to learn how to set up and use Show Profiles.

©2008 Axia Audio — Rev. 3.0

Chapter Five:

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 Show 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 Show 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 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, Control Knob 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 proceedure 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, continue on and make monitor and headphone selections and adjust Aux Send / Aux Return settings if necessary, since these are all saved in Show Profiles as well.

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.

Enter Element's IP address in your Web browser. When the Main Menu appears, choose "Configuration"

©2008 Axia Audio — Rev. 3.0

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.

Under ths Show Profiles list, 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.

You've already set up your Element, so now you can just "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!

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.

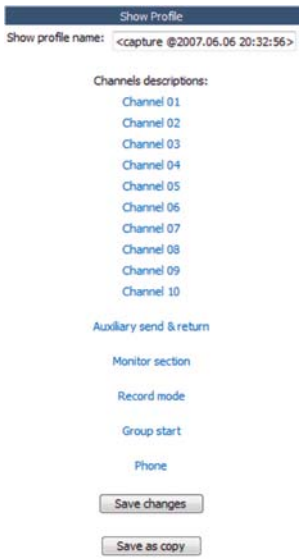


Figure 5-1: Show Profile Settings Menu

You'll see the menu shown in Figure 5-1. Notice that there is an entry for each of your Element's faders, as well as a link for **Auxiliary Send & Return**, **Monitor Section**, **Record Mode**, **Group Start** and **Phone.**

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 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

©2008 Axia Audio — Rev. 3.0

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.

Channel Description Section

- **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-1/2/3/4, AUX SEND A/B/C/D** 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 **Phone (PGM-4 pre-fader)** 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 LOCK: Unlock** or **Lock** the talent's ability to change the Signal Mode, or **Retain Source Setting.**

- **PANORAMA POSITION: Use** Pan/Balance adjustment 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:** Make EQ **Active** for this fader/dource combo, **Bypass** it, or **Retain Source Setting.**

- **EQ HIGH MODE:** Switches high-frequency EQ mode between **Shelf** and **Bandpass** (peak), 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 PGM1/2/3/4 and AUX A /B/C/D:** Checking **ON** for any of these buses assigns the fader 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 fader's source to the specified Aux Send bus either **Pre-Fader** or **Post-Fader,** or **Retain Source Settings.**

- **AUX SEND A/B/C/D GAIN:** Turn this channel's feed to the specified Aux Send bus **Off,** specify up to 10 dB of boost or 25 dB of cut, or **Retain Source Settings.**

- **NOISE GATE STATUS:** Turns the Noise Gate in the Omnia Dynamics Processing on and off. Specify **Active** or **Bypass**, or choose **Retain Source Setting.** (Only applies to Mic sources.)

- **COMPRESSOR STATUS:** Turns the compressor in the Omnia Dynamics Processing on and off. Specify **Active** or **Bypass**, or choose **Retain Source Setting.** (Only applies to Mic sources.)

- **DE-ESSER STATUS:** Turns the de-essing function in the Omnia Dynamics Processing on and off. Specify

©2008 Axia Audio — Rev. 3.0

Active or **Bypass**, or choose **Retain Source Setting.** (Only applies to Mic sources.)

- **NOISE GATE THRESHOLD:** Determines the point at which the Noise Gate will operate (if set to **Active**). Adjustment range is between 0dB and -50dB, or **Retain Source Settings**.

- **COMPRESSOR THRESHOLD:** Determines the point at which the Compressor will operate (if set to **Active**). Adjustment range is between 0dB and -30dB, or **Retain Source Settings**.

- **DE-ESSER THRESHOLD:** Determines the point at which the Compressor will operate (if set to **Active**). Adjustment range is between 0dB and -20dB, or **Retain Source Settings**.

- **COMPRESSOR RATIO:** Sets the amount of compression that will be applied to the mic audio. Adjustment range is between 1:1 and 16:1, or **Retain Source Settings**.

- **DE-ESSER RATIO:** Sets how aggressively the de-esser will remove sibilance from the mic audio. Adjustment range is between 1:1 and 8:1, or **Retain Source Settings**.

- **NOISE GATE DEPTH:** Sets the amount of noise reduction that will be applied to the mic audio if the noise gate is set to **Active** and the Noise Gate Threshold is reached. Adjustment range is between 0dB and -30 dB, or **Retain Source Settings**.

- **COMPRESSOR MODE:** Change the voice compressor between **Freeze** and **No Freeze** modes, or **Retain Source Settings**.

- **BACKWARD FEED DIM GAIN:** Set how much the program audio being fed back to this source will be dimmed when the source is being talked to using the **Talkback** feature. Adjustment range is 0dB to -30dB, or **Retain Existing Setting.**

- **CHANNEL ON/OFF STATUS:** Allows you to turn fader **On** or **Off** when the Show Profile is loaded.

- **CONTROL LOCK MAP:** Place a check mark in any of these boxes to keep the board op from changing the selections for this fader'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 **Lock**ed 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 **Unlock**ing immediately resets the gain to the fader's current position.

- **GROUP START:** Specify whether this fader is a **Master, Slave** or **Independent.** This setting is tied to those displayed in the Show Profile Group Start Screen; a change made here will apply on the Group Start Screen, and vice-versa. (For more information on the Group Start Function, please refer to the Group Start Screen description later in this chapter.

- **MOTOR FADER POSITION:** Motorized Element faders can be set to assume a predefined position when a Show Profile is loaded. Choose **Off** to disable, **Use** to specify a level to use (from -73 dB to +10 dB), or **Retain Existing Setting**.

©2008 Axia Audio — Rev. 3.0

Individual Headphones Section

> ***Note:*** These controls apply ONLY IF:
>
> • The loaded source is a Mic-type source.
>
> • The loaded source has an individual headphone feed associated with it.
>
> • The position at which the mic is installed also has an Element accessory panel that allows talent to select their own headphone feeds (such as the one shown in Figure 5-2).

- **CURRENT SOURCE:** The source currently specified for monitoring in the Monitors page of the Show Profile.

- **SOURCE FOR PRESET 1:** Specify the source that will load when the accessory panel's **Ext 1** key is pressed.

- **SOURCE FOR PRESET 2:** Specify the source that will load when the accessory panel's **Ext 2** key is pressed.

- **HEADPHONES MASTER GAIN:** Specify the initial amount of gain that the Individual Headphone Feed will have when the Show Profile loads. Turn then **Off** or **Use** up to 85 dB of cut.

Figure 5-2: an Element accessory panel with headphone selector

Finishing Up...

Finally, there are two more controls at the very bottom of the screen:

- **SAVE CHANGES** button applies any changes made to the current Channel Description. If you don't click this button, your changes won't be saved!

- **BACK TO SHOW PROFILE** link takes you back to the Show Profile Settings menu.

The Auxiliary Send And Return 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.**

©2008 Axia Audio — Rev. 3.0

- **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 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

- **TIMER MODE:** Lets you decide globally how the onscreen Count Up (event) timer will behave when this profile is loaded:
  - » Choose **Manual** to start, stop and reset the timer by hand using the controls displayed when the **Timer Options** key is pressed.
  - » Choose **Auto-Reset** and the timer will automatically reset to zero and begin counting whenever a fader is turned **On**.
  - » Choose **Auto-Add** to have the timer count up when a fader is turned **On** and stop counting when that source is turned **Off.** In this mode, the timer will not reset to zero when it is restarted.

- **SHOW TENTHS ON:** Allows you to set the Countdown and Count Up timers to show or hide tenths of a second displays. Choices are:
  - » **No Timers:** tenths of a second are hidden on both timers.
  - » **Down Timers:** tenths are shown only on Countdown timers.
  - » **Up Timers:** tenths are shown only on Count Up timers.
  - » **Both Timers:** All timers display tenths of a second readouts.

- **FADER MODE:** Sets the fader behavior globally for this Show Profile. Choose from:
  - » **Normal**: faders are turned on and off manually.
  - » **Fader Start**: faders turn on when the fader is raised and off when it is lowered to ∞.
  - » **Retain Source Setting**: Each individual fader behaves as its Source Profile dictates.

- **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 **PROGRAM** (PGM-4) bus, the source feeding the Control Room Monitors (**Moni-**

©2008 Axia Audio — Rev. 3.0

tor 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 EXTERNAL PREVIEW:** Element's External Preview allows other devices to route audio into the console's Preview system (speakers and operator headphones). Use this dropdown box to select the External Preview source for this Show Profile.

> ***What's External Preview for?*** The word "External" refers here to any source not generated by the studio you're working in. Let's say you have an intercom system that you want to feed the console's Preview channel. To do this, you would take the GPO from the intercom and use it to gate open the external preview input, which would be fed by the intercom audio. *Voila!*

- **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 EXTERNAL 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.

Sources for External 1 & 2 Section

- **SOURCE ID FOR EXTERNAL INPUT 1:** Use the dropdown box to specify a source for Monitor 1, External Input 1.

- **SOURCE ID FOR EXTERNAL INPUT 2:** Use the dropdown box to specify a source for Monitor 1, External Input 2.

Control Room Monitor (Monitor 1) Options

- **MONITOR ASSIGNMENT:** Choose to load **PGM1, PGM2, PGM3, PGM4, AUX A, AUX B, AUX C, AUX D, EXT1, EXT 2** or **No Source** if you wish the Profile to be loaded with nothing assigned to Monitor 1.

- **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 funtions. See Chapter Three, "GPIO Configuraton," for more information.

  This option should typically be programmed with the same logic channel number for all show profiles on a single Element, since this function is primarily used for "on-air" warning light control.

Control Room (Monitor 1) Headphone Options

- **MONITOR ASSIGNMENT:** Choose to load **PGM1, PGM2, PGM3, PGM4, AUX A, AUX B, AUX C, AUX D, EXT1, EXT2** or **No Source** if you wish the Profile to be loaded with nothing assigned to the board op's headphones.

- **CR HEADPHONES 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:** Set the audio feed to the Control Room headphones to **Stereo, Left, Right, Sum** or **Retain Existing Setting.**

©2008 Axia Audio — Rev. 3.0

- **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-HEADPHONES 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 (Monitor 2) 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 EXTERNAL INPUT:** Use this drop-down box to choose what source will be auditioned when **External** is chosen from the Studio 2 monitor 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 GHANNEL 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 funtions. See Chapter Three, "GPIO Configuraton," for more information.

  This option should typically be programmed with the same logic channel number for all show profiles on a single Element, since this function is primarily used for "on-air" warning light control.

Staying ahead of the game: You might not have a second studio controlled from your Element — but we still recommend that you program this option for each Show Profile. Why? The Studio Monitor logic has other functions, and works in conjunction with the Control Room Monitor logic. If you program it now, you won't have to go back and do it later.

©2008 Axia Audio — Rev. 3.0

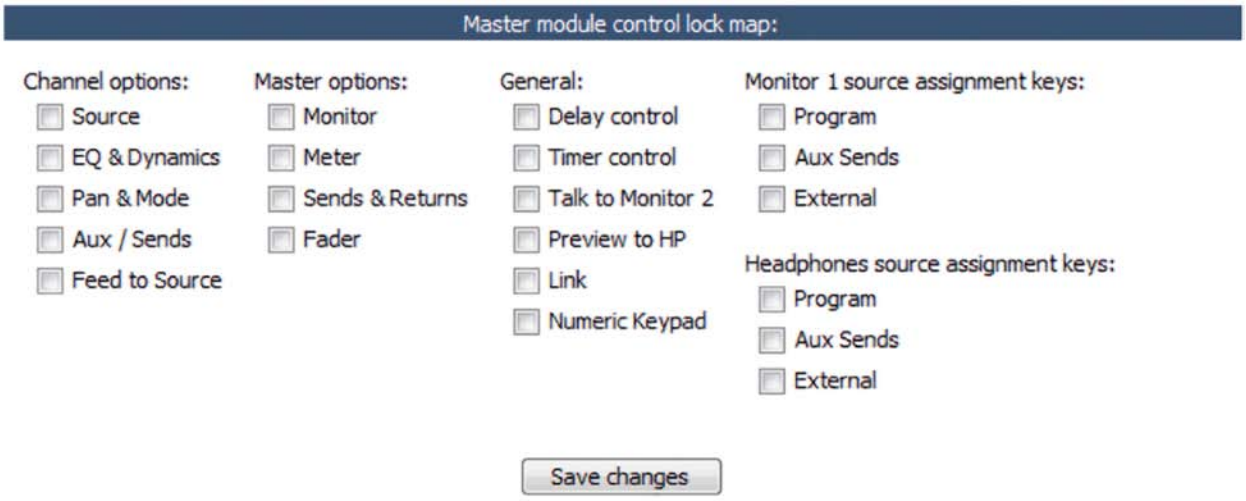


Figure 5-3: Master Module Control Lock Map. Keep those pesky jocks clamped down tight!

Master Module Control Lock Map

Element is a pretty powerful console — perhaps the most powerful and flexible your jocks have ever used. And that means that there are probably plenty of functions you'd rather they not have access to; things that could mess them up and result in the dreaded midnight phone call.

The Master Module Control Lock Map allows you to disable access to many Element functions to prevent settings from being tampered with. This can be done on a per-show basis, so remember that if you wish to disable access to a certain function globally, you must do so in *every* Show Profile.

The screenshot in Figure 5-3 shows the wide range of options you can lock down. This includes EQ and voice dynamics, mix-minus settings, even things like Monitor and Headphone assignments (for particularly clueless members of the airstaff).

Placing a checkmark in any of these boxes means that the operator will *not* be able to modify the settings for that item which you've pre-loaded in the Show Profile.

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 Screen

Element's Record Mode helps talent quickly prepare to record a phone bit, interview or other program segment for later air. Any source assigned to the **Program-4 bus** automatically feeds the **Record** and **Phone** buses as well. Sources assigned to **Program-4/Record** will follow the **Record** options in their Source Profiles; the CR Monitor assignment automatically changes to **PGM-4**, and the bus assignment keys for channels assigned to **Program-4** will flash when Record Mode is active..

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 **Recode Mode Engage** key. See Chapter Three, "Configuring GPIO," for a complete reference of Element GPIO functions.

©2008 Axia Audio — Rev. 3.0

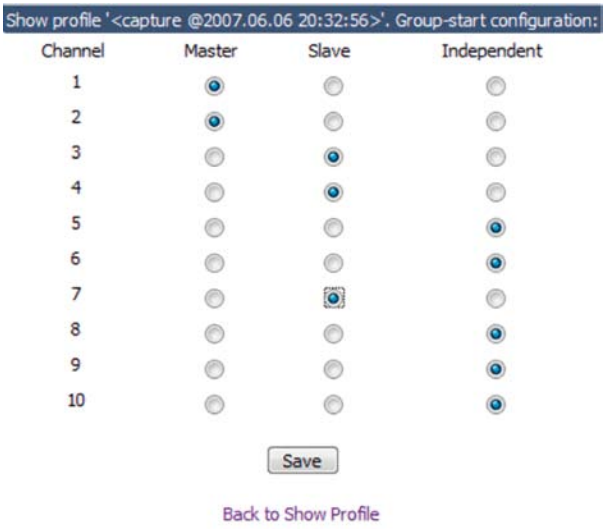


Figure 5-4: The Group Start assignment map

Group Start Screen

There may be times when you'd like to have a single keypress activate multiple faders for convenience' sake: interview shows with multiple guests, election coverage comprising multiple incoming remote lines, in-studio performances. Element's Group Start function lets you designate one or more faders as Master Faders, and one or more as Slave Faders; when a Master is turned **On** or **Off**, the Slaves turn **On** and **Off** as well.

Figure 5-4 illustrates how a Group Start map might look. The grid allows you to designate each fader on your console as either **Master, Slave** or **Independent** (normal operation). Note that you may have multiple **Master** faders; activating any of them causes all of the **Slave** channels to activate as well.

Note that making a change in this screen also changes the value of the Group Start setting in the fader's Channel Options screen (described previously in this chapter), and vice-versa.

Phone Screen

The Phone screen is used when you're setting up a Telos TWOx12, Series 2101 or Nx12 talkshow system for use with Element. The options in the screen are detailed fully in Appendix D: Working With Phone Hybrids; please refer to that chapter for details on setup. If you do not have a Phone Module installed, this screen will be empty.

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

Chapter 6 explores the use of Element's powerful Virtual Mixer capabilities, which allow you to create custom mixes of networked sources and use them like you would a single audio source. Grab a cold one and let's dive in.

©2008 Axia Audio — Rev. 3.0

Chapter 6:

Virtual Mixer

Constructing Virtual Mixes with VMIX

Virtual Mixer (or VMIX) is a cool feature present in every Element system. In addition to the mixing capabilities present on the Element console itself, there is a 40-input "virtual" mixer accessed via HTTP interface which consists of 40 stereo input channels, a direct output for each channel, 8 submixer outputs, and one master out. The 40 channels are divided equally among the 8 submixers providing 5 channels in each submixer.

The main mix output provides a mix of all active inputs, up to the maximum of 40. Each submixer input includes an on/off setting, a gain setting, and automatic fade-up/fade-down time parameters. The VMIXer works completely independently of the rest of the Element surface. In addition to controlling VMIX through its web page, Axia PathfinderPC software can also be used to dynamically control VMIX and create mixing functions based on a variety of system-wide parameters.

Notes for users upgrading from previous versions of VMIX

In the previous implementation of VMIX (prior to Element software v2.x), the StudioEngine view of VMIX was mostly status of the fields mentioned above, for the exception of enabling and assigning channel numbers to the vmix "outputs". All the other fields were controlled from the Elements screen.

While not the preferred approach with v2.x software, this previous method is still available for users that have become familiar with the earlier interface and wish to maintain the approach; simply log in to the Engine ID page of the StudioEngine and set the **Controlling V-Mixer from Element** option to **Enabled**. This allows Element to control VMIX sources.

Element control provides GPIO triggering of the VMIX, an option not yet available in the StudioEngine control pages. However, if you control StudioEngine VMIX using PathfinderPC, GPIO logic may be applied

Block diagrams for VMIX are provided at the end of this chapter.

FIgure 6-1: Engine VMIX screen

©2008 Axia Audio — Rev. 3.0

StudioEngine VMIX Controls

In an Axia system, all audio processing occurs in the StudioEngine — the Element console itself does no audio processing. That's why VMIX settings are found in the StudioEngine HTTP pages.

> You may have noted that the Element HTTP pages also have a VMIX section. This is the remnants of an earlier implementation of VMIX, whereing the control was shared between the Engine and the console. With Element 2.0 software, we have begun the process of transitioning to VMIX control on the StudioEngine side only, but controls are mirrored on the Element configuration pages to provide a smooth transition for users upgrading from earlier versions. This chapter may make reference to the earlier implementation for clarification reasons.

Even though the "V" in VMIX stands for "virtual", you can in fact perform a certain amount of "hands-on" mixing. Although you don't have physical control of the virtual faders or virtual **ON/OFF** switches, adjustments made to the VMIX take effect as soon as you apply them. While you wouldn't want to engineer an entire show this way, it's valuable to know that VMIX changes saved "on the fly" will affect your output streams immediately.

In the HTTP pages of the StudioEngine, there is a **Virtual Mixer** section and a **V-Mode** section (We'll get to the **V-Mode** portion later in this chapter). The portions of this page that display settings on a grey background are indicators only, not controls. Settings displayed on white backgrounds are controls, and can be adjusted.

At the top of the page, you can see the name of the Engine you are logged into. Below are eight submixers, each with five inputs. All eight submixes are summed into the VMIX Main. You're probably already thinking that there are a lot of things you can do with this setup!

Figure 6-1 shows a portion of the VMIX setup found in the Studio Engine web pages. It shows the first of the eight Submixes (VMIX 1). Let's look at the controls and see what they do.

VMIX Main is the sum of all eight submixes. Unless you really need a stream that combines the audio outputs of all the submixes, you may leave this disabled — it doesn't need to be **Enabled** for the submixers to work. In Figure 6-1, the channel is **Disabled** (default setting). If you need to use this mix, verify that the Channel number assigned to it is a unique channel number (just as you would in configuring the channel numbers assigned to Audio Node streams). When enabled, you can use this source anywhere in your network.

In most cases, the VMIX Subs are the only channels you will need to enable. Select the type of Livewire mode needed for the application (Live, Medium or Standard Stereo). Provide an appropriate unique channel number to generate and send a Multicast signal to the network. Only enable Submixes you intend to actively use; active submixes without any activity simply clutter up your network with empty streams!

At the bottom of each of the eight submix sections is an **Apply** button. Any changes you make will be saved when this button is pressed.

Other controls for each of the Sumbixer's five input channels include:

- **Src Name:** Enter a user friendly name for the source you want to assign to this input.
- **Channel:** Enter the unique channel number of your audio source.
- **In Stream Type:** In most cases this drop down will be set to **From Source.** If a source's Backfeed (unique feed or mix-minus audio) is desired as in input into the mixer, select **To Source.** (If this instruction is not clear, please review the chapter on backfeeds and mix-minus earlier in this manual to understand what these terms mean.)
- **Status/Audio:** These are status fields; **OK** is shown for valid streams and **ERR** for invalid streams (usually, an invalid stream error indicates a duplicate source - double-check your Channel Number entries). The **Audio** box provides a snapshot of the audio level with color indicator.
- **Enable:** Turns the input channel **ON** or **OFF;** the boxes just below provide an option to set the "fade up" and "fade down" times.
- **Fade Time:** These two fields contain time values that control the rate in which the audio goes from

©2008 Axia Audio — Rev. 3.0

OFF to ON and visa versa. If the source is off, gain is negative infinity (not adjustable, just a law of nature). When the source goes ON, the time in which the audio adjusts form OFF to ON (the gain value set in the first field) is the "Up time". It the field is set to 1 second, the audio will "fade up" to the gain value in 1 second. If the field is set to 0, the audio will have no fade period and simply just turn on at the gain value specified. The second field controls the ON to OFF fade time. This function can be used to create cross fades between sources when Pathfinder is dynamically making changes to the VMIX based on conditions met.

- **Gain:** You'll find a Master gain setting for the Submix (the mix of all five inputs) at the top of this column, followed by individual gain settings for each input that establish the gain for the input when it is in the **ON** state.
- **Out Name:** This column shows the name of the stream as it will appear when viewed in source assignment boxes elsewhere on the network.
- **Channel** *(the 2nd instance of it)***:** Assigns a channel number to the output of the source. Just as each audio source from an Audio Node and each mix and monitor feed from a StudioEngine must have a unique channel number, so do each of the VMIX sources that you want to make available to the network. In most cases only the Submix output itself needs a unique channel number, but if you so desire, each VMIX input can also be sent back to the network as a unique source, post the ON/OFF and gain stage of the VMIX. Just provide a unique channel number to each "VMIX Fader" you'd like to send out.
- **Out Stream Type:** Set the stream type you want to provide to the network for each VMIX output (Live Stream, Medium Stream, Standard Stream or Disabled) After you've given the Submix output a unique Channel number, enable it here. (Normally all of the VMIX "fader" channels remain "Disabled" unless you have other uses for them.)
- **Status/Audio:** An indicator field to tell you whether the stream is valid and audio is present. Note that if you have just enabled VMIX streams, you will need to click **Apply** to commit your changes. This will reload the browser page and refresh the Status fields.

Element VMIX Controls

Enabling Element control of VMIX

To switch control of VMIX from the StudioEngine to Element, open the StudioEngine HTTP pages and log in to the Engine ID page. set the **Controlling V-Mixer from Element** option to **Enabled**. This allows Element to control VMIX sources.

When Element has control of VMIX, the StudioEngine's presentation of the VMIX controls has most of its fields are grayed out, indicating that VMIX control has passed to the Element HTTP page.

Figure 6-2: Enabling Element control of VMIX

Fields that do not gray out are the Channel numbers and the Mode selection boxes that determine the type of audio streams to be sent to the network. The fields that are gray now function as indicators, showing what's been selected in the Element config pages; please refer to the descriptions of these fields in the previous section.

VMIX Control Description

To gain access to VMIX settings in the Element configuration Web pages, log into your Element and choose the Misc. menu item, then choose V-Mixer. This page shows eight Mixer settings — these settings are the control portion of the eight submixers mentioned previously.

Figure 6-3 shows controls for one of those eight Virtual Mixers. There are five **Faders**, which are the five inputs into the submixer, each with its own **ON/OFF** control, **Source ID** drop boxes, and **Gain**, **Up time** and **Down time** boxes.

Remember the Status indicator fields on the StudioEngine's VMIX page? When controlling VMIX with

©2008 Axia Audio — Rev. 3.0

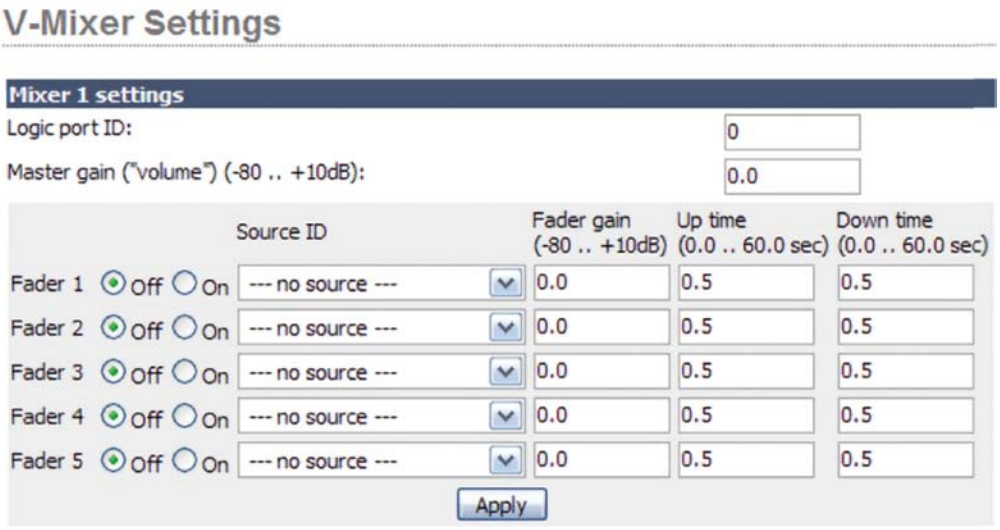


Figure 6-3: Element VMix settings page

Element, these StudioEngine indicators show what is assigned in the Element web page shown above:

- Engine's **Source Name** and **Channel Number** fields will display the **Source ID** you select here.
- Engine's **Enable** field will be on or off depending on whether you selected **OFF** or **ON** for any of the VMIX faders here.
- Engine's **Gain** field will display the number entered into the **Fader Gain** fields here.

About the Source Name drop down menu: How does a source show up in this drop down menu? It has to do with an option that appears whenever you create a Source Profile. Remember that there's a section of the Source Profile that lets you choose what fader channels the source may be assigned to, and whether it should be available to the CR monitors, Studio Monitor, or the … here's the kicker… *V-mixer input!* Gosh! Someone's been laying awake at night thinking of all this stuff.

The **Up Time** and **Down Time** fields contain values of time that control the rate at which the fader's audio goes from **OFF** to **ON** and visa versa. If the source is **OFF**, gain is negative infinity (not adjustable, just a law of nature).

When the source turns **ON**, the time in which the audio adjusts from **OFF** to **ON** (the gain value set in the earlier field) is the **Up Time.** It the field is set to 1 second, the audio will "fade up" to the gain value in 1 second. If the field is set to 0, the audio will not fade, but simply turn **ON** at the gain value specified. The second field controls the **ON** to **OFF** fade time.

This function can be used to create crossfades between sources when PathfinderPC is dynamically making changes to the VMIX based on conditions met.

Other parameters in the Mixer setting are the **Master Gain** of the mix and the **Logic Port ID.**

- The **Master Gain** field is a parameter that can be applied to the final mix of the five Submixer faders. Recall the indicator in the Engine HTTP screen for the Gain of **VMIX SUB?** That corresponds to this field. In most cases a gain of 0 is appropriate.

Note: When controlling VMIX through Element, you can set the gain of the eight Submixers, but you cannot set the gain of the **VMIX Main** output — that must be done from the StudioEngine VMIX page.

- The **Logic Port ID** for a particular VMIX Submixer must have a unique channel number, like any other system GPIO channel; this will provide GPIO functionality to the Submixer. We suggest you enter the same Channel number that you gave to the Submix itself.

"Why would a VMIX Submixer need a GPIO function?" Remember, the GPIO of your Element is grouped in fives: five **INPUTS** and five **OUTPUTS**. With this field assigned, a GPIO port or GPIO device can be used to receive status, or to command **ON/OFF** states of the five inputs in

©2008 Axia Audio — Rev. 3.0

the submix.

For example, if you assign the GPIO channel number to a GPIO port of the Element or a GPIO node, you will receive status when a source is on, and you can likewise provide a command to turn a source on. (Note that this is not a latching command).

Remember, the **Logic Port ID** is only functional if Element is controlling VMIX — if VMIX is being controlled by the StudioEngine, there is no Logic Port functionality.

» Finally, at the end of each Mixer section is an **Apply** button. Click this when you're done making changes to save the parameters you've set.

What can you do with a VMIX?

Now that you know how to set up and enable VMIX, what might you do with it? Here are some examples.

A mix of sources to monitor

Some facilities may need to monitor one or more sources in addition to program audio, like the "squawk channel" some satellite feed providers use to relay announcements. If you wanted to monitor this "squawk" audio on your Preview speaker without taking up a fader assignment, you could create a mix of the "squawk" source and the Preview mix from the engine. The Submix Out would be the audio source that you would route to the Preview speaker. *This example is possible with both StudioEngine and Element control of VMIX.*

Note: In this example, the "squawk" audio would not be muted if the CR Mic were turned on, as a normal Preview channel would be.

GPIO control of your VMIX

Imagine that you have a night jock that *should* monitor all four radio stations in your cluster. To help make sure this actually happens, you could send all four off-air signals as sources into a VMIX and take the output of its submix to a monitor. A User Module or external button wired to a GPIO port could provide a "press and hold" function to allow the jock to monitor the sources momentarily. *This example is only possible with Element Control of VMIX.*

PathfinderPC assisted VMIX

PathfinderPC server can be used to "control" VMIX. How? Pathfinder can receive commands from external devices like satellites, button panels, or automation systems and react to them by changing the state of VMIX **ON/OFF**, **Gain**, **Time Up**, and/or **Time Down** fields. This provides many different possibilities for facility automation, Intercom functions, or whatever else you might imagine. For example, the combo of PathfinderPC and VMIX could duplicate the function provided by other products that are controlling audio switching in many radio facilities. Refer to documentation on PathfinderPC for further information. *This example is possible in both Engine or Element control of VMIX.*

About PathfinderPC and VMIX: There are several ways that PathfinderPC can control VMIX. First, VMIX funxtions can be used both as qualifiers and actions in stack events. This means that a designer can select GPIO triggers, time based events, user button pushes, serial port commands, and other options and combinations of options to decide when to make changes to any fader in a VMIXER. The user can make a gain change based on these events, turn a channel off or on, and or adjust the fade times, giving complete control over the VMIXer based on any of the stacking events qualifiers.

Second, Pathfinder provides an ACU protocol transator which uses the VMIX system to mimic a Sine Systems ACU to a Prophet Automation System. In this way, a Prophet system can control the mixing capabilities of VMIX just as if they were ACU channels. In this scenario, each VMIX submixer becomes one 5 Channel ACU. GPIO hardware devices can also be mapped to the Channels within Pathfinder and it will allow those triggers to pass through to Prophet if the associated audio channels are turned on just like the ACU. The Pathfinder suite also includes a Bridge application that may be run on the Prophet system to convert the serial data to TCP, which can then be passed to multiple "clustered" Pathfinder servers for redundancy.

Finally, Pathfinder's Sofware Authority protocol translator includes commands to control any VMIXer present on Element consoles in the system; any machine that can send user defined serial or TCP commands can also control and read VMIX functions through Pathfinder.

Using these techniques VMIX can be used as a fully automated virtual mixer in the background of each Element. (See the PathfinderPC manual for the specific details on configuring Pathfinder with VMIX.)

©2008 Axia Audio — Rev. 3.0

Manipulating Streams With VMODE

New in Element 2.x, VMODE is an extension of the VMIX functionality. Its controls are found exclusively on the StudioEngine's **V-Mixer & V-Mode** HTTP page, and are not duplicated or accessible from the Element web pages.

What does VMODE do?

As you know, audio processing in the modern world requires DSP. The StudioEngine provides the DSP for Axia IP-Audio networks.

You've probably noticed that nearly all audio within an Axia network travels in stereo. Audio Node inputs (with the exception of Mic Nodes) are stereo, and so are Engine default sources. Most of the time, this is just what you need. But in some cases, an application surfaces that requires only one channel of an audio stream, or a mono sum. Until now, there was no easy way to make this happen.

Vmode is a new addition to the Element toolbox that addresses this requirement; its job is to help strip, sum, or compile a custom audio stream from available streams in the network.

VMODE Controls

VMode, as shown in Figure 6-4, has the capacity for up to eight separate stream-manipulation operations. The controls are as follows:

- **Input Selector** contains a drop down menu that allows you to select from a range of sources for each VMODE input. You may choose from **PGM-1/2/3/4,** the **Record** bus, **Phone** bus, **Aux Send A/B/C/D, VMIX SUB 1/2/3/4/5/6/7/8,** and **VMIX MAIN.** You can also choose **External,** which then allows you to select any network source by entering its channel number.

- **Src Name** and **Channel** are active when the **Input Selector** described above is set to **External**. You may type any conveniet title in the **Src Name** box to identify the source; the **Channel** column is where you would enter the channel number of the source you want to perform a VMODE operation upon.

- **Stream Type** follows the conventions found elsewhere in this manual.
  - » Select **From Source** if you're using a normal source generated by studio equipment connected to the Axia network.
  - » Select **To Source** if you plan to use the specified stream as a backfeed (mix-minus) to a phone, codec or IFB channel.
  - » **Surround** is only used if the console is in Surround (5.1+2) mixing mode.

- **Status** and **Audio** indicate the health of each particular channel.

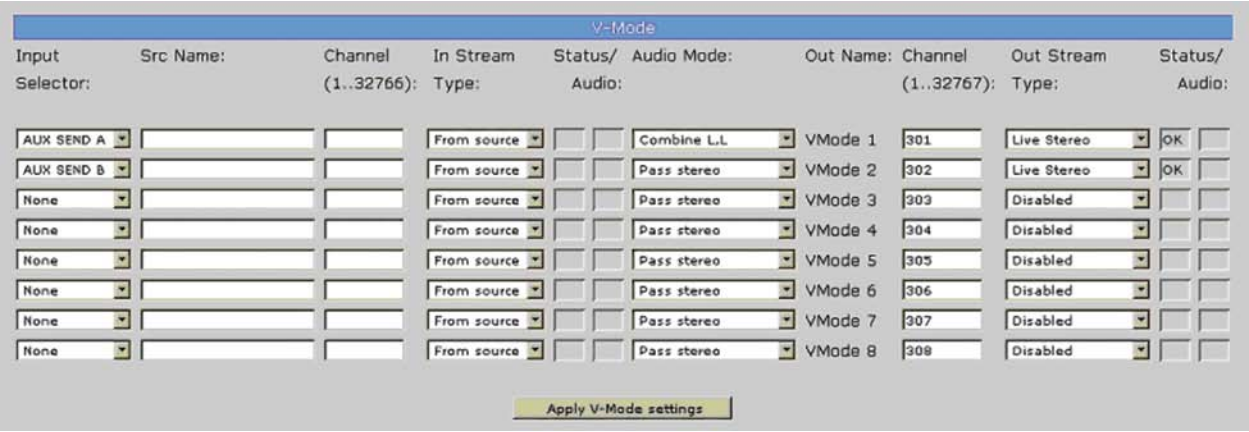


FIgure 6-4: Engine VMODE section

©2008 Axia Audio — Rev. 3.0

- » **Status** indicates **OK** to inform you that a stream is working normally.
- » **Audio** shows a colored tally to indicate audio presence.

- **Audio Mode** is the magic box that transforms your stereo stream into something else. You may choose from any of the following options:
  - » **Pass Stereo:** Passes the Left and Right channels as is, without modification.
  - » **Upmix from L:** In Surround mixing mode, takes Left channel and creates 5.1 upmix. In Stereo mixing mode, creates mono sum from Left channel.
  - » **Upmix from R:** In Surround mixing mode, takes Right channel and creates 5.1 upmix. In Stereo mixing mode, creates mono sum from Right channel.
  - » **Upmix from L+R:** In Surround mixing mode, sums Left and Right channels and creates 5.1 upmix. In Stereo mixing mode, creates mono sum from both channels.
  - » **Upmix from stereo:** In Surround mixing mode, creates 5.1 upmix from stereo input. In Stereo mixing mode, passes Left and Right channels without modification.
  - » **Downmix from 5.1:** In Surround mixing mode, takes 5.1 Surround source and creates stereo downmix. In Stereo mixing mode, muts audio (do not use).
  - » **Pass all:** In Surround mixing mode, passes 5.1 Surround input without modification. In Stereo mixing mode, passes Left and Right channels without modification.
  - » **Split L -6:** Stereo mixing mode only. Sums Left and Right channels, reduces gain of the sum by 6dB and places mix on Left output channel.
  - » **Split R -6:** Stereo mixing mode only. Sums Left and Right channels, reduces gain of the sum by 6dB and places mix on Right output channel.
  - » **Split L -3:** Stereo mixing mode only. Sums Left and Right channels, reduces gain of the sum by 3dB and places mix on Left output channel.
  - » **Split R -3:** Stereo mixing mode only. Sums Left and Right channels, reduces gain of the sum by 3dB and places mix on Right output channel.

The **Combine** options presented next are designed to work with a pair of inputs at a time. There are eight VMODE inputs (4 pairs). If you choose Combine L,L for the Audio Mode of the first input, the Left channel from that input (row 1) will be passed to the Left channel of the output, and the Left channel of the second input (row 2) will be passed to the Right channel of the output.

In Figure 6-4, Output channel 301 will have the Left channel from Aux Send A on the Left and Left channel from Aux Send B on the Right using the **Combine L,L** Audio Mode.

  - » **Combine L,L:** Left channel of first VMIX input is sent to Left output channel, and Left channel of second VMIX input is sent to to Right output channel.
  - » **Combine L,R:** Left channel of first VMIX input is sent to Left output channel, and Right channel of second VMIX input is sent to to Right output channel.
  - » **Combine R,L:** Right channel of first VMIX input is sent to Left output channel, and Left channel of second VMIX input is sent to to Right output channel.
  - » **Combine R,R:** Right channel of first VMIX input is sent to Left output channel, and Right channel of second VMIX input is sent to to Right output channel.
  - » **Combine LR, 5.1:** Used in Surround mode to construct an eight-channel (5.1+Stereo) combined output. Takes Left and Right from first VMIX input and combines it with the 5.1 audio from the second VMIX input.
  - » **Combine 5.1, LR:** Same as above, but takes 5.1 audio from the first VMIX input and combines it with Left and Right from second VMIX input to create 5.1+Stereo combined output.

- **Out Name** shows the name of the VMODE output stream as it will appear on the Axia network. This cannot be changed.

- **Channel** is the unique channel given to the stream so that it can be identified and used in the network.

©2008 Axia Audio — Rev. 3.0

Be certain to replace the default channel numbers to avoid confusion.

- **Out Stream Type: Choose from Live Stereo, Medium Stereo, Standard Stereo, Surround** and **Disabled.** Only enable VMODE streams you intend to actively use; active streams without any activity simply clutter up your network with empty streams!

- **Status/Audio:** Indicate stream health and audio presence as explained previously.

What can you do with VMODE?

Now that you know how to set up and enable VMIX, Here are a few examples of how you might use it.

Create a mono stream.

Imagine you've got a satellite receiver connected to an Audio Node. Sometimes, the program content is received on only the Left channel — there is other content on the Right channel that you don't want to air! Using VMODE, you can split the L and R channels and create a new source using just the channel you want, like so:

1. Choose **External** in the **Input Selector** box.
2. Type the receiver's name in the **Src Name** box.
3. Enter the unique channel number of the receiver in the **Channel** box.
4. Leave **In Stream Type** set as **From Source.**
5. Choose **Upmix from L** in the **Audio Mode** box.
6. Enter a unique channel number in the Output **Channel** box.
7. Choose a stream type from the **Out Stream Type** box (you can choose **Standard** because this is not live audio).
8. Click the **Apply V-Mode Settings** button.

You've just created a mono source with only the content you want that can be routed wherever you like.

Create a Split Record feed with Gain Control

Element v2.x introduces a "split" record function that can be enabled in any devices' Source Profile; this allows you to send a sources' audio to one channel of the stereo **Record** bus. You'd like to use this split-recording for an interview with a host several guests, placing the host on the Left channel and all the other sources (guest mics, phone callers, etc.) on the Right channel. Plus, each source needs to have an independent gain stage separate from the on-air (**PGM-1**) mix. Complex yes; but with VMODE you can do it.

1. Using Element console controls, place the Host's mic into the **Aux Send A** bus, pre-fader (**ON PRE**).
2. Place each additional source to be recorded on the right channel into the **Aux Send B** bus, pre-fader.
3. Make certain that the **Aux Send A** and **Aux Send B** buses are **ON** and at a suitable gain level.
4. Now, within VMODE, choose an input pair and set their **Input Selector** boxes to **Aux Send A** and **Aux Send B**.
5. Set the **Audio Mode** of the first of the two **VMODE** inputs rows to **Combine L,L**.
6. Create a unique channel number and stream type for your new VMODE stream. This unique channel will contain the Left channel of **Aux Send A** on its Left output channel, and the Left channel of **Aux Send B** on its Right output channel.
7. Load your new VMODE stream to an Element fader and assign it to **PGM-4/Record**. Voila!

What's Next

You've learned about nearly all of Element's functions, learned how to set up Show and Source Profiles, configure Virtual Mixers, and more. Now, if you're ready to explore the final frontier, join us in Chapters 7 & 8 for a look at Element and StudioEngine advanced settings.

©2008 Axia Audio — Rev. 3.0

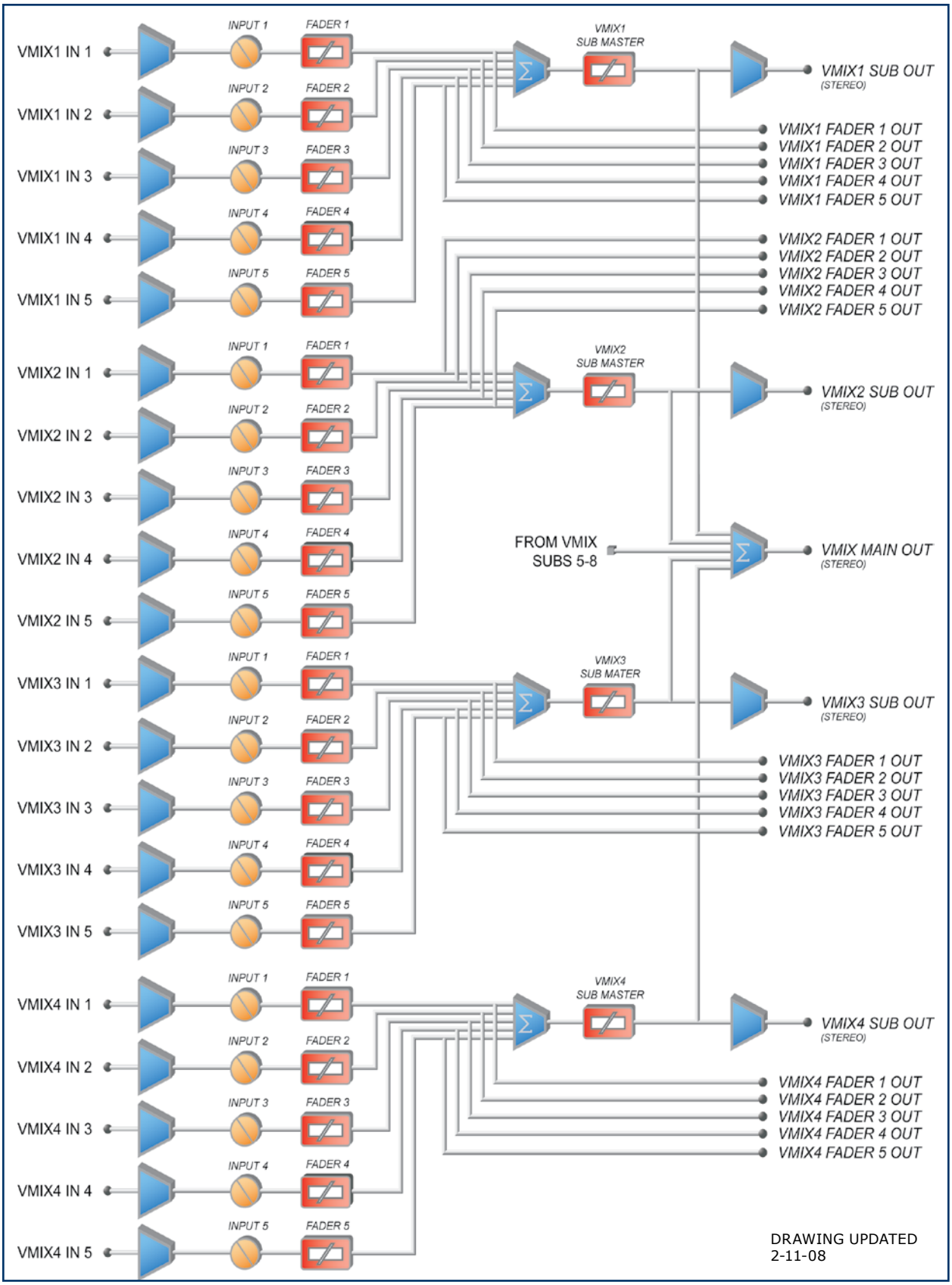


Figure 6-5: VMIX Channel Block, Part 1

©2008 Axia Audio — Rev. 3.0

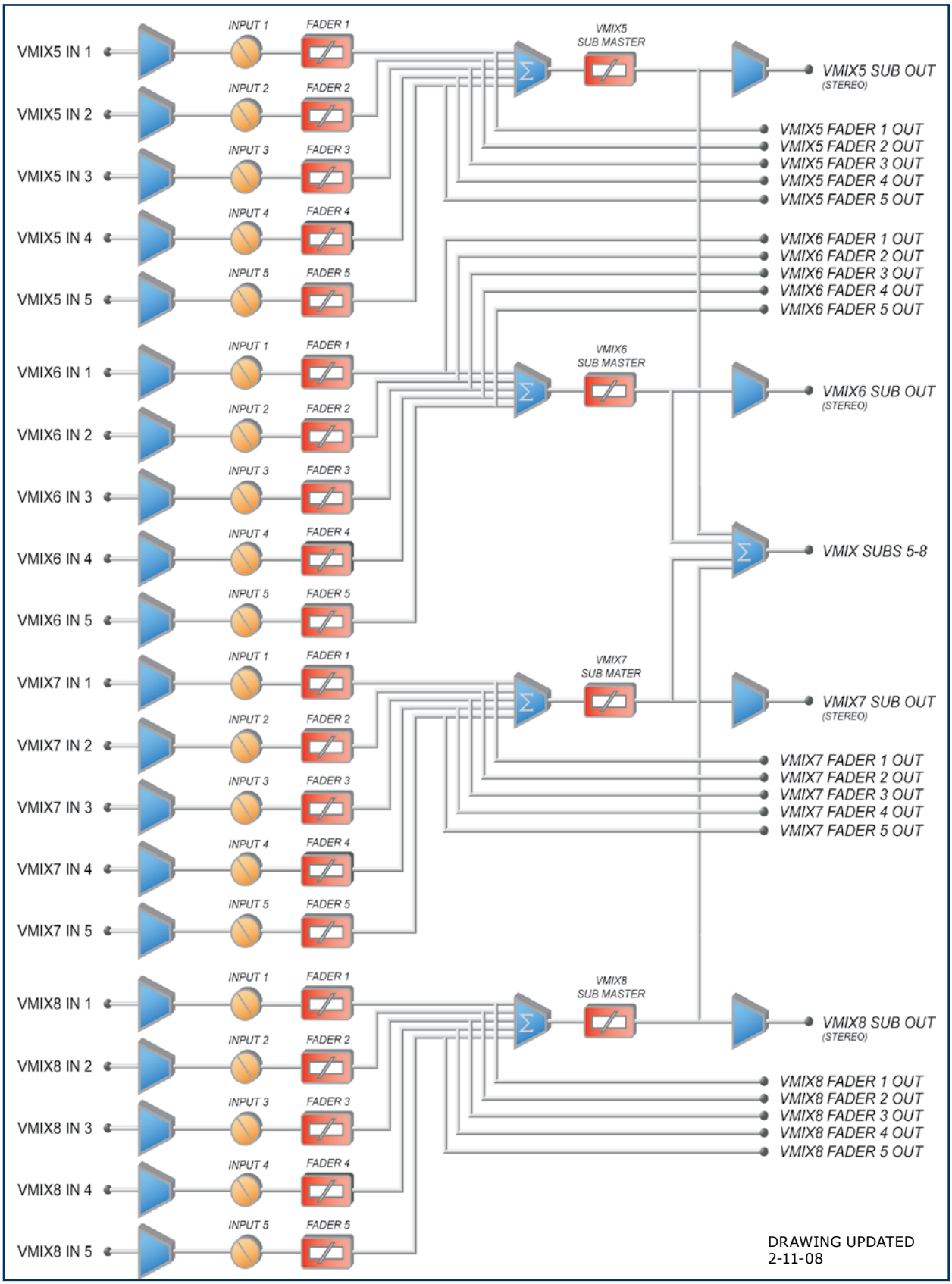
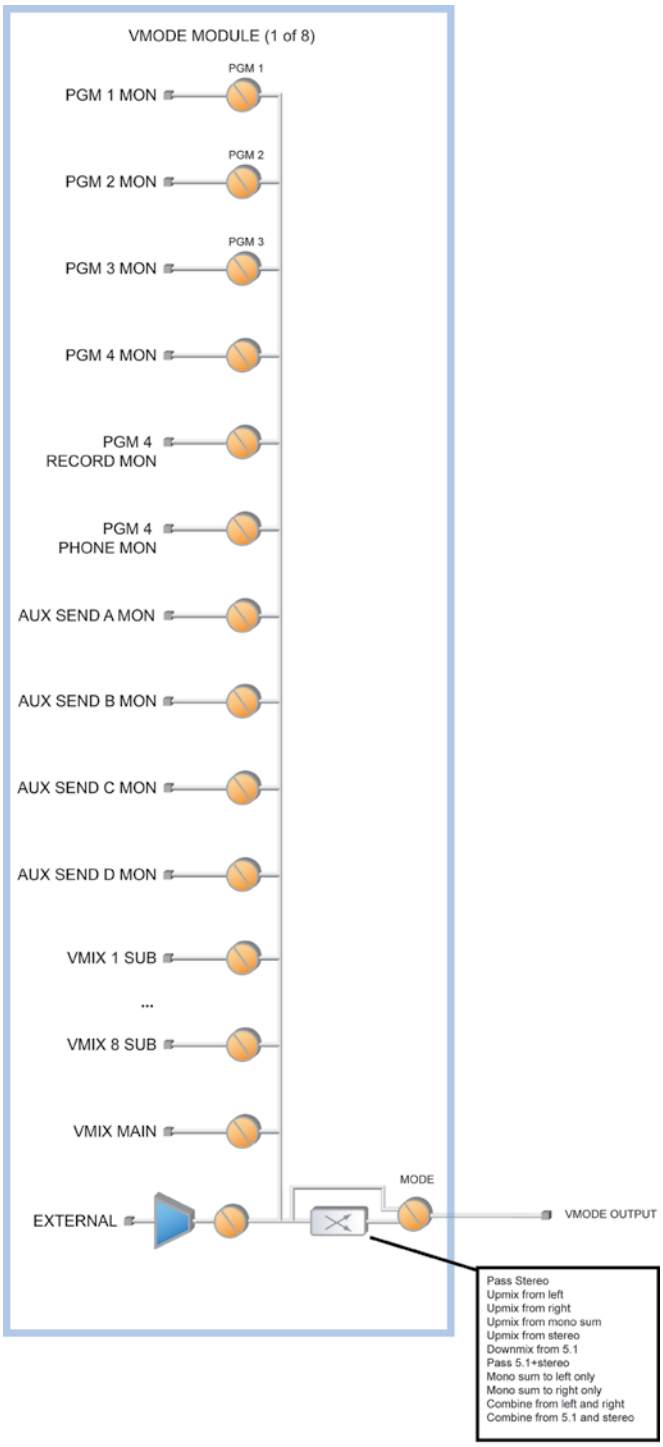


Figure 6-6: VMIX Channel Block, Part 2

©2008 Axia Audio — Rev. 3.0



DRAWING UPDATED
2-11-08

Figure 6-7: VMODE Channel Nlock

©2008 Axia Audio — Rev. 3.0

The cup steams, sweet scent

wafting lightly in the air.

Mmmmm... it's Earl Grey.

©2008 Axia Audio — Rev. 3.0

Chapter 7:
Advanced Element Controls

In Chapter 1, we covered the basics of setting up your Element and StudioEngine, in the interest of getting you up and running quickly. Now that you've read this far, you should consider yourself an advanced user! In keeping with your dawning guru status, it's time for a thorough look at all of the controls found in the Element configuration Web pages.

Element Configuration Screens

The Element Control Center

This is the first screen you see when you log into your Element console from your Web browser. The main menu runs down the left side and gives access to all of Element's options and settings. In the main window you can see at a glance what version of Element software is loaded, system information such as console uptime, network link integrity and CPU stats, and more. This information screen is accessible without a password; clicking on any menu heading will prompt you for a password.

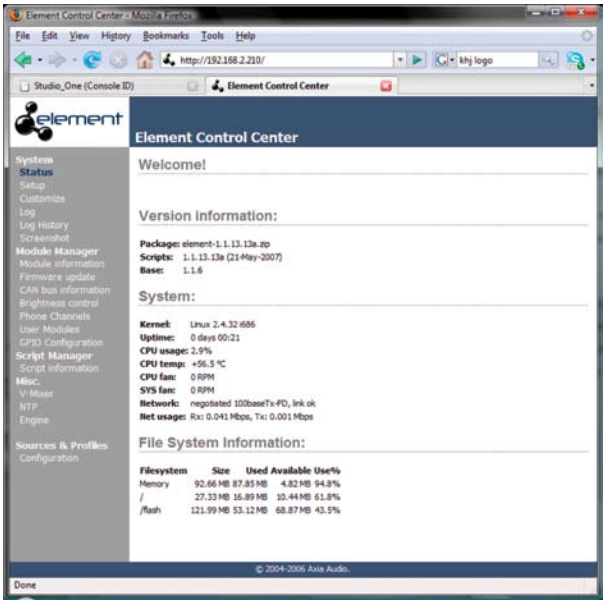


Fig. 7-1: The Element Control Center

The Setup Menu

We touched briefly on the Setup screen in Chapter 1. Now let's look at the rest of the tools on this page.

- **Network Configuration** contains the fields for the Element's IP Address and other network functions.
  - » **Hostname** is the unique name you give the console to identify it in the Web interface. You might want to name it after the studio it's in.
  - » **IP Address:** This Element's unique network address.
  - » **Netmask:** Enter your network's submask here.
  - » **Gateway:** If you have a network gateway to allow access from outside your facility, enter it here.
  - » **DNS-0 & DNS-1:** If your Axia network is connected to the "outside workd", you can specify your ISP's DNS servers in these fields.

  Why would you want to do this? Well, you could

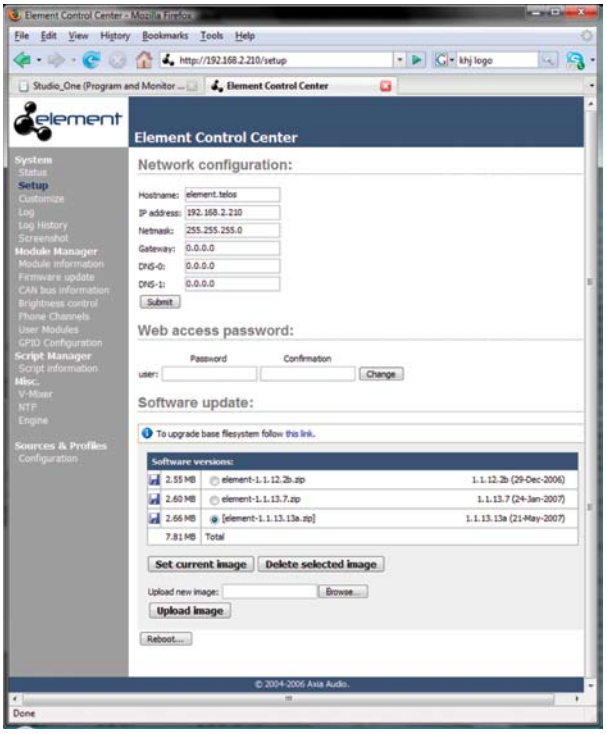


Fig. 7-2: Setup screen

©2008 Axia Audio — Rev. 3.0

simplify NTP access by entering the server's name, `tmc.edu` instead of `128.249.1.1`. Or, if you set up a DNS server in-house, you could give real names to your StudioEngines and Phone systems, then use those names instead of IP addresses in Element configuration pages.

- **Web Access Password:** By default, the username is set to `user` and the password is blank. If you want to add a password, type it in and click **Change.** Note that passwords will not display here once changed, so be sure to make a note of it.
- **Software Update** is where you upload new Element operating software.
  - » **Base File System Upgrade:** Most Element software updates do not include a Base File update, but if one is required, click this link to upload and apply the new code.
  - » **Software Versions** shows you the software packages currently stored on your Element's flash memory card. The blue radio button is displayed next to the active software image.
  - » **Set Current Image:** When you want to change the active software package, click on the radio button next to the image you wish to activate, and click this button. A reboot is necessary to load the new software choice.
  - » **Delete Current Image:** Click the radio button next to software you wish to delete and click this button. Note that you cannot delete the active software image.

Good Housekeeping: It's best not to keep too many old software images around. They clutter up Element's flash memory, leaving less space for Show Profiles, syslogs, and other important data. We recommend that you keep only one previous software package stored in Element.

  - » **Upload New Image:** From time to time, Axia makes new software packages available for download from www.AxiaAudio.com. Use the **Browse** and **Upload Image** buttons to install new software, as described in the Technical Bulletin that accompanies each software release.
  - » **Reboot** shuts down Element and restarts it. (Remember that mixing takes place in the StudioEngine, so this can be done even while you are on the air. Interruptions of your program stream, if any, will last less than 1 second.) A reboot is required after changing active software.

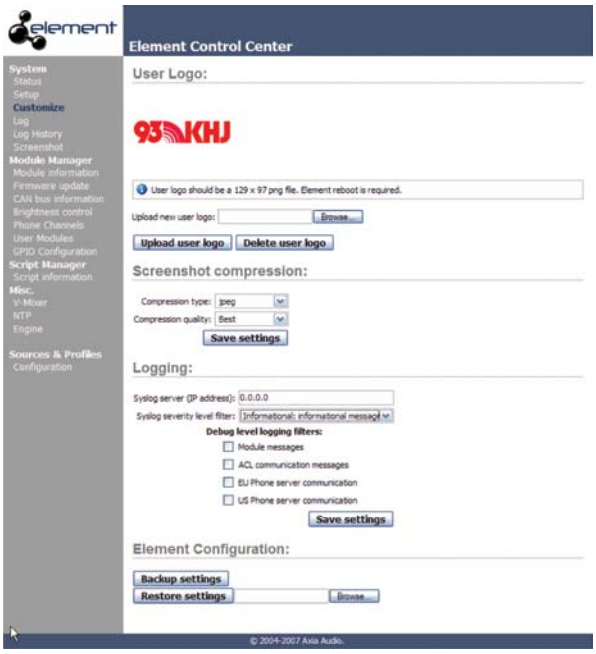


Fig. 7-3: Customize screen

The Customize Menu

- **User Logo:** This utility allows you to place your station's logo on the Element operator display. To do so, create a PNG file in Photoshop or any similar image editor. This file must measure exactly 129 x 97 pixels. Use the **Browse** and **Upload User Logo** buttons to install your logo file. Upon reboot, this logo will appear on the Operator's display. To delete a displayed logo, click the **Delete User Logo** button and reboot.
- **Screenshot Compression:** You can save a shot of the Operator's display for documentation, troubleshooting or training purposes. You may choose to save your screenshot in JPEG or PNG format, with three levels of compression quality using the **Compression Type** and **Compression Quality** drop-down boxes. Click **Save Settings** when you are done.
- **Logging:** New in v2.0 Element software, you can now save detailed syslogs. These options let you

©2008 Axia Audio — Rev. 3.0

specify the IP Address of your **Syslog Server,** set the **Syslog Severity Level** to log only the level of messages you desire, and set various other logging options. Click **Save Settings** when you're done.

- **Element Configuration:** This section lets you **Backup** and **Restore** all Element configuration settings. Backup file format is standard XML.

Fig. 7-4: Screenshot

The Screenshot Menu

Choosing this menu item immediately captures a shot of the Element's Operator screen. This, in combination with the activity log, can be used to determine just what the operator is seeing — useful for at-home troubleshooting during those 6AM Sunday phone calls. Pressing your browser's **Back** button will return you to the Element menu.

Save your work! We recommend backing up your Configuration files whenever a major change is made, or is about to be made, to your Element. Whether you are reconfiguring Show or Source profiles *en masse* or about to perform a software update, it's always a good idea to have a copy of your important data.

The Log and Log History Menu

Element keeps a detailed log of each button press, fader movement and other system events. Choosing the **Log** menu item lets you examine the last 100 logged activities; choosing the **Log History** menu item allows you to view an entire day's log at one time and delete old log files if you choose. The oldest log files are deleted automatically as storage space dictates.

The Module Information Menu

The first selection under the **Module Manager** heading displays a detailed list of the installed modules in your Element console. Some of these fields are for diagnostics; below is a list of those you might find useful:

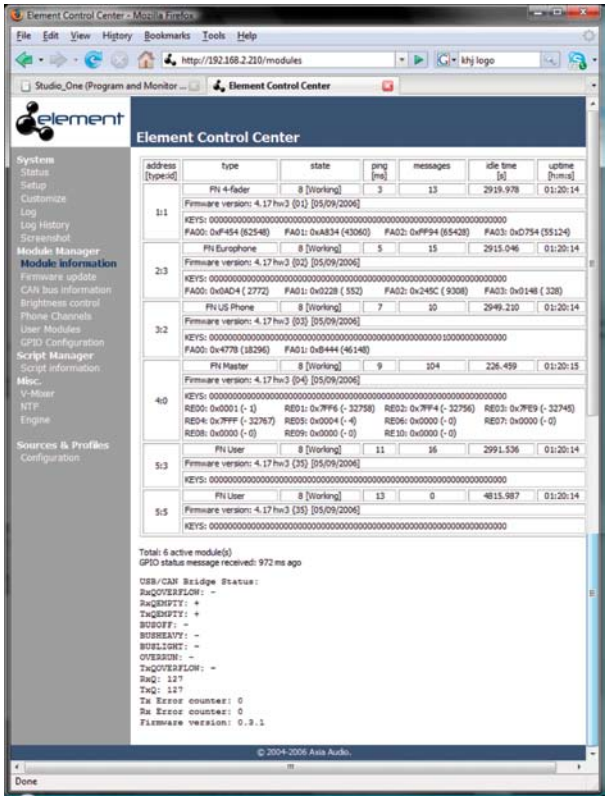


Fig. 7-5: Module Information screen. Can you spot the Module ID conflict?

- The **Address** field displays the number Axia has assigned to each fader type, followed by the setting of its module rotary ID selector switch. This is useful in determining if any modules' ID selectors are set to duplicate values (see Appendix C for information on correctly setting module ID selectors).
- The **Type** field gives you the actual name of each module (i.e., 4-Fader, Master, US Phone, etc.).
- The **Status** field tells you whether or not the module is working, i.e., communicating with the Element CPU.

©2008 Axia Audio — Rev. 3.0

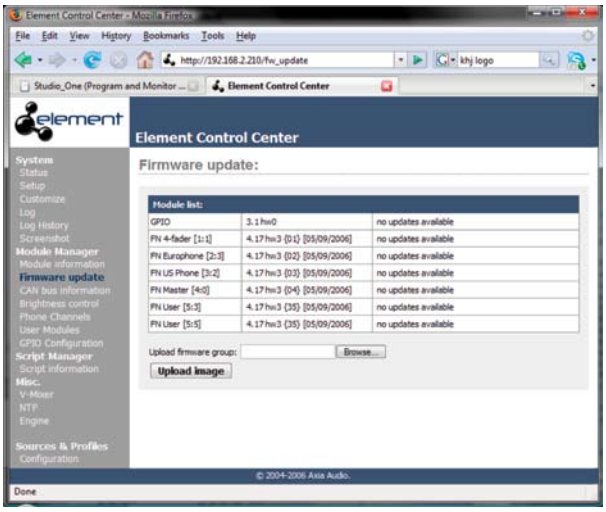


Fig. 7-6: Module Update screen

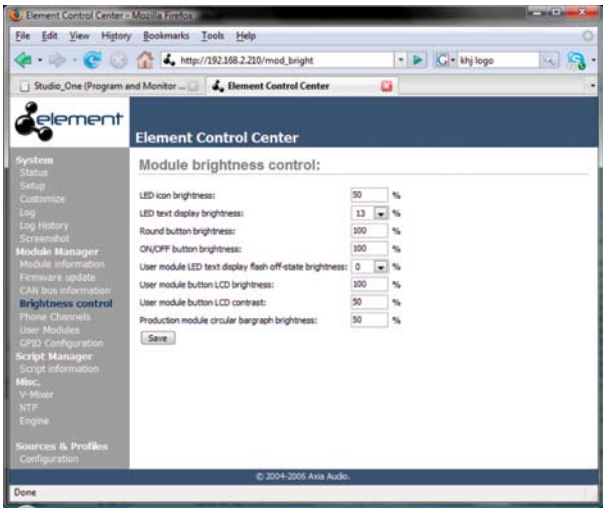


Fig. 7-7: Brightness Control screen

- The **Firmware Version** field displays the fader firmware version that's loaded on each module.

At the bottom of the page, you'll see a lot of information pertaining to the health of the Element's CANBus communication network. This is for diagnostic purposes only.

The Firmware Update Menu

From time to time, new module firmware may become available from Axia. You may confirm the current Module Firmware versions installed in your Element and install new firmware by choosing this menu item.

To update Module Firmware, use the **Browse** and **Upload Image** buttons to install the module firmware package obtained from Axia Support. Once a new software image has been uploaded, the Firmware Update screen will change to notify you which modules have new firmware available.

Each module must be updated individually. To update a module, simply click on its **Update** link.

Note that updating a module takes it offline during the update process. While audio playing through the module's faders will continue uninterrupted, keep in mind that while updating (usually about 10-15 seconds) you will not be able to adjust gain or turn faders on or off on the module being updated.

CAN Bus Information Manu

This screen shows statistics about CANBus communications between Element's CPU and control surface.

Brightness Control Menu

This menu offers several useful options for fine-tuning your Element's various LED and LCD displays to your studio's ambient lighting. All values are expressed in percent.

- **LED Icon Brightness** lets you adjust the luminance of the Status Symbols / fader numbers on the Element overbridge (the little windows just above the source identifiers).
- **LED Text Display Brightness** does the same for the alphanumic displays that indicate what source is loaded to each fader channel.
- **Round Button Brightness** adjusts the luminance of the interior LEDs that illuminate Element's **Options**, **PGM 1-4**, **Monitor**, **Headphone** and other selection keys.
- **ON / OFF Button Brightness:** You can probably figure this one out on your own :^)
- **User Module LED Text Display Flash Off-State Brightness:** If you have any SmartSwitch modules or accessory panels installed, you can use this set-

©2008 Axia Audio — Rev. 3.0

ting to determine how "dark" display text is when that text flashes for attention. A value of 0 makes the text flash from completely on to completely off; a higher value keeps the text slightly illuminated in its "off" state.

- **User Module Button LCD Brightness:** This affects only SmartSwitch modules and panels. Determines how bright the backlighting is for each SmartSwitch button's colored LCD screen.
- **User Module Button LCD Contrast:** Adjusts the contrast between the text and background areas of SmartSwitch button displays.
- **Production Module Circular Bargraph Brightness:** Determines the luminance of the lit segments that surround the adjustment knobs on the optional Element Production Module.

Phone Channels Menu

This screen allows you to set up Element for use with Telos Series 2101, TWOx12 and Nx12 broadcast phone systems. For more details, please refer to Appendix D: Working With Phone Hybrids.

User Modules Menu

This menu allows you to set up and use Axia accessory SmartSwitch and Film-Cap Button modules and accessory panels with your Element console. Please refer to Appendix E: Accessory User Panels.

GPIO Configuration Menu

This menu is where you configure the assignments for the eight opto-isolated GPIO connections found on the Element CPU connection panel. For a thorough discussion of this screen and how to configure GPIO, please refer to Chapter 3: Configuring GPIO.

Script Information Menu

Element consoles use standard Javascript code for some routine functions. This screen is for diagnostics and custom programming only and should not be used without the express instruction of Axia Support.

V-Mixer Menu

Element allows you to mix up to 40 sources using a built-in Virtual Mixer, consisting of eight submixers controlling five inputs each. Setup and use is described in Chapter 6: Virtual Mixer.

NTP Menu

Each individual Element in your network can use either its own internal clock or an external time server for timekeeping. To set up an NTP server:

1. Choose the **NTP ON** radio button.
2. Enter the address of an NTP server in the Primary box, and a Secondary server if available.
3. Set your Time Zone offset using the dropdown box.
4. Click **Save**.

To disable NTP service, simply choose **NTP OFF** and click **Save**.

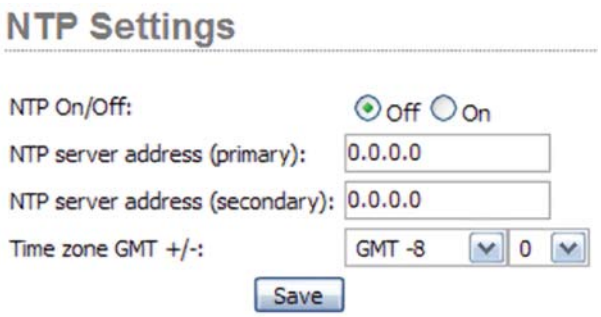


Fig. 7-8: NTP Menu controls

Engine Menu

Alternate location to enter the IP Address of the StudioEngine your Element is paired with (typically this is entered using the **IP Address Book** screen). Click **Save** if you've made changes..

Sources & Profiles Configuration Menu

Choosing this menu item takes you to the Profiles submenus.

©2008 Axia Audio — Rev. 3.0

Show Profiles and Source Profiles Menus

These menus allow you to construct and administer Show and Source Profiles specific to this Element console. Refer to Chapter 2: Configuring Inputs and Chapter 5: Show Profiles for details on using these controls.

Fig. 7-9: Profiles Maintenance screen

Legacy Configuration Menu

This screen enables you to restore any backups you may have of Element Configuration files made with software prior to Version 2.x. Use the **Browse** and **Restore** buttons to perform the desired actions.

©2008 Axia Audio — Rev. 3.0

Chapter 8:

StudioEngine Controls

In this chapter, we'll look in-depth at all of the controls and options found in the StudioEngine web pages.

StudioEngine Configuration Screens

The Main Menu

Fig. 8-1: StudioEngine Main menu

This is the "jumping-off" point for all of the StudioEngine settings pages, the page you'll see after entering your StudioEngine's IP address into your Web browser. Upon making any menu choice, you'll be asked for a password. The default factory login is `user`; leave the password field empty.

Some of the menu items presented here – such as **Fader Channels** and **Stream Statistics** – are purely informational in nature. Others – **Engine ID, System** – contain options and settings that are used for setup of your StudioEngine.

The Fader Channels Menu

Settings shown here are not adjustable. Useful for remote assistance or diagnostics, this page can tell you what sources are loaded to each fader of the Engine's associated control surface, along with the headphone monitor source and backfeed (mix-minus), if any, is being fed to each loaded source.

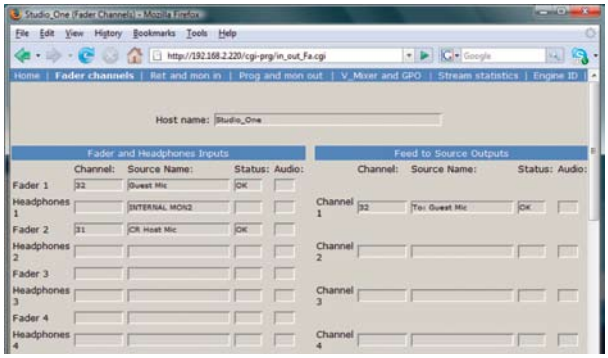


Fig. 8-2: Fader Channels screen

The Return & Monitor Inputs Menu

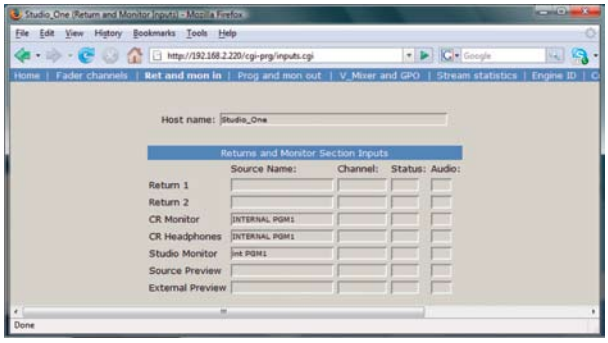


Fig. 8-3: Return & Monitor Inputs menu

Another informational screen, choosing this menu option shows you what audio sources are currently assigned to your console's:

- Auxiliary returns,
- Control Room monitor and heaphone channels,
- Studio Monitor channel,
- Source Preview (Cue) channel,
- External Preview channel.

The "Host Name" field at the top of the page shows the name you've defined for this Engine for confirmation should you be viewing the page remotely.

©2008 Axia Audio — Rev. 3.0

The Program & Monitor Outputs Menu

This is where you enable and assign unique channel numbers to all of your console's outputs. If any of your Element's outputs or monitor feeds are disabled in this menu, that channel will have no audio output.

The outputs you control are:

- **Program 1 - 4:** Your console's main Program bus outputs.
- **Program 4 Record:** Special output for recording devices which feeds the contents the Program 4 bus pre-fader and pre-On/Off switch.
- **Aux Send 1 - 4:** Your console's stereo Auxiliary Send buses.
- **CR Monitor Direct:** This is the output of the monitor selector *before* the operator's volume controls and mutes; useful for feeding a Producer's position, or any other monitoring station with independent headphone volume controls.
- **CR Monitor:** Contains whatever audio source your operator has selected to feed the Control Room monitors.
- **CR Headphones:** Contains whatever audio source your operator has selected to feed the Control Room headphones.
- **Preview:** The feed to the Preview (Cue) speaker.
- **Talk to CR:** This channel feeds audio to the Control Room talkback channel whenever the **Talk** button is pressed on any Studio Accessory panel.
- **Guest Headphones:** Contains whatever audio source has been selected to feed the Guest headphones.
- **Studio Monitor:** Contains whatever audio source your operator has selected to feed the Studio monitors.
- **Talent Headphones:** Contains whatever audio source has been selected to feed the Talent headphones.
- **Talkback:** When any console **Talk** key is pressed, CR Mic audio is routed, pre-fader, to this channel.

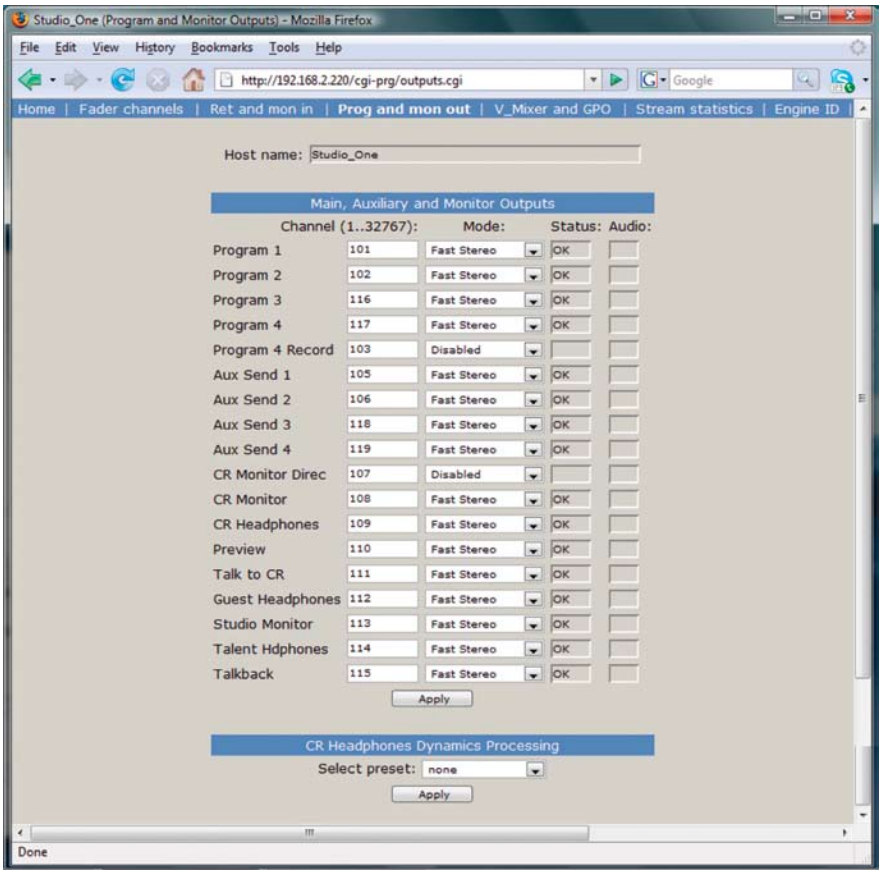


Fig. 8-4: Program & Monitor Outputs menu

The columns following each of these entries display the status of each channel and allow you to make changes.

The **Channel** column, like other similar fields used throughout the Axia system, is where you enter a unique channel number for each console output.

The **Mode** column lets you enable and disable each output, and set the priority of each stream.

- **Fast Stereo** are for live audio where lowest latency is required, like Program outputs and mic-to-headphone paths.
- **Medium Stereo** is the recommended default setting for everything else, and balances low latency with high compatibility for systems incorporating PCs.

©2008 Axia Audio — Rev. 3.0

- **Standard Stereo** is for low-priority, non-realtime audio. Standard streams have higher latency, but offer the best compatiblity for systems with older PCs.
- **Surround** should be chosen if you are going to be mixing in 5.1 audio.
- **Disabled** turns off outputs you don't intend to use.
- **CR Dynamics Headphone Processing:** Element contains special audio processing presets by Omnia that you can apply to the Control Room Headphone channel. This is provided so that you can simulate the "air sound" provided by an on-air processor, so that your jocks can have real-time Program monitoring that sounds like "air" without having to set up a separate processor chain. Presets are provided for several formats, as well as reduced-bit rate audio streams. To apply, choose one from the dropdown box and choose **Apply.**

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.

The VMIX & VMODE Menu

This menu allows you to view settings and make adjustments to the Element Virtual Mixer system. Please see Chapter 6, Virtual Mixer, for details on this page.

Stream Statistics Menu

This screen provides information about all active audio streams in and out of the StudioEngine. This is useful primarily for diagnostics and contains no settings that can be modified.

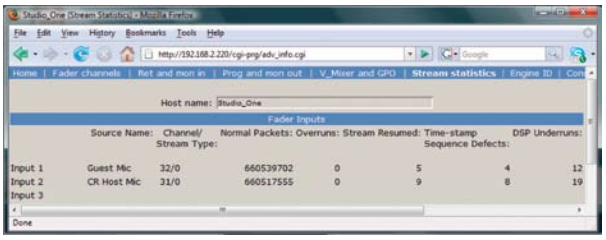


Fig. 7-14: Stream Statistics menu

The Engine ID Menu

IP Settings Section

- **Host Name:** Enter a unique name for this particular StudioEngine. This name is displayed on various information and diagnostic screens.
- **Network Address:** This field contains the IP address you've assigned for this StudioEngine.
- **Netmask:** The submask for your IP-Audio Network.
- **Gateway:** If your Axia network has a network gateway to allow outside access, enter its IP address here.
- **Syslog Server (IP Address):** Your StudioEngine can generate detailed logs of system activity. If you choose to keep syslogs, enter the IP address of the

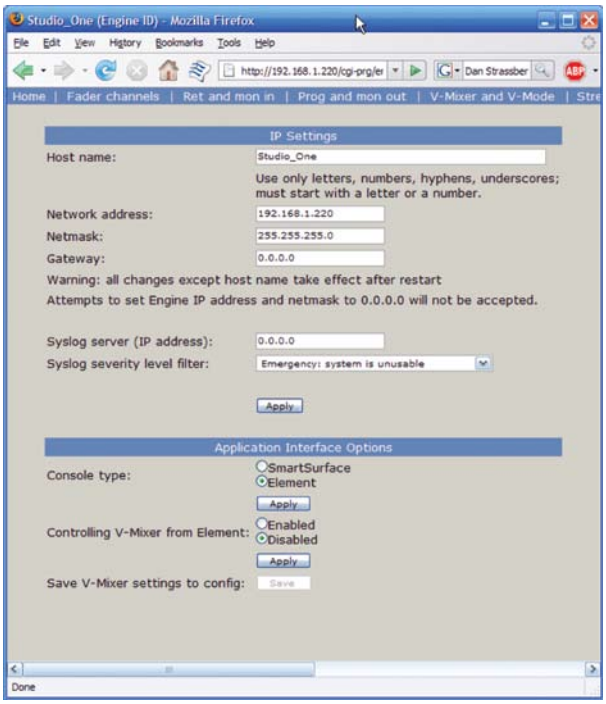


Fig. 7-15: Engine ID menu

©2008 Axia Audio — Rev. 3.0

server on which you wish the logs to be stored.

- **Syslog Severity Level Filter:** This menu choice allows you to set the level at which the Syslog begins storing system events.
  - » **Emergency:** Generates Syslog only if the system is completely unable to operate.
  - » **Alert:** Syslog contains only messages notifying you that immediate attention is needed for the system to remain operative.
  - » **Critical:** Syslog contains only messages about critical system errors.
  - » **Warning:** Syslog contains only messages about errors that could cause the system to become unstable.
  - » **Notice:** System is operating normally, but Syslog contains messages about unusual events that you should know about.
  - » **Informational:** Logs all informational messages. This includes all routine events.
  - » **Debug:** Captures all system activity for diagnostic use.

  After making changes, click the **Apply** button to save them.

Application Interface Options Section

- **Console Type:** Tells the StudioEngine whether it is interfacing with an Element or SmartSurface console. Only needed for Element consoles running software earlier than v2.0; Element v2.X and above automatically selects the correct setting upon boot.

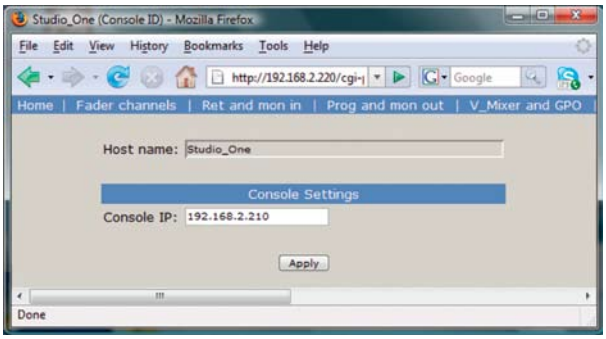


Fig. 8-5: Console ID screen

- **Controlling V-Mixer from Element: Enable** or **Disable** the ability to control VMix from the Element.
- **Save V-Mixer settings to config:** Click the **Save** button to save VMix settings in the StudioEngine's config file.

Console ID Menu

This screen is used during setup to specify the IP address of the console that will interface with this StudioEngine. Simply enter the IP address of the console and click **Apply**.

System Menu

This screen contains several software utilities and information fields pertaining to your StudioEngine.

- The **System Status** section of the page provides information about the physical state of the StudioEngine.
  - » **System Status** and **Network Status** inform you as to the health of the StudioEngine and its connection to the Axia network.
  - » **CPU Usage** indicates how hard the Engine is working.
  - » **Link Speed** tells you the type of link with which the Engine is connected to the local network switch.
  - » **Network Usage In** and **Out** show how much bandwidth is being consumed on the Engine-to-switch link.
  - » **Temperature** monitors the temperature of the StudioEngine CPU. If the temperature exceeds operating limits, a warning will appear on the Engine's front-panel display.
  - » **Fan 1** and **Fan 2 Speed:** Monitors the fan speeds of the Engine's two internal cooling fans.

- The **Configuration Management** section lets you make backups of and restore your Engine settings. Use the **Browse**, **Backup** and **Restore** buttons to accomplish these actions. Use the **Reset Cfg** button to reset all of your StudioEngine's settings to an "as-shipped" state. This command will completely erase any choices you have made and reset your Engine to the way it came from the factory, so use with caution!

- The **Command To Engine** section lets you **Re-**

©2008 Axia Audio — Rev. 3.0

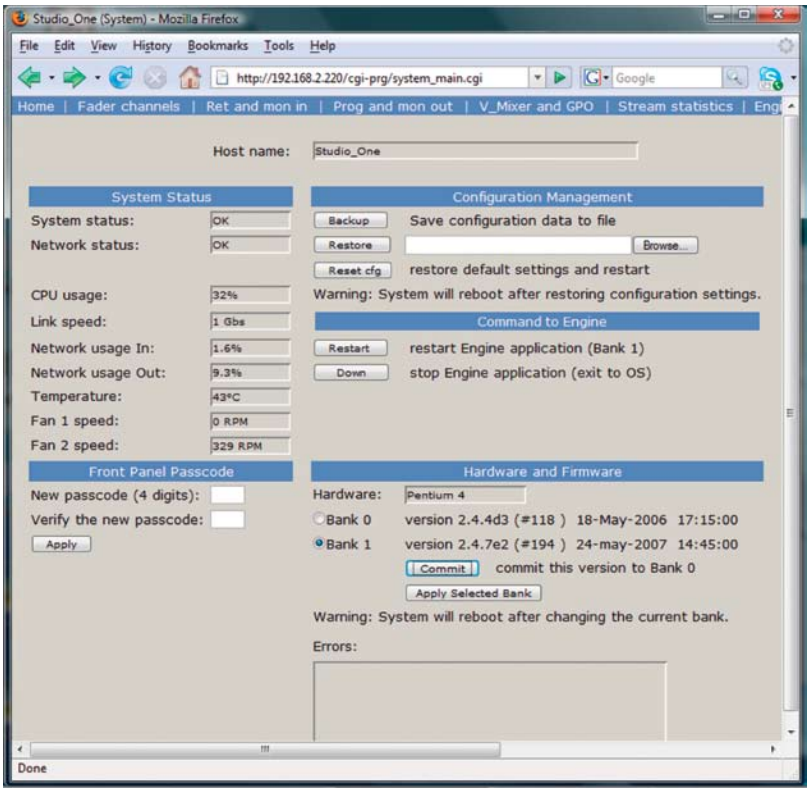


Fig. 8-6: System screen

start the StudioEngine, performing a complete reboot. Clicking the **Down** button exits the Axia StudioEngine application and allows a connected terminal to access the Linux OS of the Engine. This is a diagnostic tool and should only be used at the direction of the Axia Support team.

- The **Front Panel Passcode** fields allow you to passcode-protect the physical controls on the Engine's front panel.

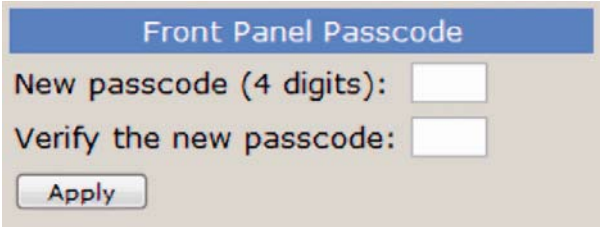


Figure 8-7: Front Panel Passcode controls

By default, access is open. However, if your Engines are located in a public space in your facility, you may wish to set a code to limit front-panel access to authorized personnel. To enable this feature, enter a four-digit code in the top box shown in Figure 7-18, then enter it again in the lower box to confirm. Click **Apply** to activate passcode protection.

The **Hardware and Firmware** section is where you upload and apply new operating software for the StudioEngine. Use the **File** field and the **Browse** and **Upload** buttons to upload new software to the Engine.

Where are my buttons? Unlike Element, where you can store multiple versions of operating software, the StudioEngine has only two software "banks." If both banks are full, you must eliminate one, or you will not see the **File**, **Browse** or **Upload** controls. Instead, you'll see...

The **Commit to Bank 0** button. Pressing this button takes the software in Bank 1 and loads it into Bank 0, which opens Bank 1 for new software.

» You can run Engine software from either Bank 0 or Bank 1. Siimply click on the radio button next to the software bank you wish to run from, and click **Apply Selected Bank**. The Engine will reboot after changing banks, so plan this operation for off-hours or switch program output to another studio to avoid program interruptions.

» The **Errors** message field will show diagnostic messages about the state of the StudioEngine should any fault conditions exist.

Diagnostics Menu

The Diagnostics menu provides tools to help Axia Support in the event your system needs attention. To enable a Diagnostics Trace, click the radio button corresponding to the **Simple, Medium** or **Detailed** options, then click **Apply**.

The other controls here provide other options relating to generating diagnostic information. Please use these as instructed by Axia Support.

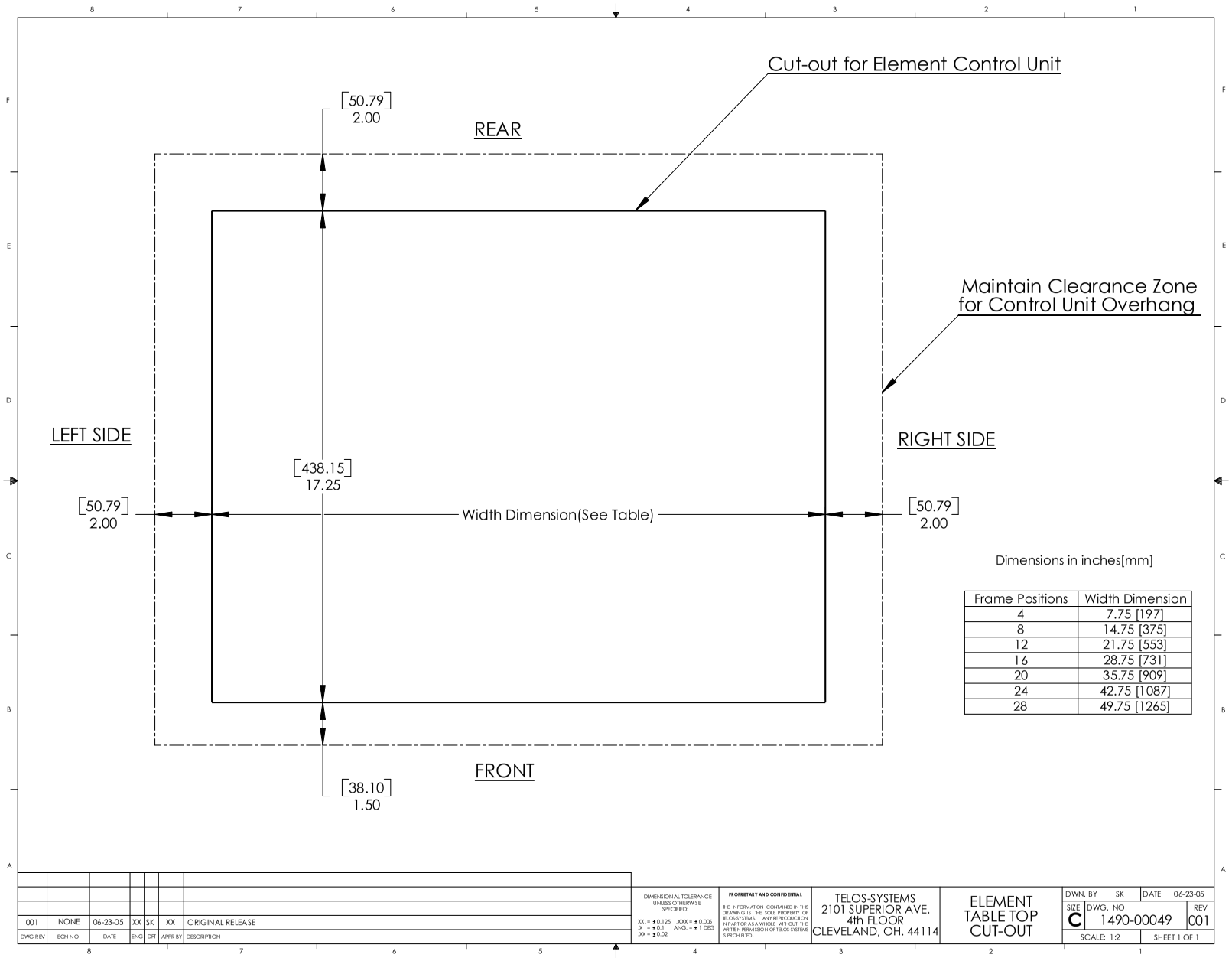
©2008 Axia Audio — Rev. 3.0

Featureless paper

this page would have been empty

save for this haiku.

©2008 Axia Audio — Rev. 3.0



Dimensions in inches[mm]

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

DWG REV	ECN NO	DATE	ENG	DFT	APPR BY	DESCRIPTION
001	NONE	06-23-05	XX	SK	XX	ORIGINAL RELEASE

DIMENSIONAL TOLERANCE UNLESS OTHERWISE SPECIFIED:
XX. = ±0.125 .XXX = ±0.005
.X = ±0.1 ANG. = ±1 DEG
.XX = ±0.02

PROPRIETARY AND CONFIDENTIAL
THE INFORMATION CONTAINED IN THIS DRAWING IS THE SOLE PROPERTY OF TELOS-SYSTEMS. ANY REPRODUCTION IN PART OR AS A WHOLE WITHOUT THE WRITTEN PERMISSION OF TELOS-SYSTEMS IS PROHIBITED.

TELOS-SYSTEMS
2101 SUPERIOR AVE.
4th FLOOR
CLEVELAND, OH. 44114

ELEMENT
TABLE TOP
CUT-OUT

DWN. BY SK | DATE 06-23-05
SIZE C | DWG. NO. 1490-00049 | REV 001
SCALE: 1:2 | SHEET 1 OF 1

©2008 Axia Audio — Rev. 3.0

Dark, dark as coal tar

it pours out, bitter and strong.

That's Starbucks for you.

©2008 Axia Audio — Rev. 3.0

Appendix A: Table of Inputs and Outputs

Main Outputs	Type	Comments
Program-1	Stereo	Main stereo bus (Usually refered to as Program)
Program-2	Stereo	Second stereo bus (Sometimes refered to as Audition)
Program-3	Stereo	Third stereo bus (Sometimes refered to as Utility), signal post fader, before on/off.
Program-4/Record	Stereo	Fourth stereo bus (Sometimes referred to as Auxiliary), signal pre-fader, before on/off.

Aux Inputs and Outputs	Type	Comments
Aux Send-1 Output	Stereo	Aux send bus can be used for processing or independent mixes
Aux Send-2 Output	Stereo	Aux send bus can be used for processing or independent mixes
Aux Return-1 Input	Stereo	Allows processing to be folded back into a mix
Aux Return-2 Input	Stereo	Allows processing to be folded back into a mix

Monitor-related Outputs	Type	Comments
C/R Monitor Main	Stereo	Control room monitor speakers, source and level controlled by SS c/r monitor control
C/R Monitor Direct	Stereo	Control room monitor output, source same as main, fixed level output
Main Headphone	Stereo	Control room (board op) headphone, source and level controlled by SS headphone control
Studio Monitor Main	Stereo	Studio monitor speakers, source and level controlled by SS studio monitor control
Studio Headphone Talent	Stereo	Studio (talent) headpones, source same as main, fixed level output, with talkback
Studio Headphone Guest	Stereo	Studio (guest) headphones, source same as main, fixed level output, no talkback
Preview	Stereo	Allows the connection of external powered speakers.
Talk to External	Mono	Allows board operator mic to talk to other devices, a logic command is associated
Talk to CR Audio	Mono	Allows the Talk to CR audio mix to drive an external destination

Monitor-related Inputs	Type	Comments
External Preview Input	Stereo	Allows an external path into the preview speakers, a logic command is associated
External Monitor Input 1	Stereo	Allows an external source to be monitored by c/r or studio selectors
External Monitor Input 2	Stereo	Allows an external source to be monitored by c/r or studio selectors
External Monitor Input 3	Stereo	Allows an external source to be monitored by c/r or studio selectors
External Monitor Input 4	Stereo	Allows an external source to be monitored by c/r or studio selectors

Source Inputs	Type	Comments
Microphone Input *n*	Mono	Typical installations have 4 or 8 microphone sources per studio
Analog Line Input *n*	Stereo	Typical installations have 16 analog line sources per studio
Digital Line Input *n*	Stereo	Typical installations have 16 digital line sources per studio

Source-related Outputs	Type	Comments
Feed-to-Source A *n*	Mono	Mono mix-minus output feeds the left side of a stereo connection. “Talk to...” function enabled.
Feed-to-Source B *n*	Mono	Mono mix-minus output feeds the right side of a stereo connection. “Talk to...” function disabled.
Individual Hdphone Feed *n*	Stereo	Individual headphone feed (for talent and guest mics) with *Talk to x* function enabled.

©2008 Axia Audio — Rev. 3.0

Dark storage unit

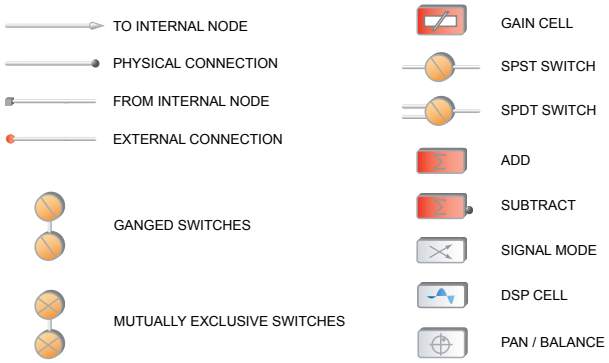
what lies past your gaping mouth?

Lonely ReVoxes.

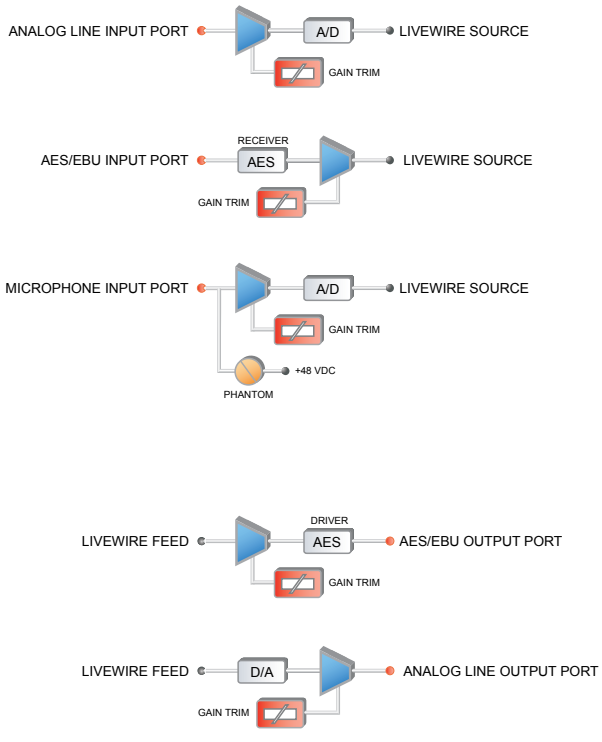
©2008 Axia Audio — Rev. 3.0

Appendix B: Block Diagrams

Legend

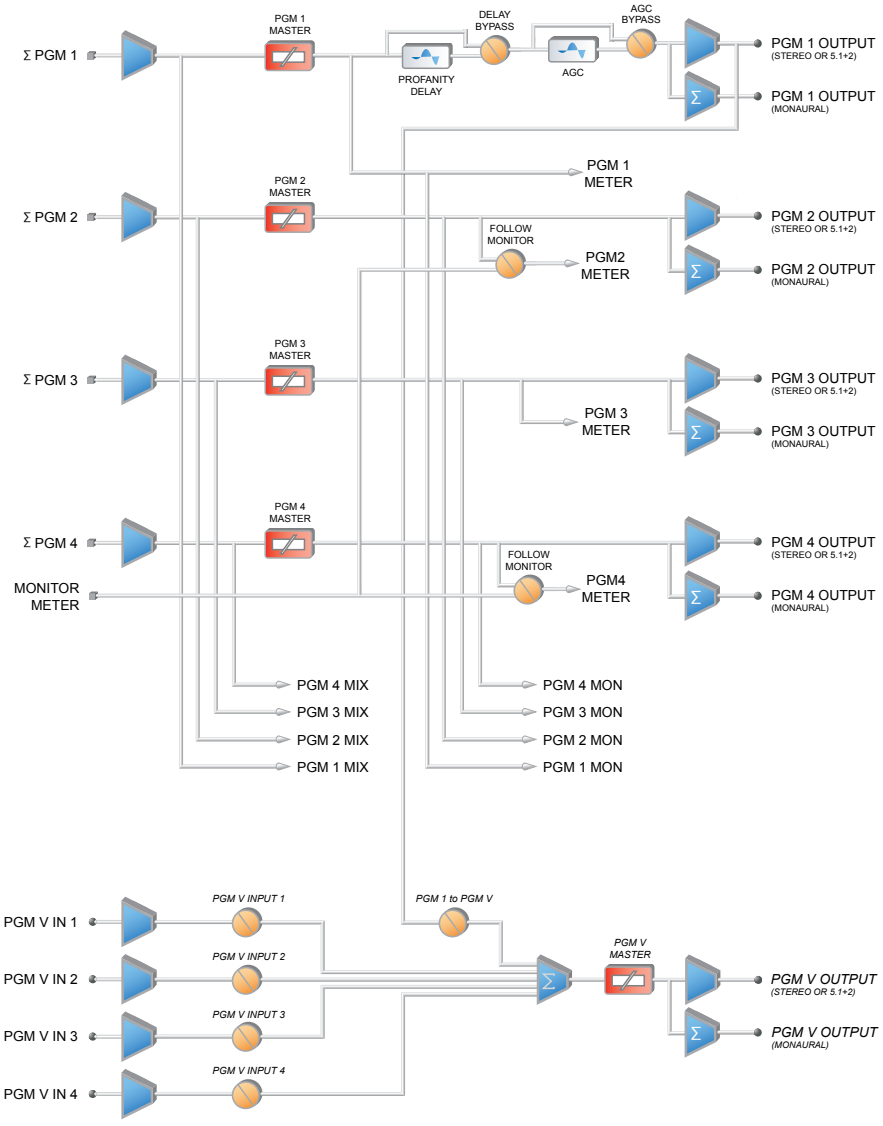


Input & Output Interface Blocks (External to Engine)



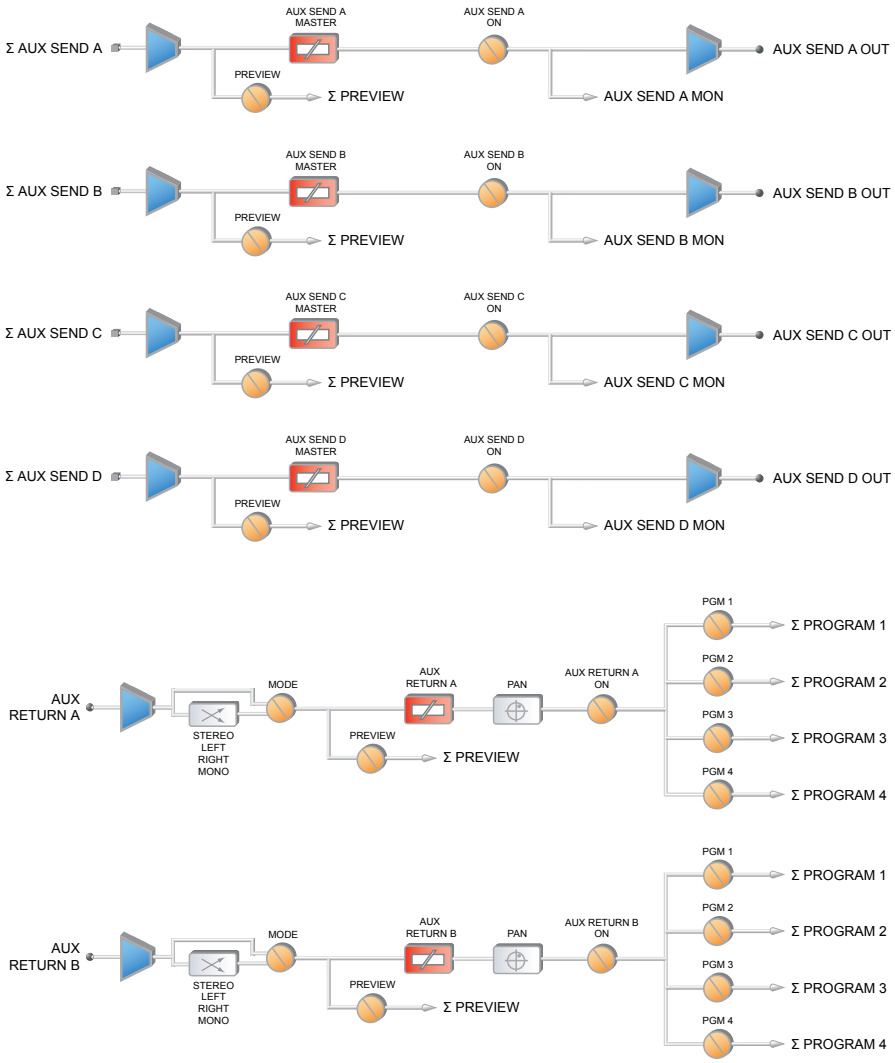
©2008 Axia Audio — Rev. 3.0

Element Master Output Block



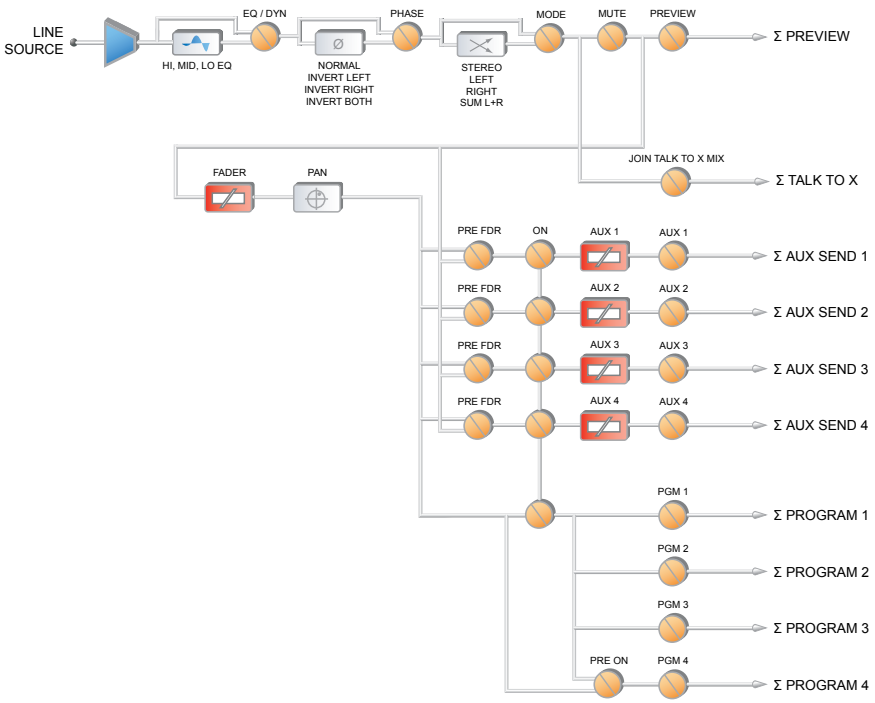
©2008 Axia Audio — Rev. 3.0

Aux Send / Aux Return Blocks

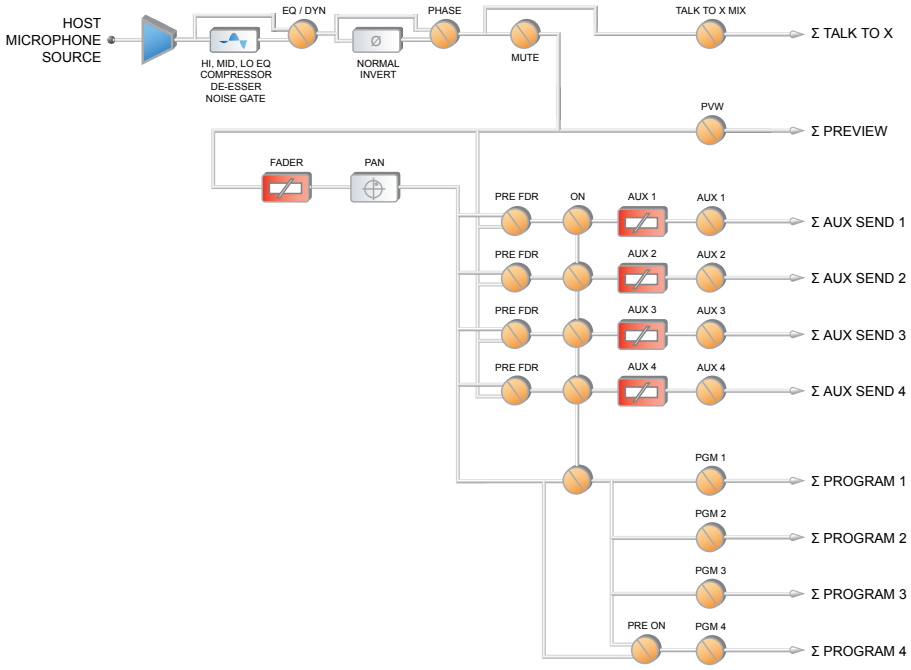


©2008 Axia Audio — Rev. 3.0

Line Input Channel Block

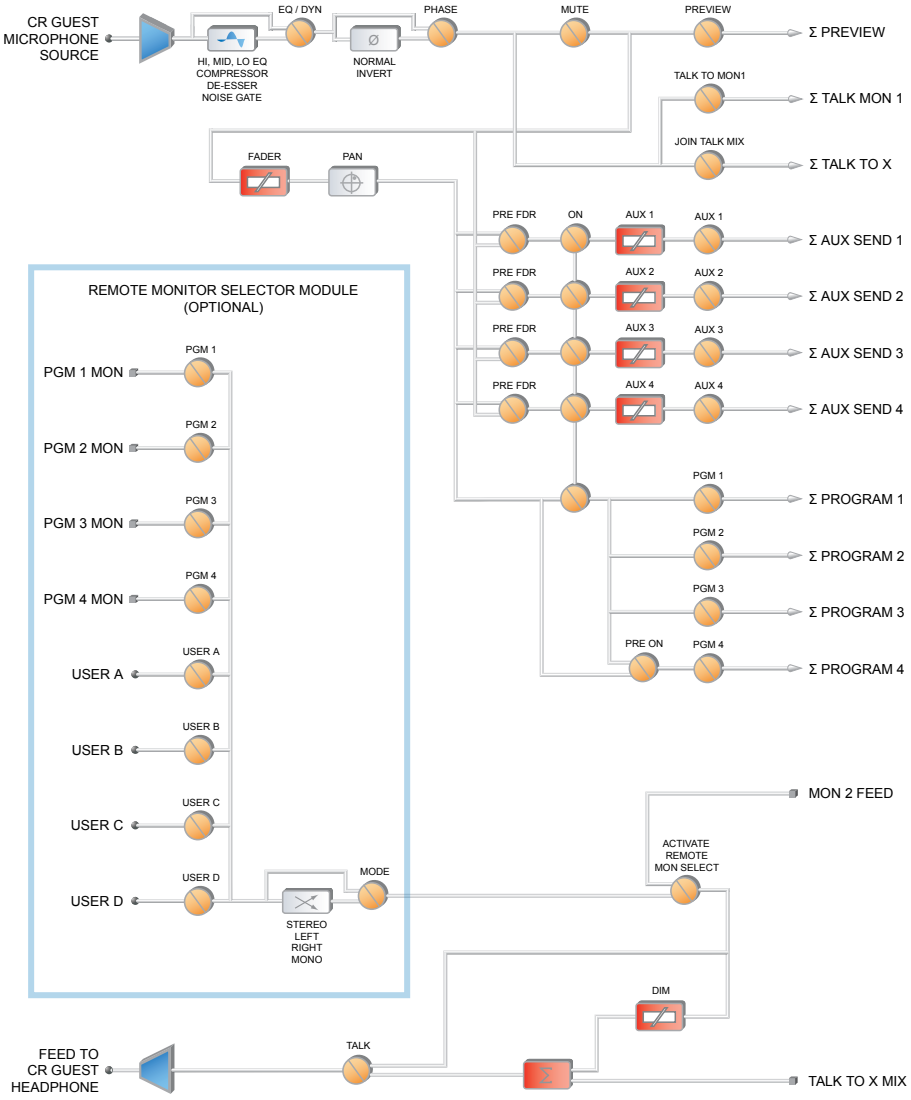


Operator Microphone Channel Block



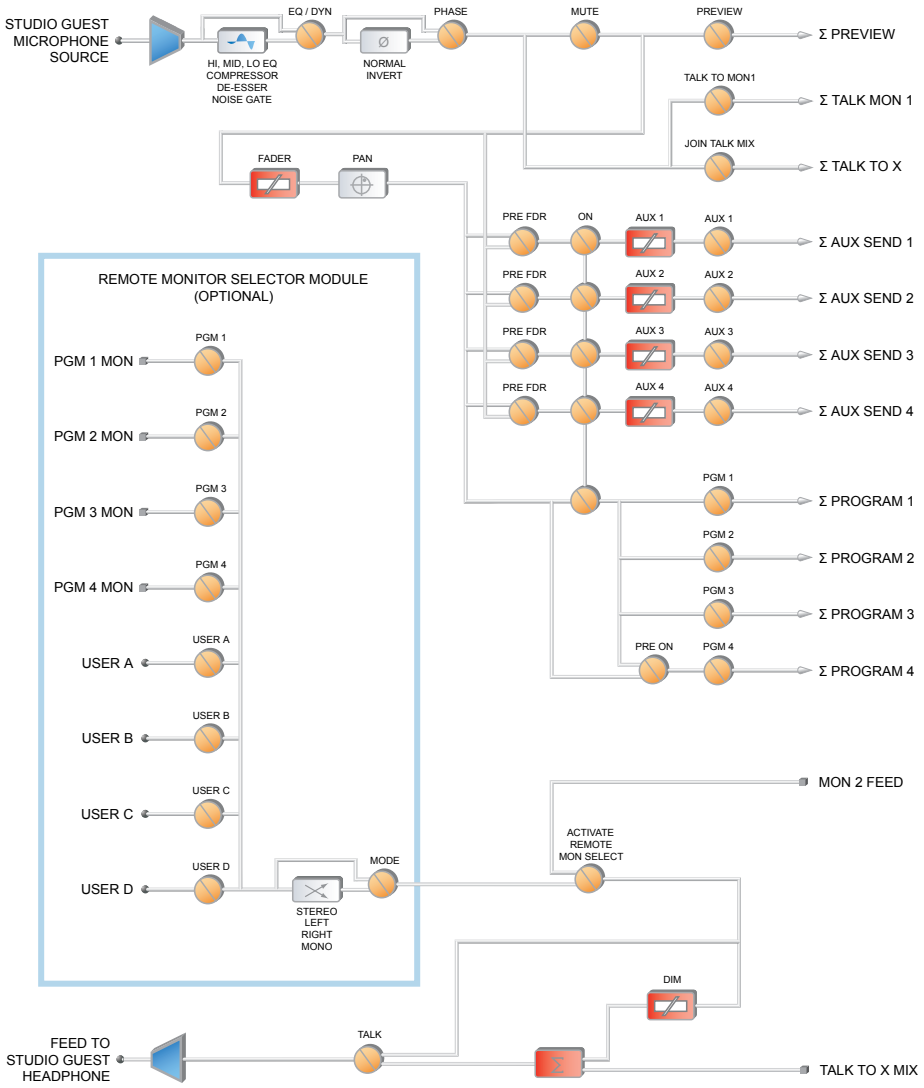
©2008 Axia Audio — Rev. 3.0

Control Room Guest Microphone Channel Block



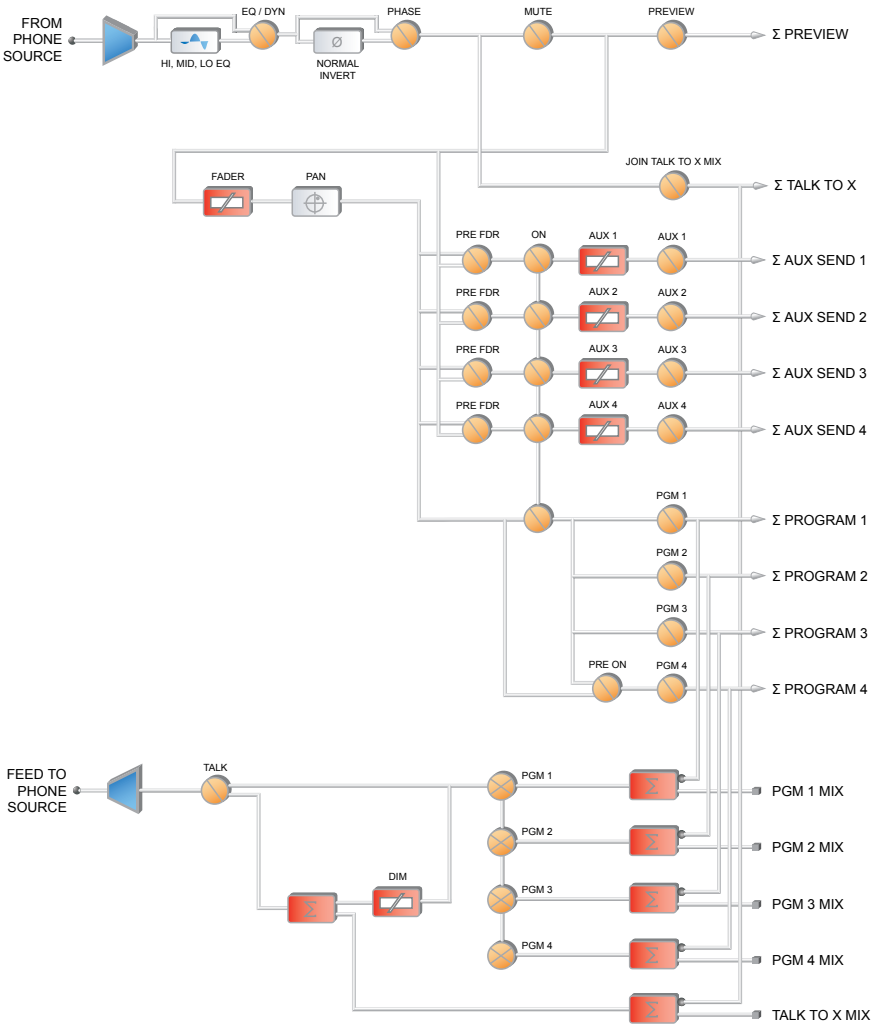
©2008 Axia Audio — Rev. 3.0

Studio Guest Microphone Channel Block



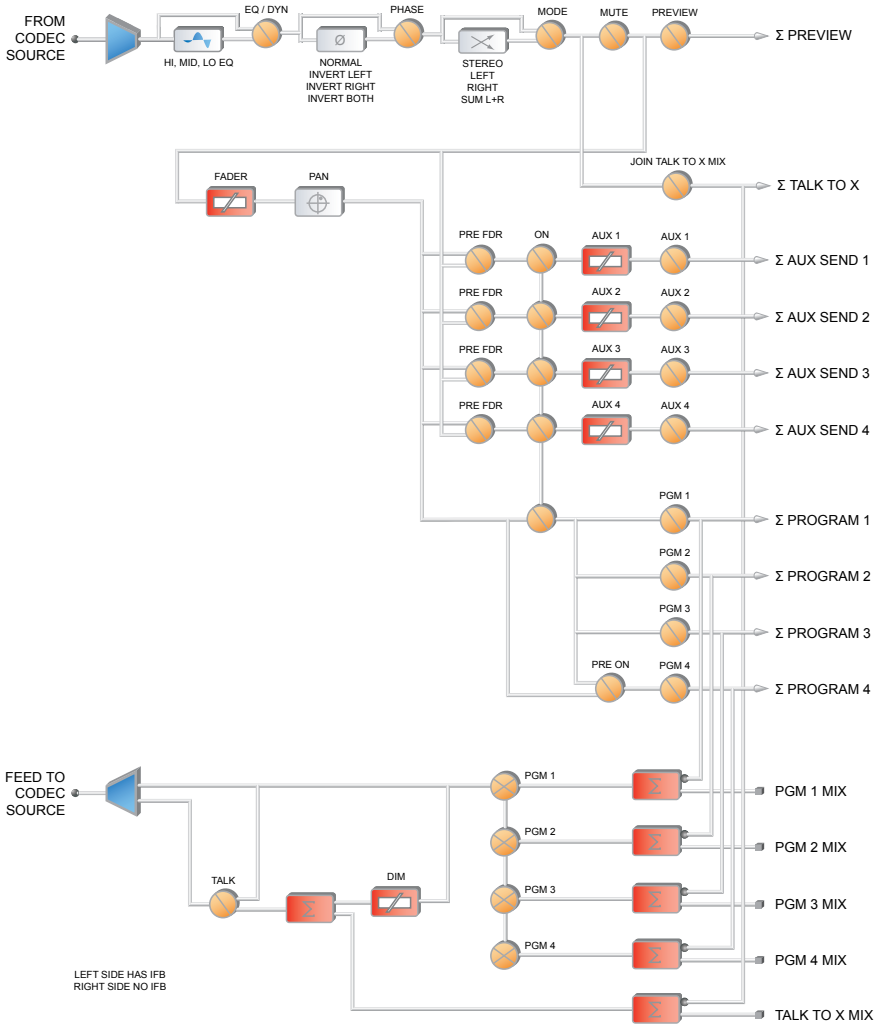
©2008 Axia Audio — Rev. 3.0

Phone Channel Block



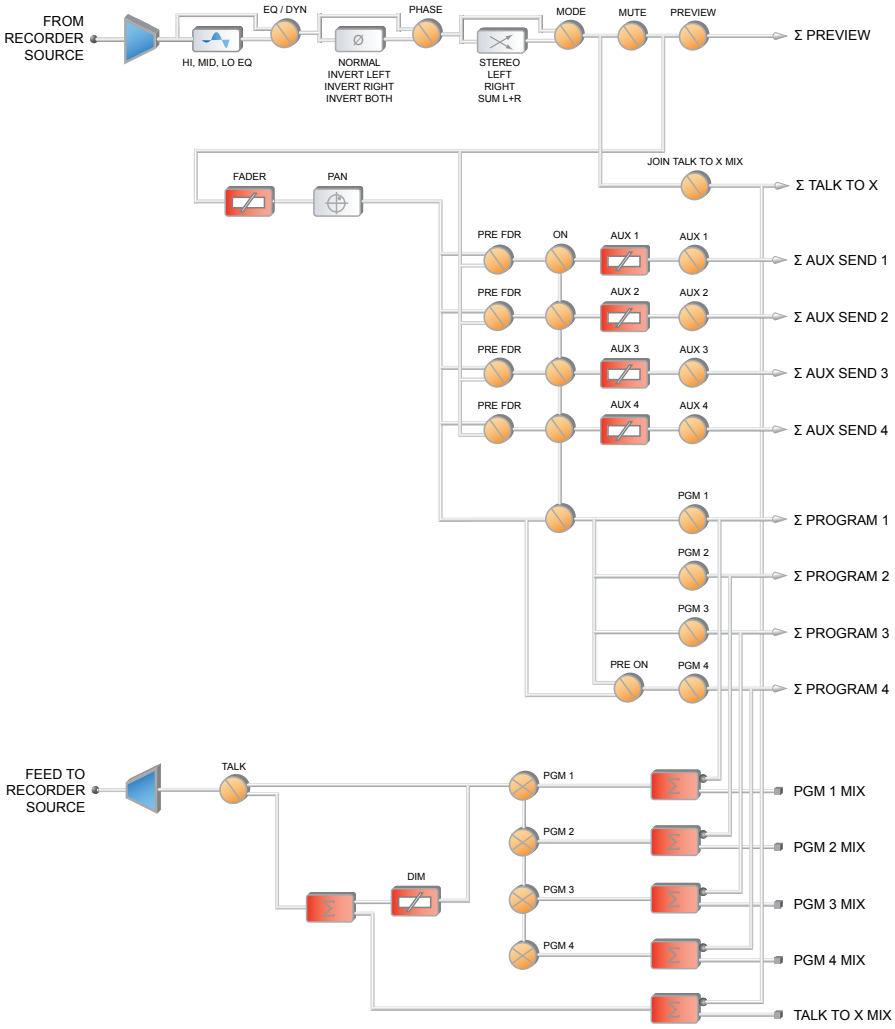
©2008 Axia Audio — Rev. 3.0

Codec Channel Block



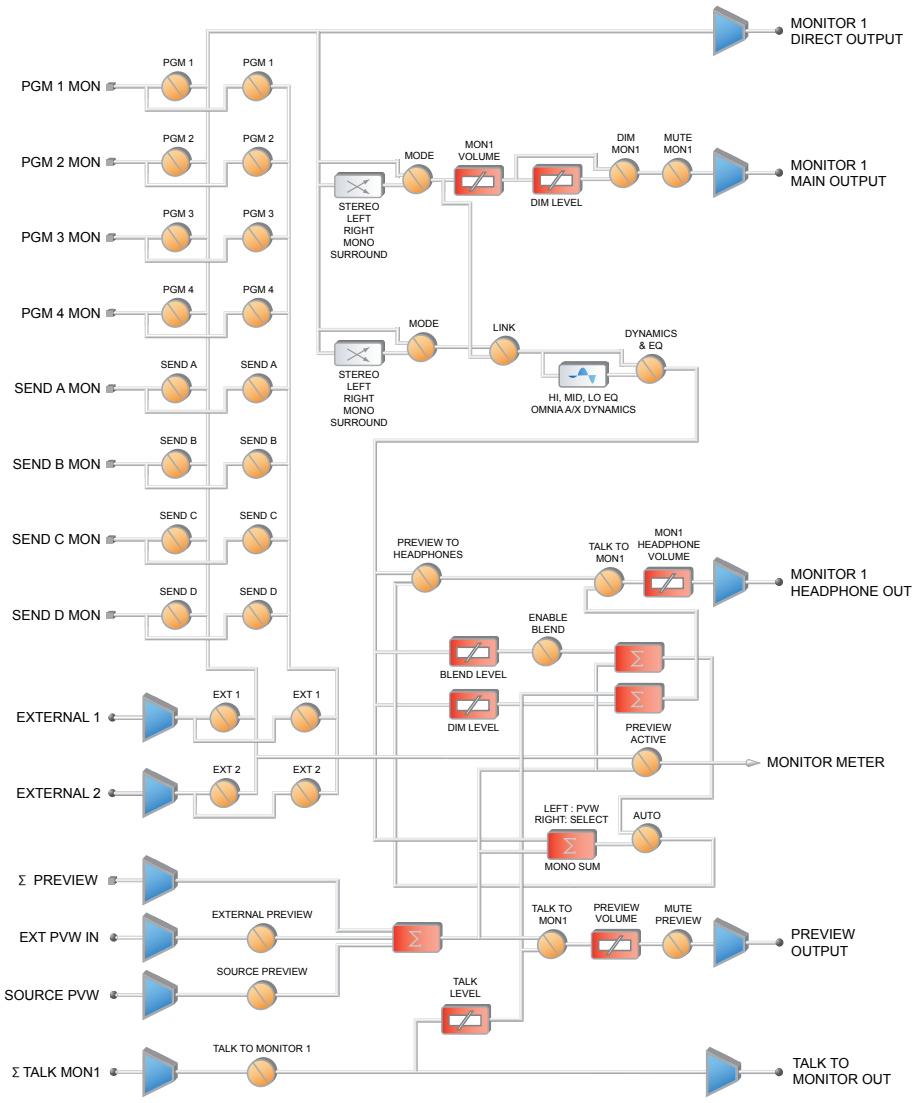
©2008 Axia Audio — Rev. 3.0

Recorder Channel Block



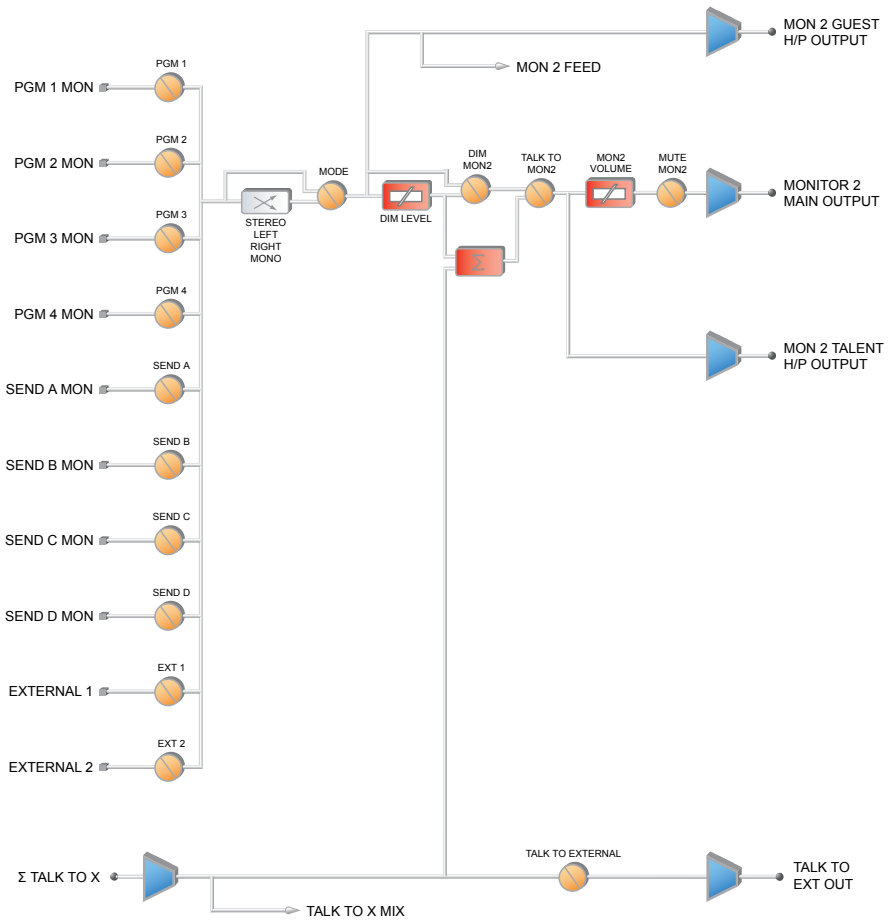
©2008 Axia Audio — Rev. 3.0

Monitor 1 Block



©2008 Axia Audio — Rev. 3.0

Monitor 2 Block



©2008 Axia Audio — Rev. 3.0

At the workbench, you

give thanks: you'll never change a

Schadow switch again.

©2008 Axia Audio — Rev. 3.0

Appendix C: FAQ / Diagnostics / Maintenance

This appendix contains answers to some frequently-asked Element setup questions, and some troubleshooting procedures intended to determine if factory service is needed. These procedures are not intended to take the place of a conversation with Axia support personnel; should you need to contact us for support, please use the contact information listed on Page iii of this manual.

Q: How do I set the IP address of Element? Press and hold the **Fader Options** key located on the upper right of the Master Module for 5 seconds; the connected VGA screen will display the IP Address Book.

Q: The Element web interface asks me for a password, but I haven't set one.
Element is password protected by default to discourage tampering. We recommend setting your own password (using the IP Setup screen described in Chapter One), but until you do, the default password is the lowercase letter "q."

Q: I just added a new module to my Element and the module display reads `INACTIVE`. What do I do?
When new modules are added to Element, you must notify the housekeeping system that they're there by using "Capture Mode." To do this, press and hold the **Help** key located next to Element's control knob on the Master Module. After five seconds, the console will enter "Capture Mode" and the new modules will be activated — you'll know because all module alpha displays read `Capture Mode`. To exit "Capture Mode", press the **Enter** key, also located next to the Control knob. Your new modules are now ready to use.

Q: I get a "Page Not Found" error when I try to connect to Element using my Web browser. What's wrong?
It's possible that your computer's network configuration needs adjustment. Livewire networks do not assign IP addresses dynamically (DHCP), so double-check to see if your computer is actually a part of the network — that its network adapter has a valid IP address and the proper subnet mask value assigned. Assuming your computer is running Windows, you can view these settings using a utility called IPCONFIG.EXE. Here's what to do:

1. Double-check Element's IP address — press and hold the **Control Options** key and all three **Talk To...** keys for 5 seconds, until the IP Setup screen appears on the right display screen. The IP address and subnet mask values are displayed on the first two lines. Write this down for reference.

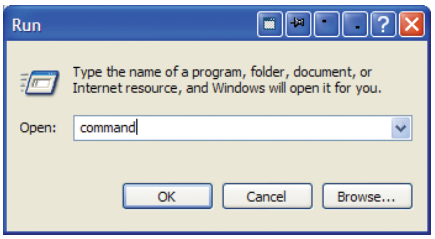


Figure C-1: starting a command box

2. On your Windows computer, click on the **Start** button and choose **Run...** from the menu. In the box that appears, type "command" (or "cmd") to start a DOS command box, then click the **OK** button.
1. A black box with white type will appear. Type **ipconfig** in the box, and press the "Enter" key. You'll be rewarded with a screen that looks like this: The screen displays your computer's IP address, Subnet Mask and Gateway settings. Write these down and compare them to those previously obtained from your Element.

Explanation: The **IP address** is usually expressed as four decimal numbers, each representing eight bits, separated by periods. This is known more technically as "dotted quad notation."

An IP address has two parts: the identifier of a particular network and an identifier of the particular device (which can be a server or

©2008 Axia Audio — Rev. 3.0

a workstation) within that network.

Since networks vary in size, there are four different address formats or classes you can use to construct them. We recommend that Livewire networks be constructed as **Class B** networks to allow enough unique device addresses for expansion.

To learn more about IP addressing and network construction, we recommend reading Cormac Long's excellent 4-part IP Network Design, available at http://tinyurl.com/64bb5 .

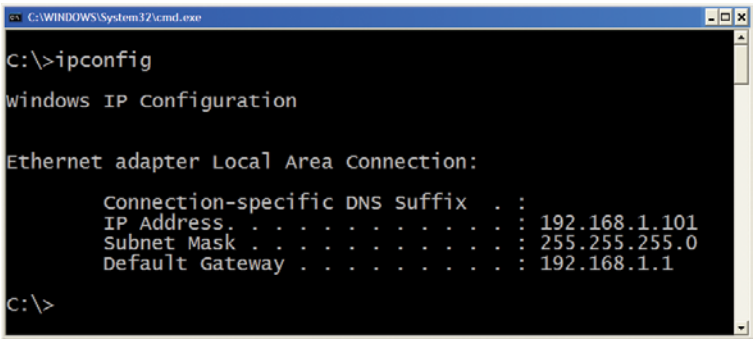


Figure C-2: IPCONFIG command displays computer's IP settings

4. Write down the values displayed and type "exit" to close the DOS command box. Compare the two IP addresses to be certain that your computer is on the same network as your Element; in a Class B network, this means that the first six digits of the IP addresses *must match*. If they do not, change the IP address of your computer's Network Interface Card. Consult your Operating System's documentation for specific instructions on how this is done. Also compare the Subnet Mask values; these must be exactly the same on both your computer and your Element. If they are not, change your computer's Subnet Mask settings to match that of Element.

Note that changing a computer's IP settings and/or Subnet Mask settings often requires a reboot before those settings take effect.

Once your computer is using the same IP network address and subnet as your Element, you should be able to access your Element's web page.

If you've confirmed network settings and still cannot access your Element, "Ping" the Surface to determine whether it is responding to your requests. Here's how:

1. Use the **Run...** command previously described to open a DOS command box. At the command prompt, type **ping** ***xxx.xxx.xxx.xxx*** (where *x* represents Element's IP address).

2. If your Element is responding to commands, the **ping** command will produce a screen resembling Figure C-3.

Results like these, showing packets sent and returned, indicate that your Element is active and responding to requests. However, if the **ping** command produces

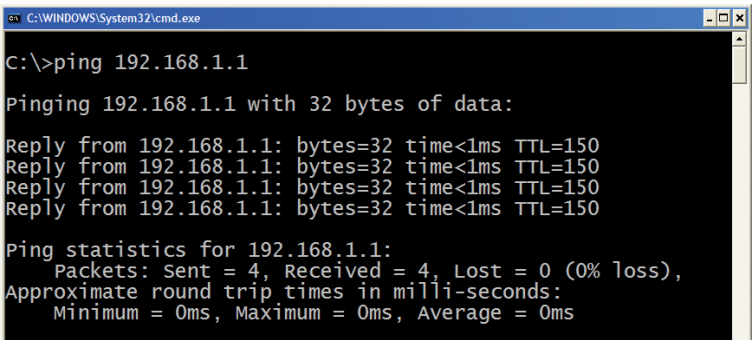


Figure C-3: Successful PING. Element is responding normally.

results like those in Figure C-4, your Element is not responding and must be restarted. Disconnect the Element power supply from the mains, wait 1 minute, then restore power. If this doesn't solve the problem, you may not have your computer and Element on the same LAN or VLAN. Try connecting the two direct-

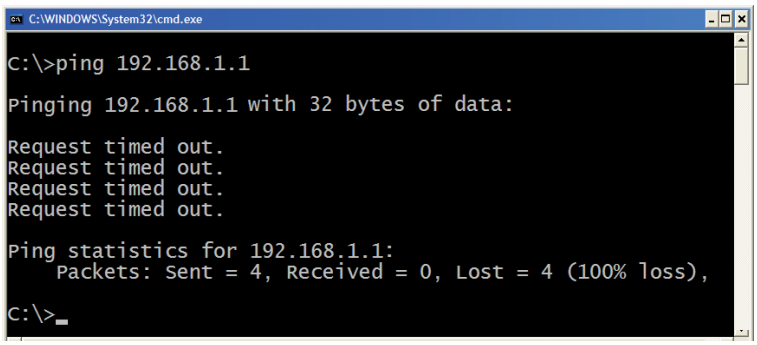


Figure C-4: PING command showing unresponsive Element

©2008 Axia Audio — Rev. 3.0

ly with an Ethernet crossover cable (for greater detail on cable diagrams, read the *Introduction to Livewire: System Design Reference and Primer*, which can be downloaded from www.AxiaAudio.com.)

Q: I don't see the meter displays in my Web browser when I open the pages for my Audio Nodes.
The web pages use Java to render the meter displays. First, double-check your browser options to confirm that Java is enabled, and turn it on if necessary. If Java is not installed on your computer, you will need to install it. Download the free Java installer from http://www.java.com and follow the instructions to install Java; once installed, the meter displays should be visible in your Web browser.

Q: How do I set the time on the Element?
Press and hold the **Timer Options** key on the Master Module. After five seconds, the Clock & Timer Options key will be displayed. All clock adjustments may be made from this screen.

Q: My Element GPIO/Power Supply/CPU doesn't boot and just beeps when I plug it in. What do I do?

The Element CPU uses a VIA EPIA ME6000 mainboard with Award BIOS. If your Element CPU cannot boot, the computer tries to tell you you what is wrong using POST codes (an acronym for "Power-On Self Test").

Award states that the only POST code their BIOS. uses is one long beep and then two short beeps, which indicate a graphics card problem. The reason that only one POST code is used, according to Award, is that the BIOS will attempt to display error codes for other, non-video-related problems on-screen if at all possible. If the BIOS cannot initiate the display adapter, is issues the POST code for a display error, which must be corrected before any other errors can be determined.

However, some other POST codes are known to exist. Consult the table that follows.

POST Code	Error
1 Long, 2 Short	Video card error — replace or re-seat the card.
Repeating beeps	Memory Error — Memory is either damaged or badly seated
Repeating Hi/Low beeps	Damaged or overheating CPU
High-frequency beeps	Overheating CPU

Jostling in transit has been known to loosen the edge-connected cards used in Element's CPU. If you hear only a POST code beep upon plugging in your CPU, disconnect power, remove the screws from the top of the Element GPIO/Power Supply and remove and re-seat all cards found on the internal mainboard; typically, this will solve the problem. If it does not, please contact Axia support for further assistance.

Installing A New Module

Adding a module to Element is a very simple, straightforward proceedure, but some minor disassembly of the surface will be necessary if the new module is accompanied by an overbridge alphanumeric display.

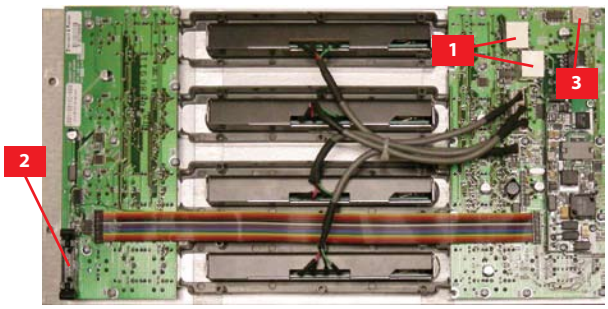


Figure C-5: Underside of module showing (1) RJ45 connectors, (2) display panel connector, and (3) module ID encoder.

Before beginning the installation, inspect your new module. Take note of three features on the underside of the module: the RJ45 connectors, the ribbon cable connector for the overbridge alpha display panel, and the module ID encoder. (Note that not all modules require an overbridge display and thus a ribbon connector will not be present.)

Your module kit will contain some or all of the fol-

©2008 Axia Audio — Rev. 3.0

lowing items:

- The specified module
- An overbridge alphanumeric display panel
- An overbridge window for the display panel
- A ribbon connector
- A 1-foot RJ45 patch cable

Figure C-6: Module kit

If you believe any of thse items are missing, please contact Axia Support immediately.

Tools you'll need for module installation are:

- A #1 Philips-head screwdriver
- A 2.5 mm. hex wrench
- A $^{5}/_{64}$" jeweler's screwdriver

Before beginning, you'll need to set the ID for your new module. Each Element module must have a unique ID. A quick way of choosing an ID is to count how many modules are installed on your Element; typically the factory installs modules with IDs set in numerical order from left to right (Master Modules should always have their ID set to "0"; more on this later). So, if you have 4 factory-installed modules in your Element, you can safely set the ID of your new module to "5". To set the module ID, use a jeweler's screwdriver to gently rotate the indicator (shown in Figure C-7) to the desired number.

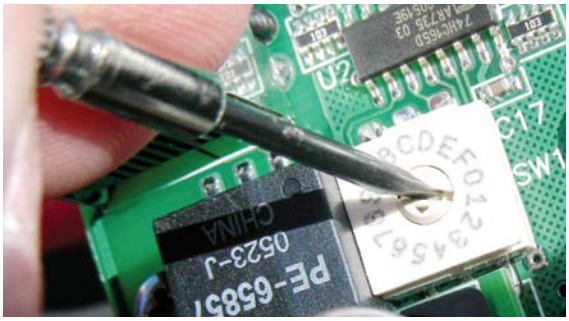


Figure C-7: Setting the module ID

To begin installation, remove the blanking panel occupying the position you wish to install your new module into. Use the hex wrench to remove the hex-head screw(s) at the top of the panel.

If your new module has an accompanying overbridge display, you will need to remove one of Element's end-panels in order to insert it. Ascertain which end panel is closest to the position your new module will be mounted in; this is the one you'll remove.

There are eight machine screws that hold each Element end panel to the frame: 5 in the "bucket" and 3 under the overbridge (shown in Figure C-8, below). To access these screws, you'll need to remove the module closest to this end panel and tilt the overbridge forward..

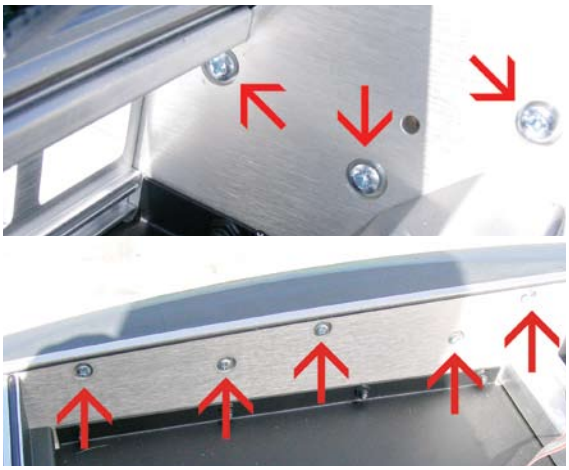


Figure C-8: Locations of end-panel fastening screws in the overbridge compartment (above) and in the "bucket" (below).

Once the screws are removed, the end panel pulls off easily. Note the locating hole in the end panel that receives the overbridge hinge pin.

If there are any overbridge displays between the end of the overbridge and the position in which you'll be mounting your alpha panel, you must remove them. To do so, remove the ribbon connector from the circuit board by gently pushing the two "ears" on each side of

©2008 Axia Audio — Rev. 3.0

the connector socket. This will eject the connector.

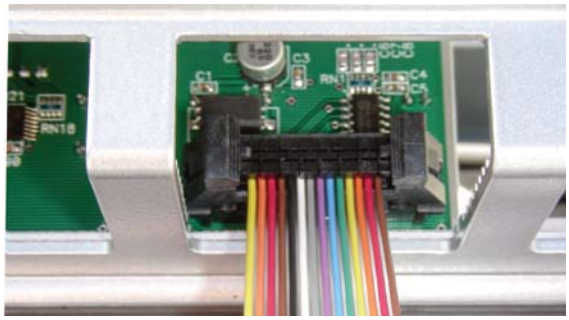


Figure C-9: Location of alpha display ribbon connectors as seen from rear of overbridge

Note: Ribbon connectors must be connected to the alphanumeric panels "tail-down", that is, with the ribbon emerging from the **bottom** of the connector.

You may now slide the disconnected alpha panels out of the overbridge mounting channel, along with the blanking panel occupying your new display's position.

Carefully slide your new alphanumeric panel into the mounting channel. Note that the metal portion of the display has rounded edges which fit into the round frontmost channel; the circuit-board portion of the display **must** have its top and bottom edges located in teh **first groove** behind the round channel.

Figure C-10: Detail of overbridge mounting channel

After all of the overbridge panels have been located in the mounting channel, reconnect their ribbon cables. Connect the ribbon cable that came with your new display, and thread it from the overbridge into the "bucket" through the passage provided.

Re-mount the endpanel, taking care to locate the overbridge hinge pin in the hole provided.

Now, connect the RJ45 patch cable supplied in your module kit to any free jack on Element's connection panel, located in the center of the overbridge compartment.

Note: You've probably noticed that 4-space Element modules have **two** RJ45 connectors on their undersides. These are "daisy chain" connectors; in the event that all of the jacks on your Element's connection panel are in use, you can connect a new module to the unused jack on any other module.

Connect the other end of the patch cable to the RHJ45 connector located on the bottom of your new module, connect the alpha display ribbon connector.

Note: Ribbon connectors must be connected to the modules "tail-up", with the cable emerging from the connector toward the **top** of the module.

Your new module should look as shown in Figure C-11. Once all cables are connected, seat the module in the frame and secure it with the provided 2.5 mm hex-head screws(s).

Figure C-11: Reverse view of connected module. Note ribbon cable is connected "tail-up".

Replace any other modules you may have removed, and secure them as well.

The final step in the module installation process is to "capture" the new console configuration. To enter Cap-

©2008 Axia Audio — Rev. 3.0

Figure C-12: Alpha display indicates Capture Mode

ture Mode, press and hold the **Help** key on the Master Module for 5 seconds. The alphanumeric displays will change to read `Capture Mode`; this enumerates all of the installed faders in the control surface. Press the **Enter** key to exit Capture Mode; your new module is active and your Element is ready to use.

Module Diagnostics

Element software contains a hidden routine that allows you to perform fader, lamp and alphanumeric display diagnostics on each module should you suspect that one of these items requires service.

To enter the diagnostic mode, press and hold the **Help** and **Right Arrow** keys found on Element's Monitor Module for 5 seconds.

Figure C-13: Press and hold these keys for 5 seconds to enter module diagnostic mode.

Upon entering the lamp diagnostic mode, the VGA monitor connected to Element will display an initial screen with a list of all installed modules (as shown in Figure C-14). You'll see a description of the type of each module; preceding the description are a set of numbers in brackets.

To begin fader diagnostics, use the Arrow keys to highlight the fader you wish to test and press the Enter

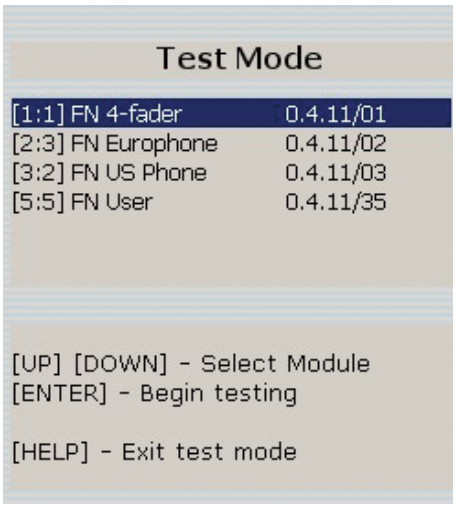


Figure C-14: Module diagnostic selection

key. You will see, onscreen, a graphical representation of the selected module's controls, including faders, lighted selection keys and on/off switches.

In the first phase of the test, all of the module's lamps are lit and you are instructed to turn each one off. This serves as a combination lamp and switch test.

When the lamp/switch test is complete, you'll be prompted to perform a fader operation test. As you move the faders on the module being tested, you will see their onscreen images move as well, and the alpha displays on Element itself will display changing encoder values as the faders are moved.

When you have finished testing a module, you have

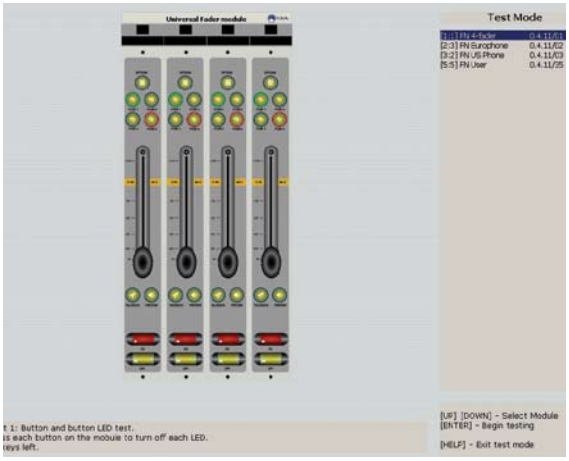


Figure C-15: Module lamp test

©2008 Axia Audio — Rev. 3.0

the option of testing other modules, or of exiting the test routine.

Fader Cleaning Procedures

Element's single-element, conductive-plastic faders were chosen for their long life and reliable operation, but jocks will be jocks: it's inevitable that sooner or later you'll need to pull a fader for cleaning.

There are no replaceable parts in the faders used in Element. If fader movement has become rough, either the lubricant on the glide rails has evaporated or foreign material has gotten into the fader. **Dow Corning 510** is the preferred glide rail lubricant as it will not migrate to the contact fingers like other lubricating oils.

Element modules can be serviced "hot." If you disconnect a module, the StudioEngine will retain the faders' most recent settings until the hardware is reconnected.

Tools and supplies you will need to remove and service a fader:

- A $^{1}/_{16}$" hex wrench (for fader removal)
- A 2.5 mm hex wrench (for module removal)
- A jeweler's screwdriver ($^{5}/_{64}$" or smaller)
- Cotton swabs
- Dow Corning 510 lubricating oil

Fader Disassembly and Cleaning

1. Use the 2.5 mm hex wrench to remove the retaining screw(s) at the top of the module requiring service. With the screwes removed, lift the module at the top, tilting it upward, and pull the locating tab at the bottom of the module from the mounting channel.
2. Turn the module over and locate the ribbon cable that connects the module to its overbridge display (if present). Gently press the "ears" on each side of the board connector to release the cable.
3. Locate and disconnect the RJ45 patch cable that connects the module to Element's connection panel.

Figure C-16: Tilt module up to remove from frame.

4. Remove the four-conductor fader cable from the pins on the bottom of the fader (Figure C-17).

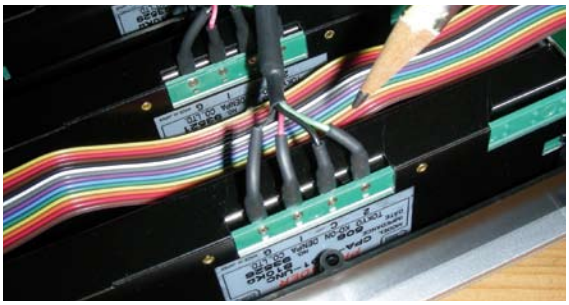


Figure C-17: Remove the fader connector cable

5. Now that the fader is disconnected, turn the module right-side up. Remove the knob from the fader to be serviced by pulling gently upward.
6. Using the $^{1}/_{16}$" hex wrench, remove the two hex screws at the top and bottom of the fader slot. The fader will drop out from the bottom of the module.
7. Lay the fader assembly on your work surface, label-side up. Remove the snap-on fader assembly

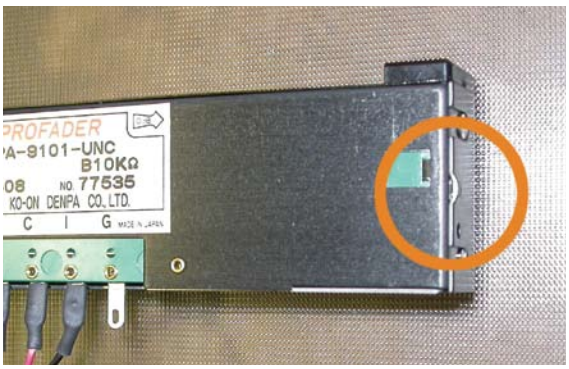


Figure C-18: Pry point on fader cover

©2008 Axia Audio — Rev. 3.0

cover; it's held in place by round stamped bosses at each end. With the fader sitting label up and the connector pins to the front, you'll see a pry-point on the right end of the fader cover (Figure C-18). Use a jeweler's screwdriver to gently pry off the cover.

8. Use a dry cotton swab, or a cotton swab wet with distilled water, to clean the fader parts.

Note: The use of chemical cleaners on the conductive plastic will substantially shorten fader life. Never touch the fader slider contact fingers while cleaning the fader parts.

Always use a clean dry swab to dry off the conductive plastic tracks after cleaning. If the fader rails are noticeably dirty, wipe them off using a dry cotton swab before lightly lubricating the top rail with Dow Corning 510.

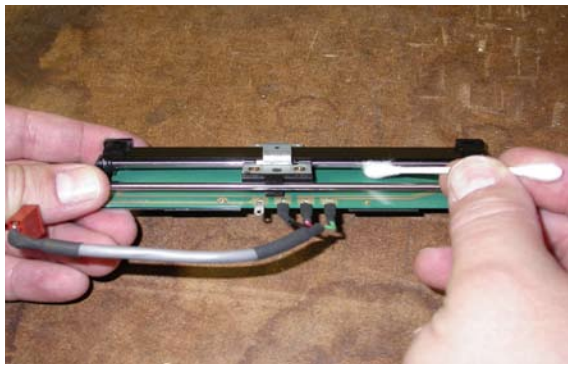


Figure C-19: Cleaning the fader glide rails

If coffee, a soft drink or other sugared liquid has been spilled into the fader, remove it from the module as soon as possible and remove the top cover of the fader. Hold the fader under hot running water while moving the fader slider back and forth to dissolve the sugars and other chemicals. Thoroughly dry the rails and conductive plastic using dry cotton swabs and then lubricate the top fader rail with Dow Corning 510.

Lubricating the Glide Rail

Move the fader slider to the middle of its travel and place one drop of Dow Corning 510 lubricant on the top rail on either side of the fader slider bushings. Move the slider through its full travel to distribute the lubricant. Wipe off any excess lubricant from the rubber stops at each end of the glide rail. Normally only the top rail (the one the fader slider bushings glide on) requires lubricant.

Reinstalling The Fader

1. Snap the cover onto the fader body.
2. From below, reinsert the fader into position in the surface, taking care that the end marked ∞ is installed adjacent to the channel's **On / Off** keys.
3. Reinstall the hex screws from the top of the surface. Do not overtighten.
4. Press the fader knob firmly onto the stem.
5. Reconnect the fader leads to their mating pins on the bottom of the fader. The sequence is:
   * Green wire to pin "2"
   * Black wire to pin "C"
   * Red wire to pin "1"
   * Bare wire to pin "G"
6. Reconnect the RJ-45 patch cable and alpha panel ribbon cable to the module.

Note: *The* ribbon connector must be connected to the module "tail-up", with the cable emerging from the connector toward the **top** of the module.

7. Lay the module into the frame, engaging the locating tab at the bottom of the module and tilting the top downward into the frame. Be careful not to pinch the ribbon cable between the module and frame.
8. Reinstall hex-head screw(s) at top of module.

©2008 Axia Audio — Rev. 3.0

Appendix D: Working With Phone Hybrids

One of the advantages of an Element/Livewire studio is the smooth integration of Telos telephone interface equipment with the mixing console. Element telephone control modules let you put the phone controls in front of the operator and near the associated faders. Mix-minus is handled naturally and automatically, without the need for the usual installation and operator contortions. And each fader channel has its own mix-minus capability, so you will never run out of busses.

Since advanced Telos telephone interfaces have Ethernet connections, they are easily connected to your studio network to be controlled by the Element phone modules. Sophisticated control signaling may then pass between the two systems without the need for the usual parallel connections.

But we recognize that not everyone owns a Telos phone system (even though we wish they did!), so we've included GPIO capability in Element to control simpler telephone equipment that does not have a Livewire interface.

This Appendix describes how to configure Element for use with the Telos Series 2101 and TWOx12 phone systems. The functions to control these systems via Ethernet were added to Element with software version 1.1.1, so you must have this software (or later) to use the features for these phone hybrids described here.

Telos TWOx12 Talkshow System

Utilizing v.3.28 or later software

The Telos TWOx12 has an Ethernet connection for control, so it can be tightly integrated with the Element, using the multi-line phone control module (p/n 2001-00180). Together, they provide powerful and easy-to-use on-air phone capabilities.

Installation and Configuration

> ***Note:*** Your Telos TWOx12 needs to have software version 3.28 or later installed. The latest software is available on the Telos website at www.telos-systems.com/support/ .
>
> The following proceedure also assumes that your Call Controller module was either installed by the factory, or that you have installed it yourself following the proceedure described in the "Installing A New Module" section of this manual's Appendix C.

Follow this proceedure to ready your TWOx12 for use with Element:

1. Connect your TWOx12 to the Axia network using it's Ethernet port, and assign it an IP address that is acceptable for use within your network. (Please refer to the TWOx12 documentation for instructions on how to configure IP addresses.)
2. If you have not already done so, connect your telephone lines to the TWOx12 and confirm that they are working by checking the unit's LCD status display. Confirm with a connected Telos Desktop Director.
3. Connect both TWOx12 audio inputs and outputs to an Axia Analog Node. *You should use only the ANALOG inputs and outputs for the hybrid audio interface.*

> ***Why use Analog I/O?*** The use of analog I/O with TWOx12 is necessary because the unit's AES input and outputs use only one connector each (AES left channel for Hybrid 1 and right channel for Hybrid 2). Due to this configuration, these channels cannot easily be separated; thus we must use TWOx12's analog I/O to ensure that each hybrid receives its own correct, unique Mix Minus.

4. Designate an Analog Node output as Program-On-Hold feed for the TWOx12 and connect it to the TWOx12's Program-On-Hold input.
5. Using the Audio Node's Source Configuration web page, enter a text name and channel number for each

©2008 Axia Audio — Rev. 3.0

TWOx12 input. Configure the inputs for low-delay "Livestream" or "Fast" audio mode. Within the Element's Source Profile configuration web page create a Source profile for this channel number and define it as a PHONE type.

6. Using the Analog Node's Destination Configuration web page, select the same channel number for each for each output that you used for the corresponding input. Select "To source" for the Type. This tells the node to receive the Mix Minus Backfeed generated by the Engine for the source of the same channel.
7. Open the Element's Module Manager/Phone Channels web page. If you have the Caller Controller Module installed, you will see the following at the top of the screen:

Figure D-1: Phone configuration box

If you don't see this, or if the ID or Module information is incorrect, you will need to "Capture" your current module configuration. This is normally done at the factory, but if you have added the phone module yourself, you will need to take care of this before proceeding. To capture, press and hold the **Help** key on Element's Navigation Module for 5 seconds, until CAPTURE appears in the source name displays. On the phone module, you will see the assigned ID number in the upper left line icon position. You can change this by pushing the adjacent line button. End Capture Mode by pressing the **Enter** key on the Navigation Module.

In Figure D-1 above, the "Module" field displays both the module type and the position. The first digit, "Type", will be "3" to identify a Call Controler module. The next digit is the module number as set by the rotary switch on the back of the module.

8. Enter the IP address you assigned your TWOx12 in the **Server Address** field, plus your TWOx12's **Login** and the **Password**. (The default TWOx12 login and password are both "telos" (lower case, no quotation marks).
9. Now, we must associate a TWOx12 "show" with an Element Show Profile. In your Web browser, navigate to Element's "Source and Profiles/Configuration/Show Profiles" web page. Select the Element Show Profile you wish to configure, and click on **Phone**.

Figure D-2: Configuring a TWOx12 "show"

Enter the TWOx12 Show Name either by typing it in, or by choosing it from the drop-down list. For TWOx12, possible default show names are "Hybrid1&2", "Hybrid1", or "Hybrid2". (The latter are for split-mode operation where the TWOx12 is shared between two studios. See the TWOx12 manual for details on setting up this operational mode.)

For TWOx12, the Host Name is *always* "TWOx12". The "Show Password" field is empty by default.

Note: Hybrid 1 is controlled by the left row of telephone line selection buttons, and Hybrid 2 by the right row. Normally, Hybrid 1 is the system's "main" hybrid, and is tied to the Call Controller's "Next" button, which advances through the lines with each press. If you wish to use Hybrid 2 as the "main" hybrid and have it controlled by the "Next" button, click the "Reversed Hybrid" box as shown in Figure D-2 above.

The TWOx12 has a maximum of 12 lines, so select either "Auto" or "12 Lines" with the Mode Selection radio buttons.

10. Click "Save" and use your Web browser to navigate back to the "Element Sources & Profiles Configuration/Source Profiles/Create New Source Profile"

©2008 Axia Audio — Rev. 3.0

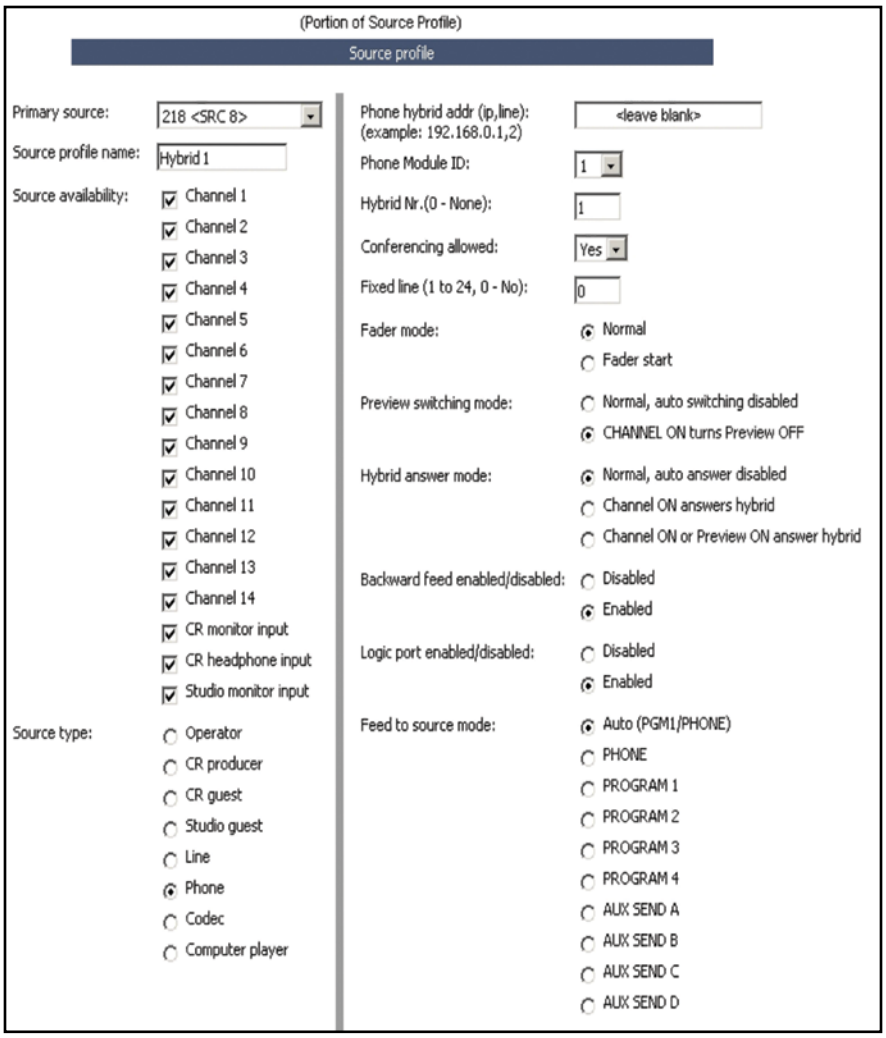


Figure D-3: A Phone Hybrid Source Profile

web page. Choose the "Primary Source" audio channel that you entered in the Analog Node. Give it an Element source name, such as "Hybrid 1", and select "Phone" for Source Type.

The Source Profile for the hybrid should be filled in as follows:

- It is not necessary to enter the **Phone Hybrid IP** address here, since you have already done it globally. **Leave this field blank.**
- The **Phone Module ID** field lets you select from the multiple phone modules that you may have installed. Normally, you will have just one, so select **1** from the drop-down box.
- For **Hybrid Nr.** select the hybrid that corresponds to the Primary Source audio you selected above. The TWOx12 has two hybrids, so enter either **1** or **2** here.
- **Conferencing Allowed** lets you decide if you want to permit locked-on lines for "button mash" conferencing. Unless you specifically wish to prohibit this type of conferencing, select **Yes** from the drop-down box.
- **Fixed Line** lets you assign a line permanently to a specific fader channel, emulating a dedicated hybrid. (Normally this is used with a 4-Phone Fader Module so that its the "Hold" button would control the phone hybrid dedicated to that fader. It can be used with one of the Call Controller's two faders, but will limit operational flexibility.)

The other fields in the Source Profile page are non-phone specific, and may he configured as you wish. (See Chapter Two, "Configuring Inputs", for more details on these options.

Remember that the Telos TWOx12 contains two hybrids, so you need to repeat this procedure to create another source profile for the second hybrid.

Important Note: The TWOx12 has an optional internal mix-minus cross-connect feature. Since Element provides this function automatically, you should disable this feature in TWOx12 to prevent unexpected results.

You'll now need to load the hybrid sources onto the fader channels. This can be done either manually, or automatically using Show Profiles. Naturally, we recommend that you make this assignment in each Show Profile which will use the phone system, since leaving manual assignment to air talent is asking for trouble!

©2008 Axia Audio — Rev. 3.0

For normal operation, you will assign Hybrid 1 to the left fader channel and Hybrid 2 to the right fader channel.

When you are properly connected you should see the following message within Element's Phone Module webpage:

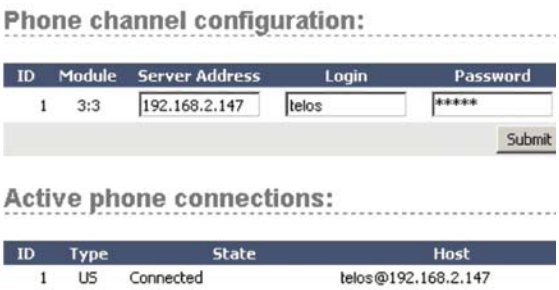


Figure D-4: Connection confirmation

Note: Within Element, each incoming "phone" type audio source has an associated back-feed for automatic mix-minus generation. There are several user configuration options that affect which audio sources are fed to the hybrid under various conditions; please refer to the "Source Profile Options" and "Feed To Source Mode" sections in Chapter 2 of this manual.

Operation

On Element's Call Controller Module, pressing any Line button either answers a call ringing-in or selects an inactive line to make an outgoing call; when an inactive line is "picked up" in this manner, the keypad located on the Element Navigation Module is active and can be used for dialing calls. You'll see visual confirmation on Element's display screen as you dial the number, and if caller ID is available, it is displayed after an incoming call is answered.

Useful Note: Pressing the pound key (**#**) after dialing a number will speed connect time by as much as 2 - 5 seconds! By default, the Telco's central office switch "listens" for a specified period of time after you dial a requested number, waiting in case you decide to dial more digits. Pressing the pound key tells the switch that the dialing sequence is complete, and bypasses this waiting period, connecting you quicker!

Pressing any **Line** key for a line that is already active causes it to be locked in this state (a Lock icon will appear in the Status Symbol window next to the locked line), so that additional lines can be selected without this or other locked lines being dropped. This is useful for making certain that lines on-air are not inadvertently dropped (such as when a talkshow guest is on a call-in line), or for "button mash" conferencing. To clear a locked line, press its **Line** button again; the Lock icon will disappear and the line can then be dropped, either by selecting another line or by pressing the **Drop** key.

Pressing the **Drop** key drops any non-locked lines on the associated line column.

Pressing the **Next** key takes the line that the TWOx12 system has determined to be next in the on-air queue for Hybrid 1. Normally, that is the line that has been on hold the longest. If there is no line on hold, pressing **Next** answers the line that's been ringing-in the longest.

Audio is normally routed to the fader channel adjacent to the **Line** key row. Pressing the **Hold** key on the fader channel puts the call on Hold, from where it can be retrieved from anywhere in the system. (The **Set** key on the fader channel is not used with the TWOx12, and pressing it has no effect.)

Block All puts the TWOx12 into "Busy" mode. This is useful for clearing lines prior to contest call-ins. When you enter "Busy" mode, inactive lines are busied so that callers will receive a busy signal when they attempt to call-in. Lines that are active, i.e. on-air and on Hold are not affected by the **Block All** key; however, if these lines are dropped while TWOx12 is in "Busy" mode, these lines will be "busied out" as well. Note that "Busy" mode has no effect at all on lines which are designated as Hot or Warm Lines. To cancel "Busy" mode, press the **Block All** key again; all busied lines will again be available for incoming callers.

The **Transfer** key activates the Transfer function within the TWOx12. Please refer to your TWOx12 documentation for more information on this feature.

©2008 Axia Audio — Rev. 3.0

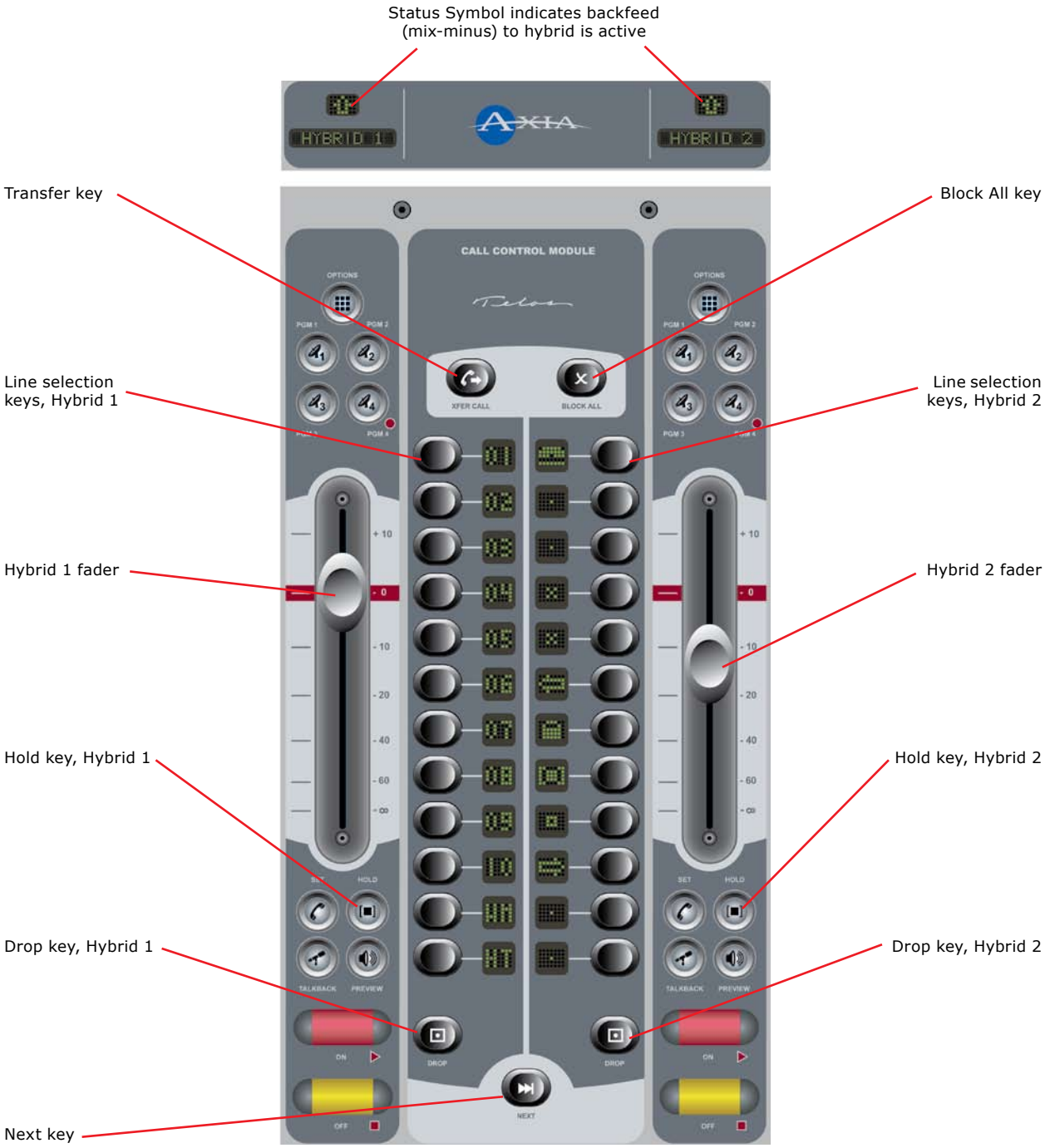


Figure D-5: Call Controller + 2-Phone Fader Module

©2008 Axia Audio — Rev. 3.0

Telos 2101 Multi-Studio Talkshow System

Utilizing v.2.2 or later software

The operation of the Series 2101 is nearly identical to the the TWOx12, but there are additional features that can be configured for use with Element:

- Either 12 or 24 lines can appear on the console (should the Series 2101 be configured for 24-line operation),
- Either two or four hybrids can be controlled by the console, each assigned to a fader channel, and
- Automatic sequential or manual assignment of calls to fader channels.

Installation and Configuration

> ***Note:*** Your Series 2101 system needs to have software version 2.2 or later installed. The latest software is available on the Telos website at www.telos-systems.com/support/ .
>
> The following proceedure also assumes that your Call Controller module was either installed by the factory, or that you have installed it yourself following the proceedure described in the "Installing A New Module" section of this manual's Appendix C.

Follow this proceedure to ready your Series 2101 for use with Element:

- Connect your Series 2101 to the Axia network using it's Ethernet port, and assign it an IP address that is acceptable for use within your network. (Please refer to the Series 2101 documentation for instructions on how to configure IP addresses.) It is recommended that the 2101 System be on its own network segment independent from the Axia network. IP routing between the Axia and 2101 networks will be required. The Element console and the 2101 Hub will then require proper gateway information in order to pass TCP/IP data between your two networks.
- If you have not already done so, connect your telephone lines to the Series 2101 configure it following the instructions in the Series 2101 User's Manual. Once configured, confirm that the system is operational using a connected Telos Desktop Director.

The following steps must be undertaken in each studion in which the Series 2101 will be used:

1. Connect the Series 2101 Studio Interface audio inputs and outputs to an Axia Analog Audio Node. (The hybrid inputs and outputs must be connected via analog since there is no way to separate the two hybrid back-feeds using the AES interface.) Wire to the Left side only of the two node inputs. (Remember, you'll need to set the Signal Mode to LEFT when creating Source and Show Profiles for these inputs.)

> ***Why use Analog inputs?*** The use of analog inputs with Series 2101 is necessary because the unit's AES input and outputs use only one connector each (AES left channel for Hybrid 1 and right channel for Hybrid 2). Due to this configuration, these channels cannot easily be separated; thus we must use Series 2101's analog I/O to ensure that each hybrid receives its own correct, unique Mix Minus.

2. Designate an Analog Node output as the studio's Program-OnHold feed and connect it to the Studio Interface's Program-On-Hold input.
3. If the studio will be using an Extended Hybrid, connect the additional two inputs, two outputs, and Program-on-Hold input as described above.
4. Within the 2101's configuration pages, create a **Login** and **Password**. Note that this step is required *for each Element console with a phone module.*
5. Using the Audio Node's Source Configuration web page, enter a text name and channel number for each 2101 input. Configure the inputs for low-delay "Livestream" or "Fast" audio mode. Within the Element's Source Profile configuration web page create a Source profile for this channel number and define it as a PHONE type.
6. Using the Analog Node's Destination Configuration web page, select the same channel number for each for each output that you used for the corresponding input. Select "To source" for the Type. This tells the node to receive the Mix Minus Backfeed generated by the Engine for the source of the same channel.
7. Open the Element's Module Manager/Phone Channels web page. If you have the Caller Controller Module installed, you will see the following

©2008 Axia Audio — Rev. 3.0

at the top of the screen:

Figure D-6: Phone configuration box

If you don't see this, or if the ID or Module information is incorrect, you will need to "Capture" your current module configuration. This is normally done at the factory, but if you have added the phone module yourself, you will need to take care of this before proceeding. To capture, press and hold the **Help** key on Element's Navigation Module for 5 seconds, until CAPTURE appears in the source name displays. On the phone module, you will see the assigned ID number in the upper left line icon position. You can change this by pushing the adjacent line button. End Capture Mode by pressing the **Enter** key on the Navigation Module.

In Figure D-6 above, the "Module" field displays both the module type and the position. The first digit, "Type", will be "3" to identify a Call Controler module. The next digit is the module number as set by the rotary switch on the back of the module.

8. Enter the IP address you assigned your 2101 Hub in the **Server Address** field, plus the 2101 **Login** and the **Password**. you created for this studio back in Step 4.
9. Now, we must associate a 2101 "show" with an Element Show Profile. In your Web browser, navigate to Element's "Source and Profiles/Configuration/Show Profiles" web page. Select the Element Show Profile you wish to configure, and click on **Phone**.

   Enter one of the 2101 Show Names that you created when setting up the Hub and its Host Name, either by typing them in, or by clicking on the "Phone Module ID" link and choosing it from the drop-down list. If a Show Password is required, enter it here as well. (By default, show passwords are not assigned; unless you have specifically assigned a show password, leave this field blank.)

Figure D-7: Configuring a 2101 "show"

Note: Hybrid 1 is controlled by the left row of telephone line selection buttons, and Hybrid 2 by the right row. Normally, Hybrid 1 is the system's "main" hybrid, and is tied to the Call Controller's "Next" button, which advances through the lines with each press. If you wish to use Hybrid 2 as the "main" hybrid and have it controlled by the "Next" button, click the "Reversed Hybrid" box as shown in Figure D-2 above.

   The 2101 can be used in either 12 or 24 line mode. Select either "Auto," "12 Lines" or "24 Lines" with the Mode Selection radio buttons. Selecting "Auto" will set the mode for the maximum number of lines that the 2101 system permits for this show.
10. Click "Save" and use your Web browser to navigate back to the "Element Sources & Profiles Configuration/Source Profiles/Create New Source Profile" web page. Choose the "Primary Source" audio channel that you entered in the Analog Node. Give it an Element source name, such as "Hybrid 1", and select "Phone" for Source Type. Please refer to Figure D-3 to see the Source Profile screen.

The Source Profile for the hybrid should be filled in as follows:

- It is not necessary to enter the **Phone Hybrid IP** address here, since you have already done it globally. **Leave this field blank.**
- The **Phone Module ID** field lets you select from the multiple phone modules that you may have installed. Normally, you will have just one, so select **1** from the drop-down box.
- For **Hybrid Nr.** select the hybrid that corresponds to

©2008 Axia Audio — Rev. 3.0

the Primary Source audio you selected above. One Series 2101 Studio Interface supports 2 hybrids; adding an Extended Hybrid expands the system to support up to 4 hybrids. Enter **1**, **2**, **3** or **4** here.

- **Conferencing Allowed** lets you decide if you want to permit locked-on lines for "button mash" conferencing. Unless you specifically wish to prohibit this type of conferencing, select **Yes** from the drop-down box.
- **Fixed Line** lets you assign a line permanently to a specific fader channel, emulating a dedicated hybrid. (Normally this is used with a 4-Phone Fader Module so that its the "Hold" button would control the phone hybrid dedicated to that fader. It can be used with one of the Call Controller's two faders, but will limit operational flexibility.)

The other fields in the Source Profile page are non-phone specific, and may he configured as you wish. (See Chapter Two, "Configuring Inputs", for more details on these options.

Repeat this procedure to create a source profile for each hybrid in your studio (two with a standard Studio Interface, four with an Extended Hybrid).

Important Note: Series 2101 Studio Interface units have an optional internal mix-minus cross-connect feature. Since Element provides this function automatically, you should disable this feature in the 2101 system to prevent unexpected results. Set the hybrid mode to "Separate."

You'll now need to load the hybrid sources onto the fader channels. This can be done either manually, or automatically using Show Profiles. Naturally, we recommend that you make this assignment in each Show Profile which will use the phone system, since leaving manual assignment to air talent is asking for trouble!

For normal operation, you will assign Hybrid 1 to the left fader channel of the Call Controller, and Hybrid 2 to the right fader channel. Additional hybrids, if you have them, may be assigned to any channel on a 4-Fader Phone Module.

Note: Within Element, each incoming "phone" type audio source has an associated back-feed for automatic mix-minus generation. There are several user configuration options that affect which audio sources are fed to the hybrid under various conditions; please refer to the "Source Profile Options" and "Feed To Source Mode" sections in Chapter 2 of this manual.

Operation

On Element's Call Controller Module, pressing any Line button either answers a call ringing-in or selects an inactive line to make an outgoing call; when an inactive line is "picked up" in this manner, the keypad located on the Element Navigation Module is active and can be used for dialing calls. You'll see visual confirmation on Element's display screen as you dial the number, and if caller ID is available, it is displayed after an incoming call is answered.

Useful Note: Pressing the pound key (**#**) after dialing a number will speed connect time by as much as 2 - 5 seconds! By default, the Telco's central office switch "listens" for a specified period of time after you dial a requested number, waiting in case you decide to dial more digits. Pressing the pound key tells the switch that the dialing sequence is complete, and bypasses this waiting period, connecting you quicker!

Refer to Figure D-5 for an illustration of the following Call Controller function keys.

Pressing any **Line** key for a line that is already active causes it to be locked in this state (a Lock icon will appear in the Status Symbol window next to the locked line), so that additional lines can be selected without this or other locked lines being dropped. This is useful for making certain that lines on-air are not inadvertently dropped (such as when a talkshow guest is on a call-in line), or for "button mash" conferencing. To clear a locked line, press its **Line** button again; the Lock icon will disappear and the line can then be dropped, either by selecting another line or by pressing the **Drop** key.

Pressing the **Drop** key drops any non-locked lines on the associated line column.

Pressing the **Next** key takes the line that the 2101 system has determined to be next in the on-air queue for

©2008 Axia Audio — Rev. 3.0

Hybrid 1. Normally, that is the line that has been on hold the longest. If there is no line on hold, pressing **Next** answers the line that's been ringing-in the longest.

Block All puts the TWOx12 into "Busy" mode. This is useful for clearing lines prior to contest call-ins. When you enter "Busy" mode, inactive lines are busied so that callers will receive a busy signal when they attempt to call-in. Lines that are active, i.e. on-air and on Hold are not affected by the **Block All** key; however, if these lines are dropped while TWOx12 is in "Busy" mode, these lines will be "busied out" as well. Note that "Busy" mode has no effect at all on lines which are designated as Hot or Warm Lines. To cancel "Busy" mode, press the **Block All** key again; all busied lines will again be available for incoming callers.

The **Transfer** key activates the Transfer function within the Series 2101. Please refer to your Series 2101 documentation for more information on this feature.

The **Set** key on the fader channel is not used with the Seriese 2101, and pressing it has no effect.

Assigning Incoming Lines to Faders

Audio is normally routed to the fader channel adjacent to the **Line** key row. Pressing the **Hold** key on the fader channel puts the call on Hold, from where it can be retrieved from anywhere in the system. (The **Set** key on the fader channel is not used with the 2101, and pressing it has no effect.)

When you have more than two hybrids, you can either assign calls to faders manually using the Call Controller's "conference" (right) phone line column, or let Element assign them automatically.

To assign lines manually, touch the **Off**, **On**, or **Preview** keys on on the fader channel you wish to assign a line to *prior to selecting the line*. You will see the text `Take Line` on the alpha display at the top of the fader channel. Then, press the **Line** key you wish to take on the Call Controller. After pressing the phone line button, the alpha display will change to `Line X`, where *x* is the number of the line you selected, to remind you which line is active on the fader channel. If you don't select a line, the mode will time-out after 10 seconds.

If you have not initiated the manual assignment procedure, the automatic assignment sequence follows the hybrid numbering from the 2101 system. But you can assign hybrids to any fader channel, so you can have any physical sequence that you choose on the Element control surface.

In 24-line mode, calls are normally taken to Hybrid 1. If this hybrid is in use and the associated line is locked, selecting another line causes automatic assignment to the next available hybrid and fader channel. You can make assignments manually by touching **Off**, **On**, or **Preview** before selecting the line, as described above.

Fader channels can also be configured as fixed lines. In this mode, pressing the fader's **On** button selects the associated line directly. Pressing the associated **Line** key on the Call Controller module will also pick up the line, which will be active on its associated fader.

When an incoming line is slaved to a single fader in this manner, pressing the fader **On** button answers a ringing-in call, or, if the line is inactive, selects it so you can make outgoing call using the keypad on Element's Navigation Module. Pressing the **Off** key once will switch off the channel audio as normal; pressing the **Off** key a second time drops the caller. Pressing the **Hold** key puts the call on hold.

Series 2101 and Show Profiles

A different 2101 Show can be assigned to each Element profile so that you can use different groups of phone lines, etc. with different profiles. This is useful if, for instance, a single studion is used to produce multiple syndicated call-in programs, and you wish to use different groups of call-in lines for each show.

To protect calls that are on-air, a new 2101 Show will not be loaded when a Show Profile is loaded *if any hybrids in the existing Show are still active* (i.e., if one or more hybrid faders are **On**). The new 2101 Show will load only after all hybrid faders are switched off, similar to the way Element handles requests for source changes on active faders.

©2008 Axia Audio — Rev. 3.0

Caller ID

Caller ID is displayed for incoming calls. The icon in the overbridge near the source text name changes to show the line status.

Telos Two ISDN Hybrid

The Telos Two has an Ethernet connection for control, so it can be tightly integrated with the telephone control modules in the Element. It uses the 4-Phone Fader Module, which includes a Hold and Set button in addition to the usual complement. All control is via the Ethernet, so no GPIO connection is necessary.

The Telos Two can be configured for ISDN system hold or local hold. The first case is used when you want to be able to retrieve the call from another device sharing the ISDN line.

Installation and Configuration

The Telos Two needs to have software version 2.0.51p or later installed. The latest software is available on the Telos web site support page.

1. Connect your Telos Two to the Ethernet and assign it an IP number that is acceptable for use within your network.
2. Connect your ISDN telephone line to the Telos Two and confirm that it is working by checking the LCD status display.
3. Connect both Telos Two audio inputs and both outputs to an Axia Analog node. The hybrid inputs and outputs must be connected via analog since there is no way to separate the two hybrid back-feeds using the AES interface. Wiring to the left side of the two node inputs will accomplish our goal. Remember to set the Signal Mode to LEFT when creating the Source and Show Profiles.
4. Using the audio node's Source configuration web page, enter a text name and channel number for each input. It might be convenient to include the telephone number as part of the text name. Configure for "Fast stream" audio mode.
5. On the node's Destination configuration web page, select the *same channel number* for each output as was used for the corresponding input. Select "To source" for the Type. Configure for "Fast Stream" audio mode.
6. Open the Element Sources & Profiles Configuration web page. Choose Source Profiles, then Create New Source Profile. In your new source profile, choose the Primary Source that you entered in the audio node's setup page. Give it an Element source name and select "Phone" for source type. Enter the phone hybrid IP address, including the hybrid number, in the form

   xx.xx.xx.xx,n

   where the "x"s represent the IP number, and the "n" represents the hybrid number.

The Telos Two has two hybrids, so you need to repeat this procedure to create another source profile for the second hybrid.

> ***Important Note:*** The Telos TWO has an optional internal mix-minus cross-connect feature. Since Element provides this function automatically, you should disable this feature in the Telos TWO (set hybrid mode to "separate") to prevent unexpected results.

Within Element, each incoming "phone" type audio source has an associated back-feed. There are some user configuration options that affect which audio sources are fed to the hybrid under various conditions, explained in the mix-minus/back-feed section of this manual.

Operation

Load your newly-created hybrid source on a 4-Phone Module fader. This can be done either manually or automatically using Show Profiles.

Pressing the fader channel **On** button answers a ringing-in call or selects an inactive line for an outgoing call. Pressing **Preview** has the same effect on the hybrid.

The keypad located on the Element Monitor module

©2008 Axia Audio — Rev. 3.0

can be used to dial outbound calls. It is automatically assigned to the last line that was selected. You'll see the number as it is being entered on the main Element screen. After the number is complete, press the Enter button to start the connection.

Caller ID is displayed for incoming calls. The icon in the overbridge near the source text name changes to show the line status.

The first time you press the **Off** button, the channel audio is switched off. The second time you press the **Off** button, the call will be dropped.

Pressing the module's **Hold** button puts the call on hold. (The **Set** button on 4-Phone Module channels is not used with the Telos Two, and pressing it has no effect.)

Telos One and Delta Hybrids

These hybrids can be controlled using Element's hardware GPIO ports. Either normal 4-Fader Modules or 4-Phone Fader Modules may be used with these hybrids, although there is no benefit from the latter as the **Hold** and **Set** buttons have no function.

First, let's set up GPIO connections. (Please refer to the GPIO section of this manual, or the Axia GPIO Node Manual, for pinout diagrams.)

1. Connect the GPIO START Pulse output to the hybrid's "Line Take" pin, and the GPIO STOP Pulse output to the hybrid's "Line Drop" pin. This will cause the **On** and **Off** buttons on athe fader to which the hybrid is assigned to operate the hybrid remotely. Here's a cable diagram:

Element-to-Telos ONE cable:

Function	Element DB-15M	Hybrid DB-9M
Common	7	6
On	11	1
Off	12	7

Element-to-Telos Delta cable:

Function	Element DB-15M	Hybrid DB-15M
Common	7	6
On	11	1
Off	12	7

2. Connect the hybrid inputs and outputs to an Audio node.
3. Using the Node's Source configuration web page, enter a text name and channel number for each input. It might be convenient to include the telephone number as part of the text name. Configure for "Fast Stream" audio mode.
4. Using the Node's Destination configuration web page, select the *same channel number* for each output as was used for the corresponding input. Select "To source" for the Type. This is how Element's automatic backfeed/mix-minus knows where the backfeed should be sent. Configure for "Fast Stream" audio mode.
5. Open the Element Sources & Profiles Configuration web page. Choose Source Profiles, then Create New Source Profile. In your new source profile, choose the Primary Source that you entered in the audio node's setup page. Give it an Element source name and select "Phone" for source type.
6. Open Element's Module Manager/GPIO Configuration web page. You will see the following:

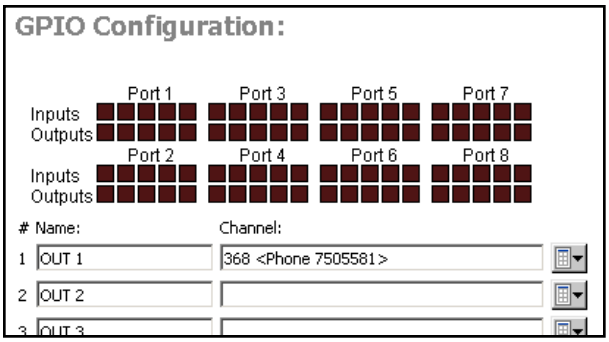


Figure D-8: GPIO setup

For the physical GPIO you will be connecting to the hybrid, enter or select from the drop-down box the Livewire source channel for the hybrid. Note the Livewire channel number and name are keys to the GPIO as well as to audio signals. Element will generate the appropri-

©2008 Axia Audio — Rev. 3.0

ate control signals for hybrids when Phone is selected as the source type.

The status display above the entry boxes show you the GPIO status and are useful for troubleshooting.

Operation

Load the hybrid source onto a fader channel. If you are using the GPIO to control the Hybrid functions for the Telos One or Delta hybrids, pressing the fader channel **ON** button turns the Hybrid ON, and pressing the **Off** button turns the Hybrid OFF.

Telos ONE-x-Six, Direct Interface and 1A2 Systems

To use these legacy hybrids, audio must be connected via Analog nodes. There is a special two-position control panel that can drop into Element to control ONE-x-Six, Direct Interface Module or 1A2 phone systems. This panel communicates directly to the phone systems using a dedicated cable rather than via Ethernet. Line Take and Drop functions cannot be controlled using GPIO on these systems, so the control surfaces supplied with the phone systems must be used to actuate these commands.

Installation & Configuration

1. Install the control module and connect it to the phone equipment.
2. Connect audio input, output, and Program-On-Hold to an Analog node. Use the left channel inputs and outputs for these mono signals.
3. Using the Analog node's Source configuration web page, enter a text name and channel number for each input. It might be convenient to include the telephone number as part of the text name. Configure for "Fast Stream" audio mode.
4. Using the audio node's Destination configuration web page, select the *same channel number* for each output as was used for the corresponding input. Select "To source" for the Type and configure for "Fast Stream" audio mode.
5. Open the Element Sources & Profiles Configuration web page. Choose Source Profiles, then Create New Source Profile. In your new source profile, choose the Primary Source that you entered in the audio node's setup page. Give it an Element source name and select "Phone" for source type. Set the default input mode to Left.

Operation

Load the hybrid source(s) onto the fader channels. Use the control surfaces directly connected to the phone system to select lines and place callers on the hybrids. Faders will not be controlled by the hybrids via GPIO and will need to be manually turned **ON** and **OFF** as needed.

©2008 Axia Audio — Rev. 3.0

Appendix E: Accessory User Panels

Axia manufactures a comprehensive line of accessory control panels that can be placed near talent, host or producer positions to give non-board-op users convenient control over options such as headphone source selection and volume, mic channel on/off/mute, talkback and routing controls. These panels connect to the Axia network in two ways, depending upon their feature set: via GPIO, and via CANBus.

Figure E-1: Axia accessory control panels.

Panels requiring a CANBus connection require a cable connection to the nearest Element console. These include the Headphone Selector Panel, Mic Control/ Headphone Selector Panel and the 4-button LCD Smart-Switch control panels.

Other panels require only a GPIO connection. These include the Mic Control Panel, Producer's Mic Control Panel, 5-Button Film-Cap Switch Panel, and the 5-, 10- and 15-Button Rackmount Switch Panels.

Installing a CANBus Accessory Panel

> ***Note:*** Before beginning installation, make certain that you have the latest software installed for your StudioEngine and Element console. Current version software is always available at www.AxiaAudio.com/downloads/ .

Pre-installation Hardware Check

Before installing your new panel, we need to perform a quick hardware check.

Open the overbridge panel on your Element and examine the Power Distribution Board (the one in the middle with all the RJ-45 connectors). Does one of the RJ jacks contain an RJ-45 termination device? If so, you are ready to install your accessory panel. Remove the terminator from the distribution board and set it aside for later.

If you do not see this terminator, please contact Axia support for more information.

Installation

User panels that connect via CANBus are much like Element fader modules, in that they require a unique ID setting in order for the system to recognize them correctly. Each Element module (and each CANBus accessory panel) has a multi-position rotary switch located on its circuit board; each device attached to an Element's CANBus must have this switch set to a unique position.

To find out what ID assignments are free, enter Element's diagnostic mode by pressing and holding the **Help** and **Right Arrow** keys on the Monitor Module for five seconds.

You'll see a description of each installed module and accessory panel device; preceding the description are a set of numbers in brackets.

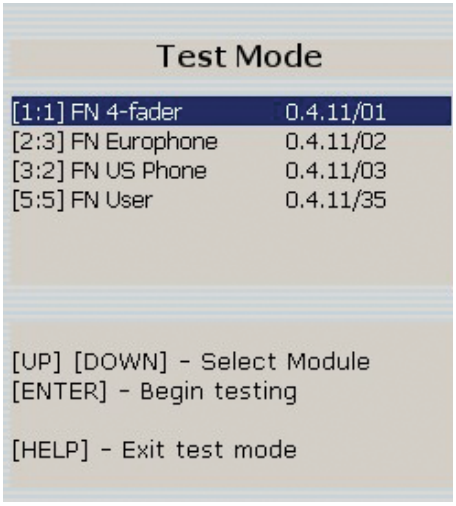


Figure E-2: Element diagnostic mode

©2008 Axia Audio — Rev. 3.0

The bracketed numbers refer to device type and ID number, respectively. The first digit identifies the type of device; the second digit the ID (controlled by the rotary switch). Select an ID for your new accessory panel by determining which IDs are not in use.

Now, using a small screwdriver, turn the switch on the bottom of the accessory panel to the ID number you've selected (as shown in Figure E-3).

Now, connect the panel to the Element CANBus using CAT-5e Shielded cable. Be sure to keep make the cable only as long as is required; 50 feet of cable for a panel located 10 feet from the console is excessive, and can cause problems later!

Shielded category cable is almost never required in an Axia system. However, we highly recommend shielded CAT-5e in this instance in order to prevent the digital control signals present in these circuits from generating RF interference after they exit the containment of the Element chassis. In a pinch, though, regular (unshielded) CAT-5e will work.

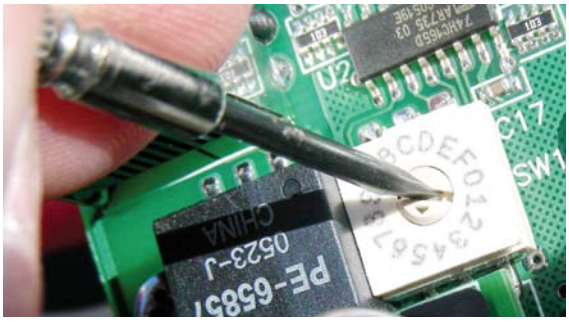


Figure E-3: Setting the ID number

Connect one end of the cable to the accessory panel you're installing, and the other end to the Power Distribution Board, using the jack you removed the termination block from.

If you have more than one accessory panel to install, connect it now to the second RJ port on the back of the first accessory panel. More panels can be daisy-chained as needed.

On the final accessory panel, place the termination block you removed from the Power Distribution Board into the panel's spare RJ jack.

Once connected, your new accessory panel will be automatically recognized and ready to use. In the rare event it is not, please "capture" the console's new configuration, following the instructions in Chapter 1, page 3 of this manual.

Programming

Using a connected computer, log into your Element's configuration web pages and select the User Modules page. You'll see your new panel listed there.

- For panels with mic functions, enter the channel number of the mic source you want it to control.
- For panels with headphone functions, enter the channel number of the mic source that the headphone panel is adjacent to. The panel will control the headphone backfeed to that mic position.

Installing a GPIO Accessory Module

Installation

Accessory panels using GPIO connections are a straightforward installation using standard multiple-conductor cable and DB-15M connectors. You can fabricate this cable, or purchase one like the Cables To Go #02639, a 10-foot version available from Provantage, Best Buy or online from CablesToGo.com.

Figure E-4: Axia rackount control panels connect via GPIO ports

Cables between the accessory panels and the Axia GPIO Node pin-for-pin, with male-to-male connectors.

Programming a Mic Control Panel connected to an Element GPIO Connection

This section applies when connecting a Mic Control accessory panel to the GPIO port of your Element Power Supply. It assumes you have already made the cable con-

©2008 Axia Audio — Rev. 3.0

nection described above in “Installation.”

Using a PC connected to the Axia network, enter the configuration page for your Element and select the GPIO Configuration section.

In the GPIO Configuration page, you will see eight listings that correspond to the eight ports that are on the back of the Element Power Supply. Create a name on the listing that corresponds to the port to which you wired your Mic Control Panel, and enter the channel number of the mic source you want it to control.

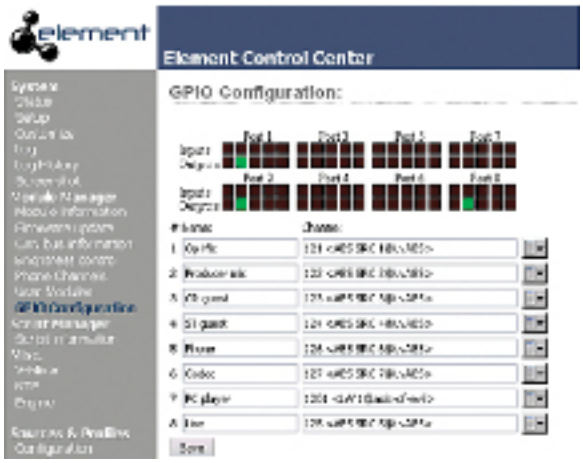


Figure E-5: GPIO status indications

Now, assign the Mic channel you’re working with to one of the channels of your Element. Navigate to the the Element GPIO Configuration page as shown in Figure E-5 above and observe the status indicators for the lights and input commands on your mic controller as you toggle the accessory panel from **On** to **Off**. Refer to the table in Figure E-6 for the Mic GPIO profile.

Programming a Mic Control Panel connected to a GPIO Node Connection

This section applies when connecting a Mic Control accessory panel to the GPIO port of an Axia GPIO Node. It assumes you have already made the cable connection described above in “Installation.”

Using a PC connected to the Axia network, enter the configuration page for the GPIO Node to which your accessory panel is connected.

Create a name on the listing that corresponds to the port to which you wired your Mic Control Panel, and enter the channel number of the mic source you want it to control.

Now, assign the Mic channel you’re working with to one of the channels of your Element. Navigate to the GPIO Node’s Configuration page and observe the status indicators for the lights and input commands on your mic controller as you toggle the accessory panel from **On** to **Off**. Refer to the table in Figure E-6 for the Mic GPIO profile.

Special Programming for Mic Control Panels

It is possible to connect the mic controller to a GPIO port either on a GPIO Node or on the Element Power Supply and program it using PathfinderPC logic control. For more information on this, please refer to the PathfinderPC User’s Manual.

Inputs	Pin	Outputs	Pin	Power/Common	Pin
ON Command	11	ON Lamp	1	Source Common	7
OFF Command	12	OFF Lamp	2	Logic Common	8
TALK Command	13	TALK Lamp	3	+5 VDC	9
MUTE Command	14	MUTE Lamp	4	Input Common	10
not used	15	*not used*	5	*not used*	6

Figure E-6 : GPIO pinout diagram for Mic profile

©2008 Axia Audio — Rev. 3.0

Meeting coffee, black.

Less Powerpoint, I beg you.

Must keep eyes open.

©2008 Axia Audio — Rev. 3.0

Appendix F: Channel / IP Worksheets

The scalable nature of Axia Livewire audio networks makes it possible to construct systems of any size - from a single room to an entire multi-studio facility.

Since Livewire components – Audio Nodes, Elements, StudioEngines, etc. – are connected with Ethernet, each one requires a unique IP (Internet Protocol) address. IP addresses are four bytes long and are written in "dotted decimal" form, with each byte represented decimally and separated by a period. For example, in the IP address 193.32.216.9, the 193 is the value for the first byte, 32 for the second, etc. Since a byte can hold values from 0 to 255, this is the range for each decimal value. If you run a public network, Host IP addresses are assigned to your organization by your internet service provider and parceled out to individual host computers by your network administrator. He may give you this number to be entered manually, or could opt for DHCP (Dynamic Host Configuration Protocol) to let your computer get the address automatically from a pool. Because Livewire devices are permanently attached and because it is more desirable to know the IP address attached to a particular node (and perhaps assign them in some kind of logical pattern), we do not support DHCP for our hardware nodes. Therefore, you will need to enter an IP address into each node.

Using the audio nodes, Livewire systems can support 32,766 channels of audio; this necessitates giving each audio source or destination its own unique identifier, called *Channel ID numbers.*

As you can imagine, keeping track of so many unique IP addresses and Channel ID numbers is a big job. Therefore, on the next few pages, we've provided you with worksheets that you can use to keep track of the numbers you've assigned.

For a more detailed explanation of IP addresses, subnet masks and related issues, we highly suggest that you refer to "Network Engineering For Audio Engineers", Chapter 8 of our *Introduction to Livewire: System Design Reference and Primer.*

©2008 Axia Audio — Rev. 3.0

Axia IP Address Assignment Worksheet

IP Address	*Device Name*	*Location*	*Notes*
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			
. . .			

©2008 Axia Audio — Rev. 3.0

Axia Audio Channel Assignment Worksheet - SOURCES

Channel ID Number	*Device Name*	*Device Location*	*Notes*

©2008 Axia Audio — Rev. 3.0

Axia Audio Channel Assignment Worksheet - DESTINATIONS

Channel ID Number	*Device Name*	*Device Location*	*Notes*

Appendix G: CE Declaration of Conformity

Supplementary Information for European Users

CE

Declaration of Conformity

Application of Council Directives: 93/68/EEC (CE Marking); 73/23/EEC (safety low-voltage directive); 89/336/EEC (electromagnetic compatibility).

Standards to which conformity is declared: EN50081-1, EN50082-1.

Manufacturer's Name: Axia, TLS Corporation

Manufacturer's Address: 2101 Superior Avenue, Cleveland, Ohio, USA

Type of equipment: Audio Mixing Surface

Model No.: Element

Serial No.: _____

Year of Manufacture: 2005

I, the undersigned, hereby declare that the equipment specified above conforms to the above Directives and Standards.

Place: Cleveland, Ohio, USA 44114

Date: October 30, 2005

Signature: _____(on file)_____

Full Name: Michael Dosch

Position: President

CE Mark Conformance

October 30, 2005
Axia, TLS Corp.
2101 Superior Avenue
Cleveland, OH 44114 USA

Supervision and summary by: _____
Michael Dosch

1.0 Introduction

1.1 Scope This record is intended to document the Axia Element audio mixing surface conformance with Council Directives: 93/68/EEC (CE Marking); 73/23/EEC (safety-low voltage directive); 89/336/EEC (electromagnetic compatibility).

©2008 Axia Audio — Rev. 3.0

CE Mark Conformance, Cont'd

Testing was conducted during the period September 22 - October 19, 2005.

1.2 Purpose Testing was performed to evaluate safety, emissions and immunity, conformance to enable CE marking.

1.3 Conclusions The Element audio control surface was found to be compliant and CE marking was applied to the product.

Conformity is declared to the standards: EN50081-1, EN50082-1.

2.0 HF Radiated Emissions

2.1 Standards Applied The unit was laboratory evaluated according to standard: EN50081-1 using Measurement Document: EN55022, Class B.

2.2 Testing The unit was tested and evaluated by: Smith Electronics*

2.3 Test Results The unit was found to be in conformance for both radiated and line-conducted emissions.

3.0 Immunity

3.1 Standards Applied

3.1.1 ESD Susceptibility Measurement Document: IEC 1000-4-2 (IEC 801-2). Immunity to static electricity.

3.1.2 Radiated Susceptibility - HF Electric Field Document: IEC 1000-4-3 (IEC 801-3).

3.1.3 Conducted Susceptibility - Power Line Measurement Documents: Burst/Transients: IEC 1000-4-4 (IEC 801-4), Surge: IEC 1000-4-5 (IEC 801-5) Voltage Fluctuations: IEC 1000-4-1 (IEC 801-1)

3.2 Testing

3.2.1 ESD Immunity The unit was tested and evaluated by: Smith Electronics*

3.2.2 Radiated Immunity - RF Electric Field The unit was tested and evaluated by: Smith Electronics*

3.2.3 Conducted Immunity - Power Line Testing to IEC 801-4 was conducted by: Smith Electronics*

*Smith Electronics
8200 Snowville Rd.
Cleveland, OH 44141 (USA)

CE

CE Conformance

This device complies with the requirements of EEC Council Directives 93/68/EEC (CE Marking); 73/23/EEC (safety low-voltage directive); 89/336/EEC (electromagnetic compatibility). Conformity is declared to the following standards:: EN50081-1, EN50082-1.

©2008 Axia Audio — Rev. 3.0

Warranty

Axia Audio Limited Warranty

This Warranty covers “the Products,” which are defined as the various audio equipment, parts, software and accessories manufactured, sold and/or distributed by TLS Corp., d/b/a Axia Audio (hereinafter “Axia Audio”).

With the exception of software-only items, the Products are warranted to be free from defects in material and workmanship for a period of one year from the date of receipt by the end-user. Software-only items are warranted to be free from defects in material and workmanship for a period of 90 days from the date of receipt by the end-user.

This warranty is void if the Product is subject to Acts of God, including (without limitation) lightning; improper installation or misuse, including (without limitation) the failure to use telephone and power line surge protection devices; accident; neglect or damage.

EXCEPT FOR THE ABOVE-STATED WARRANTY, AXIA AUDIO MAKES NO WARRANTIES, EXPRESS OR IMPLIED (INCLUDING IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE).

In no event will Axia Audio, its employees, agents or authorized dealers be liable for incidental or consequential damages, or for loss, damage, or expense directly or indirectly arising from the use of any Product or the inability to use any Product either separately or in combination with other equipment or materials, or from any other cause.

In order to invoke this Warranty, notice of a warranty claim must be received by Axia Audio within the above-stated warranty period and warranty coverage must be authorized by Axia Audio. If Axia Audio authorizes the performance of warranty service, the defective Product must be delivered, shipping prepaid, to: Axia Audio, 2101 Superior Avenue, Cleveland, Ohio 44114.

Axia Audio at its option will either repair or replace the Product and such action shall be the full extent of Axia Audio’s obligation under this Warranty. After the Product is repaired or replaced, Axia Audio will return it to the party that sent the Product and Axia Audio will pay for the cost of shipping.

Axia Audio’s authorized dealers are not authorized to assume for Axia Audio any additional obligations or liabilities in connection with the dealers’ sale of the Products.

Axia Audio’s products are to be used with registered protective interface devices which satisfy regulatory requirements in their country of use.

©2008 Axia Audio — Rev. 3.0

TCP/IP,

How did we live without thee?

Analogicly.

©2008 Axia Audio — Rev. 3.0

AXIA
A TELOS COMPANY