

SmartSurface

Installation & User's Guide



Revision 1.07 for software versions
1.49 and higher — November, 2005

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.”

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 use convection cooling. Do not block the ventilation openings in the top and rear of the unit. SmartEngine and the PowerSupply/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 side 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 Axia 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 SmartSurface 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, SmartSurface, SmartQ, 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

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

Copyright © 2005 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.

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

Many thanks to Bruce Wilkinson of Pippin Technical Services and Telos' Rolf Taylor, without whose helpful suggestions and keen eyes this manual would not exist.

About This Manual

This manual covers the details of the SmartSurface 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.

This is being written in January 2005, shortly after the release of Livewire. Everything here is new and fresh. There will no doubt be many updates to this manual over the coming months and years.

As always, we welcome your suggestions for improvement. Contact Axia Audio with your comments:

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

Table of Contents

<i>Customer Service</i>	iii
<i>Warranty</i>	iv
<i>Service</i>	iv
<i>Credit Where Credit's Due</i>	iv
<i>About This Manual</i>	iv
<i>A Note From The Founder/CEO of Telos</i>	vii
<i>A Note From The President of Axia</i>	viii
Quickstart	ix
Making Connections	ix
Software Setup	ix
Chapter One: Setup and Connections	1
Introduction	1
SmartSurface Preliminary Setup	2
StudioEngine Preliminary Setup	3
Additional SmartSurface Settings	5
Chapter Two: Configuring Inputs	7
Understanding Source Profiles	7
Source Profile Setup	7
<i>Source Profile Options</i>	8
<i>Adding Backfeeds and GPIO to a Source Profile</i>	11
Chapter Three: Configuring GPIO	13
GPIO Port Definitions	13
<i>GPIO Control Room Guest Microphone Logic</i>	14
<i>GPIO Studio Guest Microphone Logic</i>	15
<i>GPIO Producer's Microphone Logic</i>	16
<i>GPIO Line Input Logic</i>	17
<i>GPIO Codec Logic</i>	18
<i>GPIO Telephone Hybrid Logic</i>	19
<i>GPIO Control Room Monitor Logic</i>	20
<i>GPIO Studio Monitor Logic</i>	21
<i>GPIO Computer Playback Device Logic</i>	22
<i>GPIO Recording Device Logic</i>	23
Assigning GPIO to a Source	24
Chapter Four: SmartSurface Operations	27
Overview	27
<i>Displays</i>	27
<i>Show Profiles</i>	27
<i>Sources and Channels</i>	27

<i>Mix-Minus</i>	28
<i>GPIO</i>	28
<i>Soft Keys and Scrub Wheel</i>	28
<i>Software</i>	28
Control Callout	28
Control Details	30
<i>General Channel Controls</i>	30
<i>Channel Status Display</i>	30
<i>PGM-1 & PGM-2 Keys</i>	30
<i>Record Key</i>	30
<i>Phone Key</i>	30
<i>Channel Options Key</i>	30
<i>Channel Preview Key</i>	34
<i>Fader</i>	34
<i>On and Off Keys</i>	34
<i>Source-Specific Channel Controls</i>	35
<i>Operator Microphone Channel Operation</i>	35
<i>Control Room Guest Microphone Channel Operation</i>	36
<i>Control Room Producer Microphone Channel Operation</i>	37
<i>Studio Guest Microphone Channel Operation</i>	38
<i>Line Channel Operation</i>	39
<i>Phone Channel Operation</i>	39
<i>Codec Channel Operation</i>	41
<i>Monitor Section Controls</i>	42
<i>Talk To... Keys</i>	42
<i>Monitor Volume / Selection Knobs</i>	42
<i>Monitor Options Key</i>	44
<i>Studio Options Key</i>	46
<i>Transport and Other Controls</i>	46
<i>Transport Keys</i>	46
<i>Soft Keys</i>	46
<i>Record Mode Key</i>	47
<i>Control Options Key</i>	47
<i>Timer Mode Key</i>	49
<i>Scrub Wheel</i>	50
<i>Status Symbol Displays</i>	50
Chapter Five: Show Profiles	51
Creating A Show Profile	51
<i>Build A Show</i>	51
<i>Capture It!</i>	52
Show Profile Options	52
<i>The Channel Description Screen</i>	53

<i>The Auxiliary Send Description Screen</i>	54
<i>Monitor Section Data</i>	55
<i>General Monitor Options</i>	56
<i>Control Room Monitor Options</i>	56
<i>Control Room Headphone Options</i>	57
<i>Studio Monitor Options</i>	57
<i>Save and Exit</i>	58
Record Mode	58
<i>Record Mode Configuration</i>	58
<i>Mixing</i>	58
<i>CR Monitor</i>	58
<i>CR Headphones</i>	59
<i>Studio Monitor</i>	59
<i>Save and Exit:</i>	59
<i>Cutout Dimensions, In Inches</i>	61
<i>Cutout Dimensions, In Centimeters</i>	62
Appendix A: Table of Inputs and Outputs	63
Appendix B: Block Diagrams	65
<i>Operator / Control Room Producer Microphone Channel Block</i>	66
Appendix C: FAQ / Diagnostics / Maintenance	71
<i>Lamp & Display Diagnostics</i>	73
<i>Fader Cleaning Procedures</i>	74
<i>Fader Disassembly and Cleaning</i>	74
<i>Lubricating the Glide Rail</i>	75
<i>Reinstalling The Fader</i>	75
Appendix D: Phone Controller Installation	77
Appendix E: Menu & Screen Reference	79
<i>SmartSurface Operator's Option Menus</i>	79
<i>Web-based Option Menus</i>	80
<i>Web-based Option Menus</i>	81
Appendix F: Channel / IP Worksheets	83
Warranty	87

A Note From The Founder/CEO of Telos

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

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

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

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



add an Ethernet switch and more interfaces to build a system with as many inputs and outputs as you want.

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

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

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

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

Steve Church

A Note From The President of Axia

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.

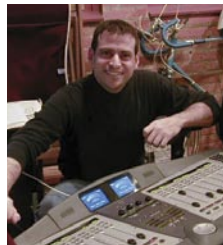
Then a group of Telos engineers developed a method of using Ethernet to network real-time audio devices, allowing computers and consoles, controllers and peripherals to interact smoothly and intelligently. Powerful, flexible networks had finally come to our studios. As with the transition from carts 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...

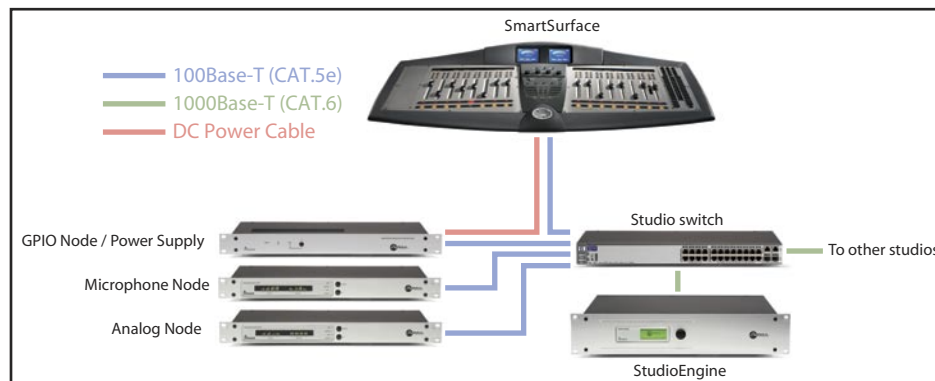
Michael “Catfish” Dosch



Quickstart

The following chapters of this manual will give you an in-depth understanding of the capabilities of your new SmartSurface, 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 sections, but it will help you get everything connected fast, and then point you to the appropriate parts of the manual to get you 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, wondering 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 SmartSurface to a 100Base-T port on the switch.

4. Packed with your SmartSurface is a cable with 5-pin XLR connectors. This is the power cable. Connect one end to the SmartSurface port labelled "DC Input 1". Connect the other end to the matching jack on the back of your GPIO Node/Power Supply unit. Connect the GPIO's Ethernet port to your local switch with CAT.5e or CAT.6.
5. If you've opted for dual redundant power supplies, repeat Step 4 to connect your secondary GPIO Node/Power Supply.
6. Connect all Audio Node power cords and plug them into your studio's power receptacles.
7. Follow the instructions found in Chapter 1 of this manual to configure IP address settings for SmartSurface 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 SmartSurface 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, "SmartSurface 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 SmartSurface! 🎧

World, now digital

Analog memories fade.

The future beckons!

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 all the protective packaging is stripped away; that "new gear" smell... it's exhilarating!

If you're reading this, there's a 99.9794% 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 SmartSurface, its GPIO Node/Power Supply and StudioEngine — and wondering what to do next.

The purpose of this chapter is to provide you with clear, concise instructions to get you up and running. (First, of course, we recommend you read the *Introduction to Livewire: System Design Reference and Primer*, available from www.AxiaAudio.com.) We'll install the SmartSurface first, followed by the StudioEngine.

Be Prepared

Important note: Axia strongly recommends that SmartSurface be the final peripheral installed in any studio.

Because so much of the SmartSurface configuration process depends upon having access to audio sources and destinations previously defined, you **must** connect and configure your other Axia peripherals (Audio Nodes, GPIO Nodes, Router Selectors, etc.) prior to configuring SmartSurface, according to the instructions supplied with each.

To install SmartSurface, prepare a cutout in your studio counter top (a full-size dimensional drawing is included with your SmartSurface for your cabinet provider's use. There's also a scaled-down version with all pertinent measurements in this manual's Appendix). Double-check to ensure adequate clearance for the re-

quired cable connections and grounding stud; the connection panel is in the center rear of the unit.

Before beginning any part of your studio configuration, determine a range of IP addresses to assign to each studio and log each device's IP as you assign it; the appendix also 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 for use with local networks, when it isn't necessary (or even desirable, for security reasons) to use a public IP address.

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 configure SmartSurface, 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, such as Belden MediaTwist or similar high-performance bonded cable if your studio design calls for cable runs through areas containing high levels of RF. (For greater detail on cable selections, read *The Axia Guide to Choosing Category Cable*, 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.

SmartSurface Preliminary Setup

If you haven't done so yet, it's time to get your SmartSurface out of its packing crate.

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

Position one person at each end of SmartSurface 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 SmartSurface into the cutout in your studio counter top. Be mindful of the connector panel located in the center rear of the electronics enclosure; take care not to catch the grounding stud on the counter top edge or damage may occur.

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

Connections and IP Configuration

Connections

Cable connections for SmartSurface are entirely dif-

ferent from any other mixing surface you may be familiar with. There are no audio inputs; SmartSurface needs only power and Ethernet connections to operate.

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.
2. Route another length of CAT. 6 from your SmartSurface to the switch, but do not connect it yet — we'll do that later.
3. SmartSurface **must** be connected to a grounded metal, permanent wiring system or other equipment grounding conductor using the threaded grounding stud located on the connections panel. For ground sources, we recommend, in order of preference:
 - » "Station Ground," the heavy copper strap running through the walls and floors of many radio studios.
 - » 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 SmartSurface.

Important: Your SmartSurface 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

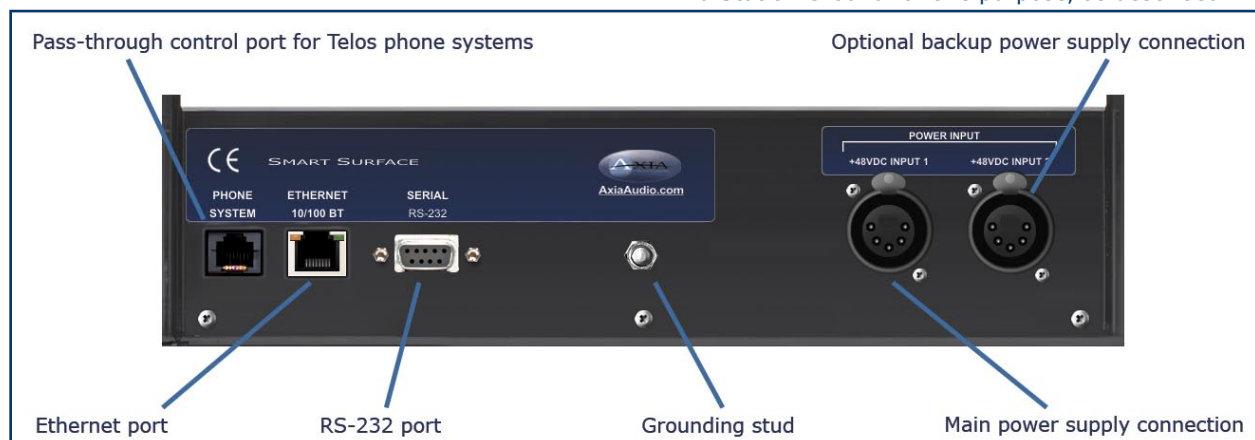


Figure 1-1: SmartSurface Connection Panel

above, use the specific “Safety Ground” your local regulations mandate.

4. Use the 5-pin XLR cable provided to connect the GPIO Node/Power Supply to the SmartSurface connector labeled “DC INPUT 1.” (Note that there are two power inputs to accommodate redundant power supplies should you wish to equip your studio with this option.)
5. Whew, that was hard work. Better have a beer.

IP Address Configuration

Using your IP Assignment Worksheet, you’ll now choose an IP address for both your SmartSurface 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.

Once you’ve determined the IP addresses you’ll assign, power on the SmartSurface power supply.

When SmartSurface has completed its start cycle (meter screens appear), push and hold the **Control Op-**



Figure 1-2: SmartSurface IP configuration screen.

tions key located at the bottom of the center section, then push and hold all three **Talk To** keys located just below the meter screens. You must hold these keys for five seconds to enter the IP Setup screen (Figure 1-2).

Notice that the legend beneath the third Monitor volume control is now blinking **NET CONFIG**. Push the knob once; you’ll see a box appear onscreen around the first value, marked **SmartSurface IP Address**. Push the knob again and the box turns solid; rotate the knob to set the desired value. Push the knob to “take” the new value, then turn it to select the next hex value. Repeat the process until you’re entered the desired IP address, then do the same to set your subnet mask value.

Continue down the screen and set the appropriate values for your network gateway (optional, if you wish to remotely administer your system from offsite), and your chosen NTP (Network Time Protocol) server. The Phone Server IP Address field is reserved for use with Telos broadcast telephone systems with IP connectivity.

Note that the last option on this setup screen allows you to set a password for HTTP administration of SmartSurface; you may enter any string of standard ASCII characters, from 1 – 12 characters in length. If you leave this option undefined, you will not need to enter a password to access SmartSurface options via Web browser.

Connect the Ethernet cable from your SmartSurface to any 100Base-T port on the switch. Proceed to the next section to set up your StudioEngine.

StudioEngine Preliminary Setup

Connections and IP Configuration

Connections

Note: We recommend that you take the same grounding precautions with your new StudioEngine as you did installing your SmartSurface.

We also recommend leaving 1RU of “breathing space” for ventilation above and below the Studio Engine when you rack-mount the unit.

1. Using the supplied AC power cord, connect the SmartEngine to the mains, but do not switch the unit on at this time.
2. Route a length of CAT. 6 cable from SmartEngine to a **Gigabit** (1000Base-T) port on your studio’s Ethernet switch. Do not connect the Ethernet cable to your SmartEngine yet; we’ll do that after we assign its IP address.

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 right to scroll to the tab marked **EID** (“Engine ID”, Figure 1-3), and push to select.

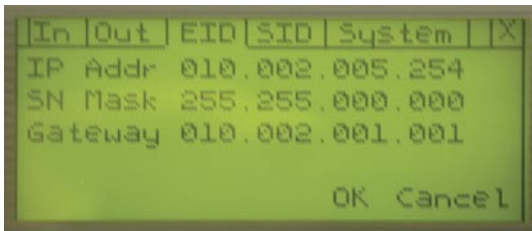


Figure 1-3: StudioEngine IP configuration screen.

3. Push the control knob and you’ll see that the first line of the display, marked **IP Addr** is underlined. Push the knob again to select this line.
4. You’ll see a flashing underline under the first digit 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 value.
6. When you’ve set the subnet mask and gateway IP address (optional), 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**. While your StudioEngine reboots, connect its Ethernet cable.

8. After reboot, navigate to the tab marked **SID**. If your StudioEngine is running software version 2.3.27 or higher, and your SmartSurface is running version 2.4.9 or higher, the IP address of your studio’s SmartSurface will be displayed here. Confirm the IP address, then turn the control knob until the **X** symbol is selected and push.

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 SmartSurface and StudioEngine.

Note: Axia web interfaces are tested for use with Microsoft Internet Explorer, version 5 and later, and Firefox version 1.0.0 and 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.

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: We’ve observed that 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



Figure 1-4: StudioEngine Output configuration

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, we'll need to change those default values to new channel numbers.

1. Open the browser on your computer and point it to the IP address you previously set for your StudioEngine. Choose the **Outputs** option to configure the output busses of the SmartSurface.
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 1-4. This screen allows you to set channel numbers for all of the audio outputs generated by SmartSurface; 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 columns entitled **Livestream** (low delay) and **RTP/IP Standard**. Place check marks in the **Enable:** boxes of each feed.
5. Choose **Apply**.

Additional SmartSurface Settings

There is an additional page of SmartSurface options accessible via your Web browser. You probably won't

need to change these options, but the explanations and default values are listed for your reference.

Enter the IP address of SmartSurface in your browser and navigate to the "General" page. You'll notice that the page is divided into sections; we'll look at each section in order.

General

- **Startup Show Profile:** The dropdown box lists all of your available Show Profiles; you can choose one to be loaded whenever SmartSurface boots. At this point, the box only shows "Default Show Profile" and "Restore Previous State." For now, choose "Default Show Profile."

Note: When you've constructed custom Show Profiles, you can use this dropdown box to specify which saved profile to use when SmartSurface starts. Since radio studios are generally 24-hour operations, the profile you specify here should be the one you wish to load automatically after restart should there be a power outage to your facilities. If you choose "Restore Previous State," the surface will boot the console to exactly the state it was in prior to shutdown.

To learn how to construct Show Profiles, please refer to Chapter 5 of this manual.

- **Source Profile Table Path and Show Profile Table Path:** These two boxes define the directories in which SmartSurface will store Source and Show Profile settings. These parameters should not be altered unless you are directed to do so by Axia Support.

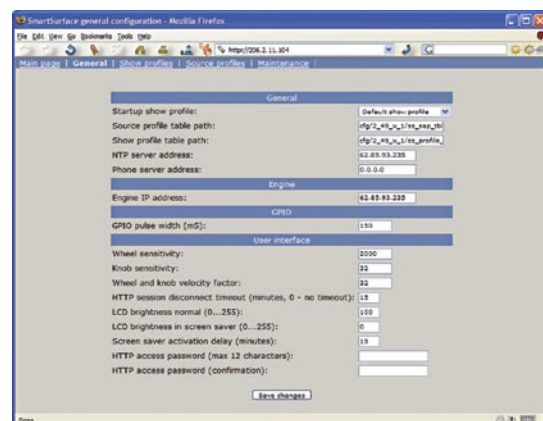


Figure 1-5: SmartSurface General Options page

- **NTP Server Address:** Allows you to enter the IP address of a Network Time Protocol server. You can also set this address from the IP Configurations Screen (as described earlier in this chapter).
- **Phone Server Address:** This field is reserved for use with Telos broadcast telephone systems with IP connectivity.

Engine

- **Engine IP Address:** The IP address of the StudioEngine that this SmartSurface is paired with. You can also set this address from the IP Configurations Screen (as described earlier in this chapter).

GPIO

- **GPIO Pulse Width:** This value determines the width of the logic pulses generated by GPIO Nodes operated by this SmartSurface. The default value is 100 milliseconds; do not alter this value unless instructed to do so by Axia Support.

User Interface

- **Wheel Sensitivity:** Adjusts the sensitivity of the Scrub Wheel found at the bottom of SmartSurface's center section. Default value is 2000; adjusting this to a lower value increases the sensitivity of the wheel, decreasing the number of turns necessary to make an adjustment. Entering a higher value has the opposite effect.
- **Knob Sensitivity:** Works like the Wheel Sensitivity value described above, but affects the four Monitor Section knobs located just beneath SmartSurface's meter screens. Default value is 32.
- **Wheel and Knob Velocity Factor:** Increasing this number cause the Scrub Wheel and Monitor Volume Knobs to effect greater volume changes when spun with greater speed. Default value is 32.
- **HTTP Session Disconnect Timeout:** This value sets the number of minutes SmartSurface's internal webserver will remain connected to a browser without input before the session times out and requires

the user to log in again. 15 minutes is the default value; if set to "0", timeout is disabled and the session will remain open until the browser is closed.

- **LCD Brightness Normal:** Sets brightness of the meter screen backlights during normal operation, i.e., periods of user activity. Default value is 100; value can be adjusted from 0 (lowest brightness) to 255 (full bright).
- **LCD Brightness in Screen Saver:** During periods of inactivity (no user input), SmartSurface will dim the meter screen backlights to help extend their lifespan. This setting determines the brightness of the meter screen when in "screen saver" mode. Default value is 0; value can be adjusted from 0 (lowest brightness) to 255 (full bright).
- **Screen Saver Activation Delay:** The number of minutes without input before SmartSurface's meter screens enter Screen Saver mode. Default value is 15 minutes.
- **HTTP Access Password and HTTP Access Password Confirmation:** Use this field to specify or change a the password required for logging into a SmartSurface web session. By default, these fields appear blank when this screen is loaded.
- **Save Changes:** Click on this button to save all changes made to values on this screen and refresh the screen.

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 SmartSurface concepts: Source Profiles. 🔄

Chapter Two:

Configuring Inputs

In this chapter we'll discuss how to set up inputs for your SmartSurface. We'll be using the HTTP servers built into the SmartSurface and SmartEngine 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 addresses of entire rooms worth of Audio Nodes, then recall them in tabbed groups with a single click.

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.

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?

SmartSurface 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.

To understand how Source Profiles work, think of

a sandwich: the meat is the "payload" and the bread is the "delivery system," and other "associated data" (cheese) comes along for the ride.

In a Livewire network, a Source Profile works much the same way, aggregating all information about a given source into one tidy package, so that the audio (payload) is transported by the network (delivery system), accompanied by control definitions, source type information and PAD (program associated data) which allow SmartSurface to automatically generate and send associated backfeeds and mix-minuses, determine monitor settings and enable remote-control device logic.

The result of all this is that each SmartSurface on your network is always presented with complete, consistent information about how you want a given audio source handled, so that it's treated the same way no matter what studio it's used in.

Note also that with SmartSurface, it is possible to create more than one Source Profile for a single audio source — one with logic enabled, and one without, for instance, or one with preset EQ and one without. This allows you unparalleled flexibility in tailoring operational capabilities to your air talent's needs.

Source Profile Setup

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

Note: You can't configure SmartSurface I/O unless you've already connected inputs and outputs to your studio's Audio Nodes. Make sure you've done this before proceeding through this chapter.

Your computer must be assigned a valid network IP address in order to "see" Livewire web pages. Livewire networks don't contain DHCP servers, so you must manually assign your computer an IP address within the

Note: Even if no GPIO is assigned to Mic sources, they will still mute CR Monitor speakers when their faders are **ON**.

- » **Operator** is the board operator's mic. It is the source mic for SmartSurface'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 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. 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.
 - **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 happens to a fader's **Preview** channel assignment when the surface channel is turned **On**.
 - » **Normal, Auto Switching Disabled:** Any fader assigned to **Preview** when a surface channel is **Off** stays assigned to **Preview** when the channel is turned **On**.
 - » **Channel On Turns Preview Off:** Any fader assigned to **Preview** is de-assigned when the channel is turned **On**.
 - **HYBRID ANSWER MODE:** This setting allows you to tailor SmartSurface's Hybrid Auto Answer mode to suit your facility's operating style.
 - » **Normal, Auto Answer Disabled:** SmartSurface provides no auto-answer logic when a surface channel with a hybrid assigned is turned **On**.
 - » **Channel ON Answers Hybrid:** When a hybrid has an incoming call, turning its assigned surface channel **ON** will answer the call.
 - » **Channel ON or Preview ON Answers Hybrid:** When a hybrid has an incoming call, turning its assigned surface channel **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.
 - **LOGIC PORT ENABLED/DISABLED:** Allows you to enable or disable GPIO machine logic to this audio source device.
 - **FEED TO SOURCE MODE:** If there is a backfeed associated with this source, this option determines what content is fed back from the board.
 - » **PGM1 mix-minus** feeds the Program 1 bus, minus the source.
 - » **PGM2 mix-minus** feeds the Program 2 bus, minus the source.
 - » **PHONE mix-minus** feeds whatever is on the Phone bus, minus the source.
 - » **Auto** feeds the Program 1 bus (minus the source) when the channel is **ON**, and feeds the Phone (minus the source) bus when the channel is **OFF**.
-
- Note:** PGM1 and PGM2 feeds are post-fader; Phone feed is pre-fader and allows speaker-phone-style operation.
-

- **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).
 - » **Sum** creates and L+R (-3dB) 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 the "Channel Options" menu on the SmartSurface.
 - » **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 using the **Channel Options** menu on SmartSurface.
 - **AUTO-START TIMER:** Determines whether the event timer found in SmartSurface'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, or, if Auto-Add mode is being used, timer will resume counting.

About SmartSurface 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 SmartSurface 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 SmartSurface's **Channel Options** controls, and can be employed to trim EQ on-the-fly .

- **EQ STATUS:** Enable or disable 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 knee 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.

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 SmartSurface 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 input #7 of your Analog Node and assigned it a channel number of 77. You’ve just 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 the Source Profile for your hybrid has **Source Type** set to “Phone” **Backward Feed** “Enabled”. Leave **Feed to Source Mode** set to “Auto”.
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 2-2); since our source was on input #7 of this Node, 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.

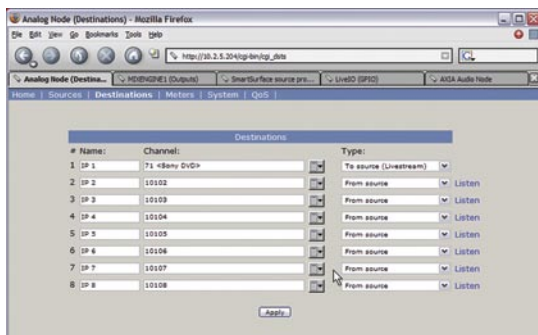


Figure 2-2: Analog Node Destinations screen



Figure 2-3: Picking a source from the list box

A list of available sources will pop up (Figure 2-3). 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 (Livestream)**.

Note: For a detailed description of these options and how they operate, please refer to the *8x8 Analog Node / 8x8 AES Node User’s Manual*.

5. Click **Apply**.
6. Now point your browser to your GPIO Node and choose **GPIO** from the main menu.
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

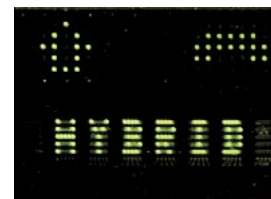


Figure 2-4: Status Symbols; “up” arrow indicates backfeed; “handset” indicates Phone bus.

SmartSurface fader channel and load your new source; you'll see the Status Symbols appear in the window above the channel to indicate that SmartSurface is back-feeding a mix-minus to this telephone hybrid. (Status Symbols are described in detail at the end of Chapter Four, "SmartSurface Operations.")

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

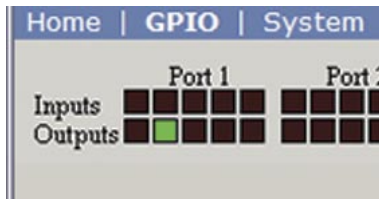


Figure 2-5: GPIO status indicators

SmartSurface channel on and off and you'll observe the active tally change pins.

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. [↻](#)

Chapter Three:

Configuring GPIO

Since the earliest days of simple, rotary-switch source selectors, it's always been a challenge to properly route machine logic along with audio. Some complicated schemes have been devised over the years, but still broadcasters keep asking for 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 fresh 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 on which your studio's audio travels, further reducing infrastructure, cost and tedium.

Along with controlling external audio devices, SmartSurface'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 will explain how to set up a SmartSurface 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: Some GPIO Nodes also contain power supplies for SmartSurface studio control surfaces. We recommend that, when rack-mounting these units, you leave 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 functions such as machine control, lamp drives and remote channel controls.

GPIO ports are programmed to support eight 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. 8). This is important, because when a GPIO port is assigned to a source, the GPIO Node looks at this Source Profile parameter to determine what logic functions the assigned port should provide.

So, if the GPIO “sees” in the Source Profile that the assigned device is a microphone, it provides logic for **On**, **Off**, **Remote Mute** and **Remote Talk** commands. If it “sees” a line input defined in the Source Profile, it provides **Start**, **Stop** and **Reset** commands, plus **Ready** lights, etc.,

Earlier we mentioned that a GPIO port can deliver eight different command sets. These are:

1. Microphone
2. Line Input
3. Codec
4. Telephone Hybrid
5. Control Room Monitor
6. Studio Monitor
7. Computer Playback Device
8. Record Device

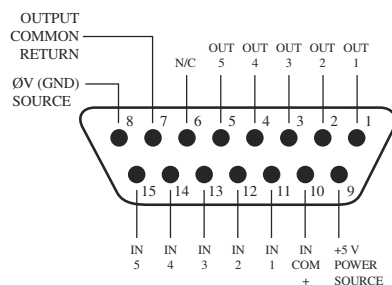
The next few pages contain tables that explain what function the pins provide in each different device mode.



Figure 3-1: Rear panel of GPIO/Power Supply showing DB-15 connectors.

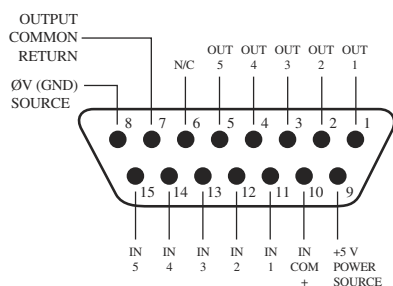
GPIO Control Room Guest Microphone Logic

Name	Pin	Type	Notes
INPUTS			
ON Command	11	Active Low Input	Turns channel ON
OFF Command	12	Active Low Input	Turn channel OFF
Not Used	13	Active Low Input	
MUTE Command	14	Active Low Input	Mutes channel outputs
TALK (To Source) Command <i>(ELEMENT only, not used for SmartSurface)</i>	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 is active
OFF Lamp	2	Open Collector to Logic Common Return	Illuminates when channel is OFF
Not Used	3		
MUTE Lamp	4	Open Collector to Logic Common Return	Illuminates when MUTE is active
TALK (to Source) Lamp <i>(ELEMENT only, not used for SmartSurface)</i>	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 the common of the controlled device. Should be connected to Pin 8 if the node's internal 5 volt supply is used
Logic Common	8	Internal 5 Volt return	Must be tied to Pin 7 if used
+ 5 Volt supply	9	Internal + 5 volt source	Can be used to power external devices or connected to input common (pin 10) for use with dry contact inputs. If used pins 7 & 8 must be connected
Input Common	10	Common for all 5 inputs	Tie external or internal power supply feed positive lead here
NOT CONNECTED	6		



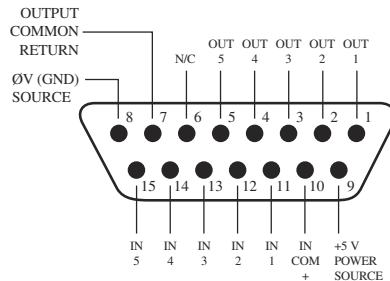
GPIO Studio Guest Microphone Logic

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 <i>(ELEMENT only, not used for SmartSurface)</i>	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 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 <i>(ELEMENT only, not used for SmartSurface)</i>	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 the common of the controlled device. Should be connected to Pin 8 if the node's internal 5 volt supply is used
Logic Common	8	Internal 5 Volt return	Must be tied to Pin 7 if used
+ 5 Volt supply	9	Internal + 5 volt source	Can be used to power external devices or connected to input common (pin 10) for use with dry contact inputs. If used pins 7 & 8 must be connected
Input Common	10	Common for all 5 inputs	Tie external or internal power supply feed positive lead here
NOT CONNECTED	6		



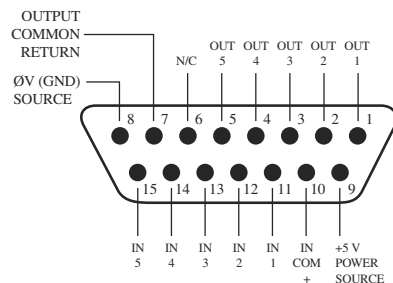
GPIO Producer's Microphone Logic

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 STUDIO/MON2) Command	13	Active Low Input	On SmartSurface, activates TALK TO STUDIO function; on Element, activates TALK TO MON2. 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 STUDIO/MON2) Lamp	3	Open Collector to Logic Common Return	Illuminates when TALK TO STUDIO (SmartSurface) or TALK TO MON2 (Element) 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 the common of the controlled device. Should be connected to Pin 8 if the node's internal 5 volt supply is used
Logic Common	8	Internal 5 Volt return	Must be tied to Pin 7 if used
+ 5 Volt supply	9	Internal + 5 volt source	Can be used to power external devices or connected to input common (pin 10) for use with dry contact inputs. If used pins 7 & 8 must be connected
Input Common	10	Common for all 5 inputs	Tie external or internal power supply feed positive lead here
NOT CONNECTED	6		



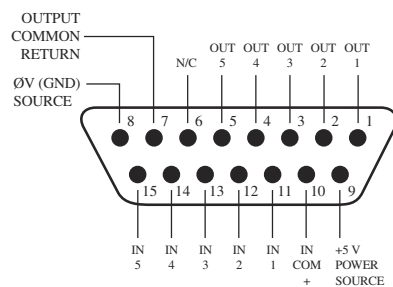
GPIO Line Input Logic

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 the common of the controlled device. Should be connected to Pin 8 if the node's internal 5 volt supply is used
Logic Common	8	Internal 5 Volt return	Must be tied to Pin 7 if used
+ 5 Volt supply	9	Internal + 5 volt source	Can be used to power external devices or connected to input common (pin 10) for use with dry contact inputs. If used pins 7 & 8 must be connected
Input Common	10	Common for all 5 inputs	Tie external or internal power supply feed positive lead here
NOT CONNECTED	6		



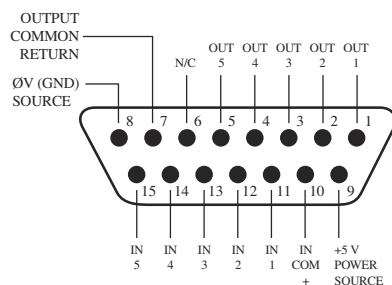
GPIO Codec Logic

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
Not used	15	Active Low Input	
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
TALK 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
Not used	5	Open Collector to Logic Common Return	
POWER & COMMON			
Source Common	7	Logic Common	Connect to the common of the controlled device. Should be connected to Pin 8 if the node's internal 5 volt supply is used
Logic Common	8	Internal 5 Volt return	Must be tied to Pin 7 if used
+ 5 Volt supply	9	Internal + 5 volt source	Can be used to power external devices or connected to input common (pin 10) for use with dry contact inputs. If used pins 7 & 8 must be connected
Input Common	10	Common for all 5 inputs	Tie external or internal power supply feed positive lead here
NOT CONNECTED	6		



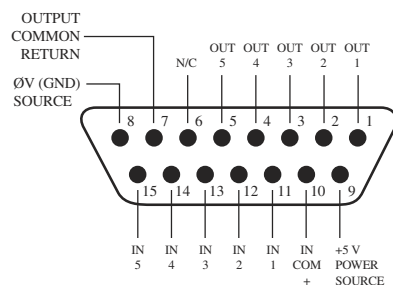
GPIO Telephone Hybrid Logic

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 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 & COMMAND			
Source Common	7	Logic Common	Connect to the common of the controlled device. Should be connected to Pin 8 if the node's internal 5 volt supply is used
Logic Common	8	Internal 5 Volt return	Must be tied to Pin 7 if used
+ 5 Volt supply	9	Internal + 5 volt source	Can be used to power external devices or connected to input common (pin 10) for use with dry contact inputs. If used pins 7 & 8 must be connected
Input Common	10	Common for all 5 inputs	Tie external or internal power supply feed positive lead here
NOT CONNECTED	6		



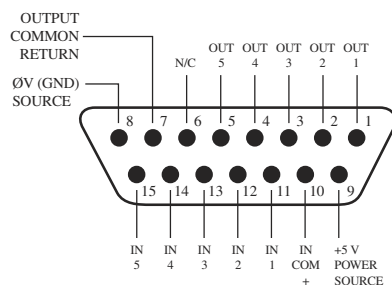
GPIO Control Room Monitor Logic

Name	Pin	Type	Notes
INPUTS			
MUTE CR Command	11	Active Low Input	Mutes CR monitor and Preview speakers
DIM CR Command	12	Active Low Input	Remotely DIMS CR monitor speakers
Enable EXT PREVIEW Command	13	Active Low Input	Feeds External Audio Input to PREVIEW
Not Used	14	Active Low Input	
Not Used	15	Active Low Input	
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
EXT PREVIEW Lamp	3	Open Collector to Logic Common Return	Illuminates whenever EXT PREVIEW is enabled
TALK TO EXTERNAL active lamp	4	Open Collector to Logic Common Return	Utility lamp feed. Illuminates when TALK TO EXTERNAL keys are pressed.
TALK (to CR) 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 the common of the controlled device. Should be connected to Pin 8 if the node's internal 5 volt supply is used.
Logic Common	8	Internal 5 Volt return	Must be tied to Pin 7 if used.
+ 5 Volt supply	9	Internal + 5 volt source	Can be used to power external devices or connected to input common (pin 10) for use with dry contact inputs. If used pins 7 & 8 must be connected.
Input Common	10	Common for all 5 inputs	Tie external or internal power supply feed positive lead here.
NOT CONNECTED	6		



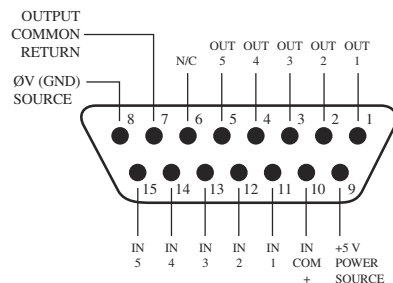
GPIO Studio Monitor Logic

Name	Pin	Type	Notes
INPUTS			
MUTE Studio Command	11	Active Low Input	Mutes Studio monitor speakers
DIM Studio Command	12	Active Low Input	Remotely DIMS studio monitor speakers
Timer Trigger Command	13	Active Low Input	Resets event timer to zero and starts timer.
Remote Countdown Timer Trigger Command (Element only, not applicable for SmartSurface.)	14	Active Low Input	Resets Countdown timer to preset max time and starts countdown timer.
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 Lamp	2	Open Collector to Logic Common Return	Illuminates whenever studio monitors are DIMMED
Timer Trigger Output	3	Open Collector to Logic Common Return	Sends pulse whenever event timer is started from zero.
Countdown Timer Trigger Output (Element only, not applicable for SmartSurface.)	4	Open Collector to Logic Common Return	A 100 mS PULSE sent when countdown timer is started from preset max
TALK TO STUDIO Lamp	5	Open Collector to Logic Common Return	Active whenever a source has activated its TALK TO STUDIO function
POWER & COMMON			
Source Common	7	Logic Common	Connect to the common of the controlled device. Should be connected to Pin 8 if the node's internal 5 volt supply is used.
Logic Common	8	Internal 5 Volt return	Must be tied to Pin 7 if used.
+ 5 Volt supply	9	Internal + 5 volt source	Can be used to power external devices or connected to input common (pin 10) for use with dry contact inputs. If used pins 7 & 8 must be connected.
Input Common	10	Common for all 5 inputs	Tie external or internal power supply feed positive lead here.
NOT CONNECTED	6		



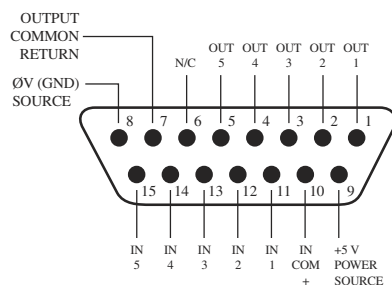
GPIO Computer Playback Device Logic

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 msec pulse when the ON button is depressed, except when the channel is already 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 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 the common of the controlled device. Should be connected to Pin 8 if the node's internal 5 volt supply is used
Logic Common	8	Internal 5 Volt return	Must be tied to Pin 7 if used
+ 5 Volt supply	9	Internal + 5 volt source	Can be used to power external devices or connected to input common (pin 10) for use with dry contact inputs. If used pins 7 & 8 must be connected
Input Common	10	Common for all 5 inputs	Tie external or internal power supply feed positive lead here
NOT CONNECTED	6		



GPIO Recording Device Logic

Name	Pin	Type	Notes
INPUTS			
RWD Indicator	11	Active Low Input	Recorder RWD status output to illuminate remote lamp
FFWD Indicator	12	Active Low Input	Recorder FFWD status output to illuminate remote lamp
RECORD Indicator	13	Active Low Input	Recorder RECORD status output to illuminate remote lamp
PLAY Indicator	14	Active Low Input	Recorder PLAY status output to illuminate remote lamp
STOP Indicator	15	Active Low Input	Recorder STOP status output to illuminate remote lamp
OUTPUTS			
RWD Pulse	1	Open Collector to Logic Common Return	A 100 msec pulse when the remote RWD button is pressed
FFWD Pulse	2	Open Collector to Logic Common Return	A 100 msec pulse when the remote FFWD button is pressed
RECORD Pulse	3	Open Collector to Logic Common Return	A 100 msec pulse when the remote RECORD button is pressed
START Pulse	4	Open Collector to Logic Common Return	A 100 msec pulse when the channel status changes from OFF to ON or the remote PLAY button is pressed
STOP Pulse	5	Open Collector to Logic Common Return	A 100 msec pulse when the channel status changes from ON to OFF or the remote STOP button is pressed
POWER & COMMON			
Source Common	7	Logic Common	Connect to the common of the controlled device. Should be connected to Pin 8 if the node's internal 5 volt supply is used
Logic Common	8	Internal 5 Volt return	Must be tied to Pin 7 if used
+ 5 Volt supply	9	Internal + 5 volt source	Can be used to power external devices or connected to input common (pin 10) for use with dry contact inputs. If used pins 7 & 8 must be connected
Input Common	10	Common for all 5 inputs	Tie external or internal power supply feed positive lead here
NOT CONNECTED	6		



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 physical audio source, all that’s left 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 an audio source? It’s very easy; let’s do it step-by-step.

Note: This procedure assumes that you have set up at least one audio source according to instructions from your Axia Audio Node manual.

1. Open your Web browser and enter the IP address of a GPIO Node connected to your system.

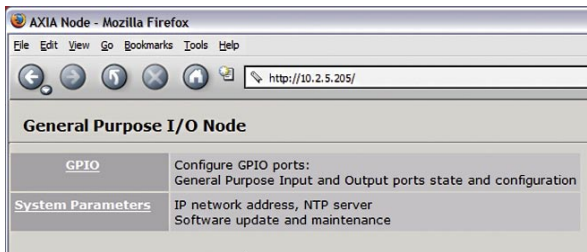


Figure 3-2: GPIO Main Menu

2. When the GPIO main menu opens, click **GPIO** to open the GPIO definitions page. You will receive a login prompt (the default login is “user”; leave the password field blank).

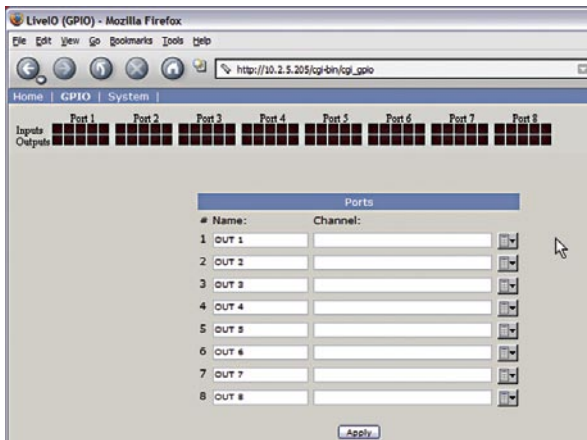


Figure 3-3: GPIO definitions page

3. If you haven’t previously assigned any GPIO ports, the GPIO definitions screen will be blank, as shown in Figure 3-3. 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.

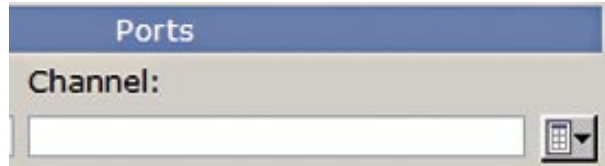


Figure 3-4: Channel box and list icon.

4. When you click on any list icon, a small popup window will open (Figure 3-5), enumerating all of the audio sources available on the Livewire network. 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.

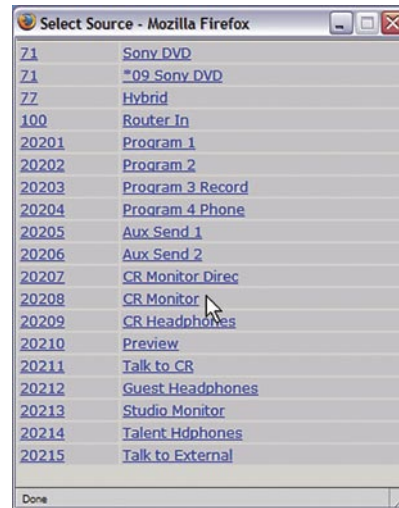


Figure 3-5: GPIO Select Source popup list

5. Type a descriptive name in the **Name** field, and click on the **Apply** button at the bottom of the page.



Figure 3-4: Name and Channel box filled.

6. Assign the source for which you just created a GPIO

link to a SmartSurface channel; operate the **ON** and **OFF** keys for the channel and watch the pin status indicators for Port 1 change as you do so.

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 3-5.

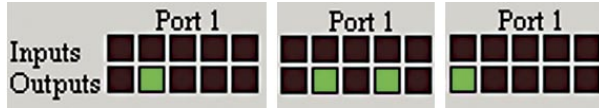


Figure 3-5: Pin status indicators showing GPIO port activity

Referring to the Hybrid Logic Chart on Page 19, we can see that when the SmartSurface 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 once again become active.

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

What's Next

When you're ready, join us in Chapter Four for a comprehensive walk-through of SmartSurface operations. [↻](#)

Arrive at work: late.

Wait — boss is on vacation!

Much loud rejoicing.

Chapter Four:

SmartSurface Operations

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

SmartSurface isn't a console; it doesn't actually mix or process audio. It would be more accurate to say that SmartSurface 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 SmartSurface, 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, SmartSurface brings a whole new level of control and sophistication to the broadcast studio, while providing very intuitive operation.

Overview

SmartSurface 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 various 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. One of the most powerful tools on the SmartSurface is the "Soft Key" section in the middle of the board. These context-sensitive buttons and associated text display change dynamically, following the operator through various tasks.

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

Displays

Dual active matrix TFT (thin-film transistor) displays constantly give operators status and configuration information. These displays normally show meter levels for PGM-1, PGM-2, the time of day clock and the event timer, as well as important status messages. When the operator adjusts console options, the right screen is used to help navigate the options, which are mirrored on the Soft Keys for fast access to the presented functions.

Show Profiles

SmartSurface can be completely reconfigured, instantly, to suit different types of shows. By using predefined Show Profiles, talent can change board settings from a phone and microphone intensive morning show to a personality-based music show at the touch of a button.

Show Profiles are easily accessed through software by pressing the **Control Options** button above the scrub wheel, the Soft Keys then allow instant access to stored Show Profiles and other settings. Construction and administration of Show Profiles is covered in Chapter Five. "Show Profiles."

Sources and Channels

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

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

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

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

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

Mix-Minus

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

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

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

SmartSurface can accommodate up to 16 phone and/or codec inputs at a time, each with its own automated mix minus feed.

GPIO

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

terfaces are connected to SmartSurface 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 SmartSurface as well. *Start/stop* messages can also be passed, via Ethernet, to a computer-based playout system.

Soft Keys and Scrub Wheel

The Soft Keys in the center section, with their alphanumeric displays, work in conjunction with the display screens to give board ops quick access to menu functions. The scrub wheel can be used to control monitor or headphone volume.

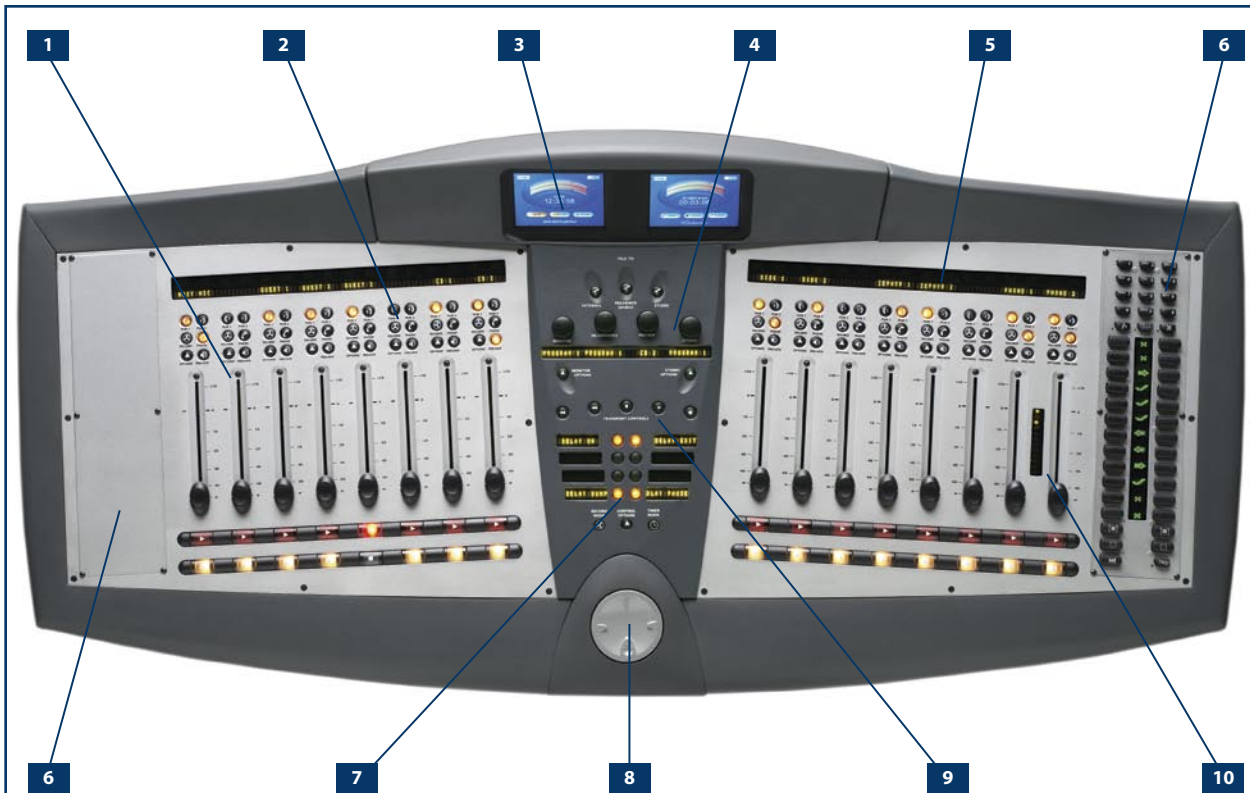
Software

The SmartSurface operating system uses a high-performance embedded Linux kernel to help ensure bullet-proof, 24/7 operation. Software updates, when available, can be obtained from Axia Audio via the Internet and applied by the station engineer.

Updating system software will not affect Show Profiles or other configuration data already programmed into the system.

Control Callout

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



1. **Surface Channels** can accommodate any input source. Each channel's **Options** button calls up expanded channel features.
2. **Assignment Buttons** for the four stereo main output buses – **Program-1**, **Program-2**, **Record** and **Phone** – are located at the top of each channel fader. **Record** and **Phone** buttons let talent quickly set up to record phone bits or send custom mixes to phones and codecs.
3. **Displays.** Default screens show metering for Program buses, clock, event timer, and status annunciator for key functions. When setting optional functions, the right display becomes a navigator.
4. **Monitor Section** with individual volume controls for Control Room and Studio monitor speakers, Preview (cue) and Headphones, Pushing a volume control lets you change monitor sources. Monitor options include “split” headphone operation, Preview in Headphone and more. These option settings are included in the Show Profiles, so that talent always gets the right preferences when loading a show.

5. **Channel Displays** contain a 10-character text readout and two Status Symbols icon displays. 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.
6. **Expansion bays** are for optional panels, such as the control panel for a Telos phone system shown installed in the right bay.
7. **Soft Keys** are context-sensitive and have 10-character alpha displays. When an **Options** key is pressed on a channel, the Soft Keys display the menu of options available for that channel.
8. **Scrub Wheel** can be programmed to change monitor or headphone volume, based on talent's personal preference.
9. **Transport Keys** can control a connected recording or playback device.
10. **Status Symbols** icons between Channels 15 and 16 mirror those on the phone controller when a Telos Series 2101 or TWOx12 broadcast telephone system is in use.

Control Details

General Channel Controls

SmartSurface has 16 individual channels to which audio sources can be assigned at will. Some fader functions change their behavior depending upon the type of source assigned; we'll cover these specific functions in the next section. This section will familiarize you with the controls that are common to all channels.

Channel Status Display

The 10 character alphanumeric display shows the operator which source is presently assigned to this channel. If the **Options** button is pressed and the operator is changing sources, this display will change accordingly.

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

Just above the alpha display is a Status Symbol display, which tells the operator when a mix-minus is being generated for a given channel. The Status Symbols remain dark when no mix-minus is present, but when a mix-minus has been selected the display will show an arrow and a **PGM-1**, **PGM-2** or **Phone** icon to tell the board operator what is being fed to the caller. During a **Talk To Previewed Source** operation, the icon will display the word **HOST**, indicating that the board operator has interrupted the feed.

PGM-1 & PGM-2 Keys

Each channel is assigned to either of two main stereo output buses by selecting either (or both) of these two program keys. Generally, **PGM-1** is the main air bus and **PGM-2** is used for production and special programming requirements. Both program outputs are post-fader and post on-off function.

Record Key

A third stereo bus is provided especially for recording and dubbing. Channels assigned to this bus will be mixed and fed to a special output, intended to feed a recorder/editor. The record bus is fed after the fader, but is independent of on/off status — the **Record** bus is fed whether the channel is on or off. This makes recording off-air phone calls or even producing commercials off air very easy; the operator only has to push the **Record** key on any channel whose source they wish to record, and activate their recording device.

Phone Key

A fourth stereo bus, the **Phone** bus, determines what phone callers will hear when they are not on the air. The console can accommodate up to 16 callers at a time, and will automatically generate unique mix-minus feeds to all callers.

In “Auto” mix-minus mode (the most common mode of operation - see p.9), when the caller's channel is **ON** he will hear the output of the **PGM-1** bus, minus himself. When the channel is **OFF**, he will hear *anything assigned to the Phone bus, less himself*. The phone bus receives pre-fader audio, independent of on/off status.

Channel Options Key

Pressing this key displays the **Channel Options** screen shown in Figure 4-2.

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

For example, if you need to adjust **Aux Sends** on several channels, select the first channel using the **Options** key and navigate to the **Aux Sends** screen. Once the desired settings are made, simply press the **Options** key on the next desired channel; the **Aux Sends** screen will remain active, but you're now adjusting settings for the new channel. The channel number is displayed onscreen as confirmation.

Placing a channel in **Options** mode lets you change inputs, apply EQ, change pan position, invert phase, and



Figure 4-1:
Fader Channel

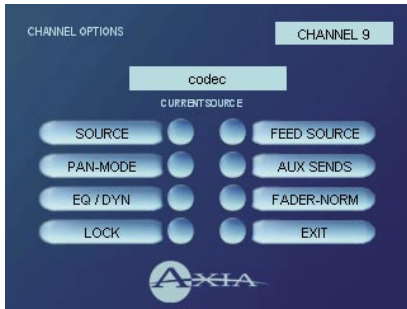


Figure 4-2: Channel Options screen.

set send and return levels for that channel. The **Options** display also shows detailed channel configuration information. Both the SmartSurface channel number (the physical fader position, from left to right) and the source assigned to that channel are displayed for confirmation.

- **Source** allows the operator to choose a new source to assign to the selected channel. Pressing the upper left Soft Key (marked **SOURCE**) will cause the right-hand screen to display the Source Selection menu (Figure 4-3).

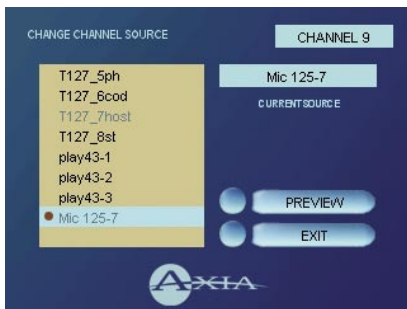


Figure 4-3: Source Selection menu screen.

- The left side of the Source Selection screen displays a list of available sources. There are two ways to scroll this list and “take” a new source:
 - » The Studio Monitor Volume Control knob (3rd from the left) now flashes **SOURCE ##** (## indicates the number of the channel you’re working with). You can turn this knob to scroll through the source list; when your selection is highlighted, push the knob to “take” the new source.
 - » The top three left-hand Soft Keys displays (from top) **UP**, **DOWN** and **SELECT**. Use the **UP** and **DOWN** keys to navigate the list; **SELECT** “takes” the highlighted source.

Note: When you assign a new channel source, SmartSurface remembers the last assigned source. In the Source Selection menu, the lower-left Soft Key is labeled **PREVIOUS**; this functions as a “speed key,” highlighting the previously assigned source in the source list. This is convenient when talent wishes to load a source for a short on-air segment, then return to the previous source when the segment is over.

- » **Preview** allows you to hear the highlighted source through the Preview speakers prior to selecting it. Press and hold the lower-middle right Soft Key labeled **PREVIEW** to listen; release to stop.
- » **Exit** (the bottom right Soft Key) returns you to the meter screen.

- **Pan/Mode** lets the operator adjust the pan/balance of the channel source and correct signal phase errors. Pressing the upper-middle left Soft Key (marked **PAN/MODE**) causes the right-hand screen to display the Pan-Mode Menu (Figure 4-4).

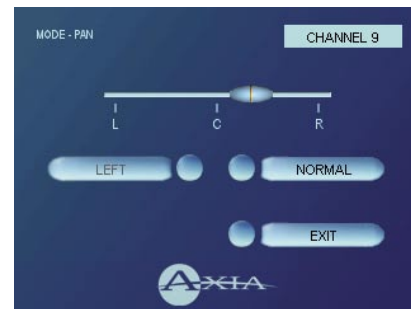


Figure 4-4: Adjusting pan/balance.

- » The alpha display beneath the Studio Monitor Volume Control knob now reads **PAN/BAL**; the board op can use this to pan a mono source or adjust the balance of a stereo source.
- » **Stereo/Left/Right/Sum:** If the selected source is in mono, selecting this option produces no change. 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), this control lets the board op switch source audio between **Stereo** (discrete left/right), **Left** (left source channel fed to L/R input), **Right** (right source channel fed to L/R input) and **Sum** (left and right source

channels summed to mono and fed to L/R input). If available, this command appears on the left upper-middle Soft Key.

- » **Normal/Inv Left/Inv Right/Invert:** if the selected source is in stereo, this control lets the board op compensate for out-of-phase audio sources. **Normal** is used for in-phase sources; **Inv Left** inverts the phase of the left input channel only, **Inv Right** inverts the phase of the right input channel only, and **Invert** reverses phase of both stereo channels. If the selected source is in mono, the choice are **Normal** and **Invert** only. If available, this command appears on the right upper-middle Soft Key.
 - » **Exit** (the bottom right Soft Key) returns you to the meter screen.
- **EQ/Dyn** allows three-band parametric equalization to be applied to the selected audio source. Pressing the left lower-middle Soft Key, labeled **EQ/DYN**, changes the right-hand screen to show the Channel EQ screen (Figure 4-5).
 - » Pressing the Soft Key marked **HI** activates high-band (1 kHz - 16 kHz) adjustments:
 - * The top right Soft Key toggles the high-band EQ method between **Band-pass** and **Shelf** modes.



Figure 4-5: Channel EQ screen.

- * The Preview Volume Control knob now flashes **HI GAIN**. Rotating it adjusts the high-band EQ gain.
- * The Studio Monitor Volume Control now flashes **HI FREQ**. Rotating it adjusts the center frequency of the band-pass filter if high EQ is set to **Band-pass** mode, or the top fre-

quency of the shelf if in **Shelf** mode.

- * Pressing the Soft Key marked **MID** activates mid-band (100 Hz - 1 kHz) adjustments. Gain and center frequency controls now read **MID GAIN** and **MID FREQ**, and are adjusted as described above. The **Band-pass / Shelf** option is not available for the midrange EQ band.
- » Pressing the Soft Key marked **LOW** activates low-band (25 Hz - 400 Hz) adjustments. Gain and center frequency controls now read **LOW GAIN** and **LOW FREQ**, and are adjusted as described above. The **Band-pass / Shelf** option is not available for the low EQ band.
- » Pressing the Soft Key marked **EXIT** returns you to the **Monitor Options** screen.

Note: SmartSurface EQ features SmartQ™ automatic bandwidth system that varies the Q of the selected parametric band to provide the most pleasing EQ effect. For details on SmartQ, please refer to Page 7 of this manual.

- **Lock / Unlock** allows the board operator to “lock” the channel state in its current state. This is useful to prevent changes to the on-air program in situations where the Control Room is to be left unattended for a long period of time.

Application: 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 if the fader is moved while the channel is **Locked**, unlocking the channel will *immediately* change audio gain to match the current fader position.

When a channel is **Locked**, you can't change::

- » **Bus assignments.** The channel cannot be assigned to any new channel or removed from any channel previously assigned.
- » **Fader setting.** When **Lock** is selected, the position of the fader is locked as well, so that the input level of the source cannot be changed (regardless of any changes made to the fader itself).

- » *Channel on/off state.* If the channel is **On** at the time **Lock** is selected, it cannot be turned off. Conversely, the channel can be **Locked Off** as well.

This command appears on the lower left Soft Key. Pressing this key toggles between **LOCK** and **UNLOCK**.

- **Feed Source** changes the right-hand screen to display the Feed To Source menu (Figure 4-6). This item only appears on the **Channel Options** menu if the assigned source (i.e., a codec or phone hybrid) has an associated backfeed or mix-minus, and determines the source of the audio being backfed to the channel source. This selection is normally pre-

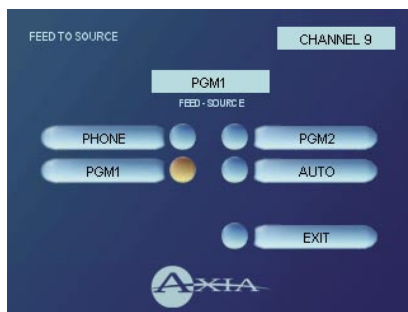


Figure 4-6: Feed To Source menu screen.

set when a Source Profile is constructed (see Chapter Two, “Configuring Inputs,” for information on Source Profiles), but can be modified on-the-fly.

- » **Phone** backfeeds audio from the Phone bus, with mix-minus if needed.
- » **PGM1** backfeeds audio from the Program-1 bus, with mix-minus if needed.
- » **PGM2** backfeeds audio from the Program-1 bus, with mix-minus if needed.
- » **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.

Pressing the Soft Keys labeled **PHONE**, **PGM1**, **PGM2** or **AUTO** changes the channel backfeed selection. **EXIT** returns you to the **Channel Options** menu.

- **Aux Sends** changes the right-hand screen to display the Channel Aux Sends menu (Figure 4-7). SmartSurface has two stereo Aux buses that can be used as utility buses for mixing, for

constructing custom IFB mixes, or as effects buses for production. The Soft Keys and Volume Control knobs mirror the options onscreen. The controls for **Send 1** and **Send 2** are arranged in columns onscreen. **Send 1** controls are on the left, **Send 2** on the right.

- » **Send 1/2** switches this channel’s send to the Aux 1 or Aux 2 bus on and off. The top left Soft Key

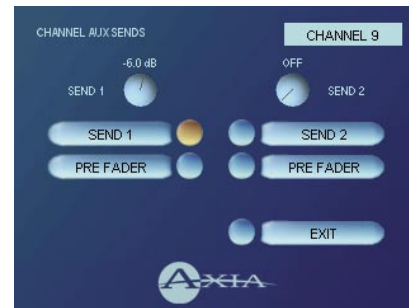


Figure 4-7: Channel Aux Sends menu

toggles this function for **SEND 1**, the top right Soft Key toggles **SEND 2** on and off. Onscreen, the indicators next to the **Send 1** and **Send 2** options will turn gold when these feeds are on, blue when they are off.

- » **Pre Fader/Post Fader:** you have the option to send channel audio to the Aux buses either **Pre Fader** (fixed gain) or **Post Fader** (controlled gain). The upper-middle left and upper-middle right Soft Keys toggle the **PRE FADER** and **POST FADER** send selection for the Aux 1 and Aux 2 buses, respectively.
- » **Send 1 Level:** the Preview Volume Control’s alpha display is now flashing **SEND_1**. Turning it adjusts the gain of the Aux 1 send level.
- » **Send 2 Level:** the Studio Monitor Volume Control’s alpha display is now flashing **SEND_2**. Turning it adjusts the gain of the Aux 2 send level.
- » **Exit** returns you to the **Channel Options** menu. Choose the bottom right Soft Key to **EXIT** this menu.
- **Fader Norm/Fader Start** allows you to set the channel activation method of the fader .
 - » **Fader Norm** conforms to the US method of requiring talent to manually turn the channel on and off.

- » **Fader Start** follows the European convention of activating the channel and associated machine logic when the fader is raised from ∞ .

This choice applies only to the current channel. The bottom-middle right Soft Key toggles between **FADER NORM** and **FADER START**.

- **Exit** returns the channel to normal operation mode. Press the bottom right Soft Key to **EXIT** the menu.

Channel Preview Key

Preview (called “cue” in the analog days) allows operators to listen to sources prior to airing them. The Preview bus auditions content in full stereo.

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

If you want to preview multiple channels, pressing and *holding* any **Preview** key disables the interlock; other sources can be added to the Preview mix by pressing them. Conversely, channels can be removed from a multiple Preview selection by pressing and *holding* any lit **Preview** key and deselecting the individual channels you wish to remove from Preview. A *momentary* press of any lit **Preview** key will remove all channels from Preview.

Note: If you desire, the Preview Interlock function can be disabled entirely using Show Profile settings — See Chapter 5, “Show Profiles.”

The audio from the Preview bus feeds the operator’s headphones (if enabled), as well as any speakers dedicated to the Preview bus. These speakers mute whenever the console operator’s microphone is **ON**, or assigned to the Preview or Record buses.

The Preview bus is also the key to a powerful **Talk to Preview** function provided in the Monitor section.

Pressing the **Talk to Previewed Source** key routes the board operator’s mic to any outputs associated with an input that’s been placed in Preview mode, i.e.: you can talk to a phone caller’s mix-minus feed by simply placing the caller’s channel in Preview mode and pushing **Talk to Previewed Source**. You can do the same with individual headphone feeds for guest position mics.

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. (**On** and **Off** keys still function normally.)
2. In **fader-normal** mode, the on/off status and start/stop commands will follow the channel on/off buttons independent of fader position.

On and Off Keys

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

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

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

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

Now, let’s see what happens when specific types of sources are assigned to a fader channel.

Source-Specific Channel Controls

As mentioned previously, some fader-channel functions change their behavior to suit the specific type of source assigned to the fader. An Operator's 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.

Operator Microphone Channel Operation

The Operator microphone is the board operator's mic. It's always located in the Control Room, so activating a channel designated as the Operator mic will affect the muting of the CR monitor speakers and the Preview speakers.

When the board operator pushes a **Talk to Studio** or **Talk to Previewed Source** key, all output bus assignments for the Operator microphone are temporarily muted, while his mic's audio is routed to the requested **Talk to...** destination. When the button is released, the channel outputs return to normal.

A powerful **Talk to Preview** function is provided for the board operator using the Operator mic. Pressing

the **Talk to Preview** key routes the operator's microphone to any source(s) in Preview mode. For example, the board op can talk to a caller's mix-minus feed by placing the caller's channel in Preview mode and pushing the **Talk to Preview** key. (Since this channel is configured to be a host microphone, placing this channel in Preview and hitting **Talk to Preview** will do nothing.)

Depressing the channel **On** key for the Operator Mic while the channel is already on will cause the channel to mute 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 Operator microphone channel 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.

The block diagram for this channel selection is shown in Figure 4-8.

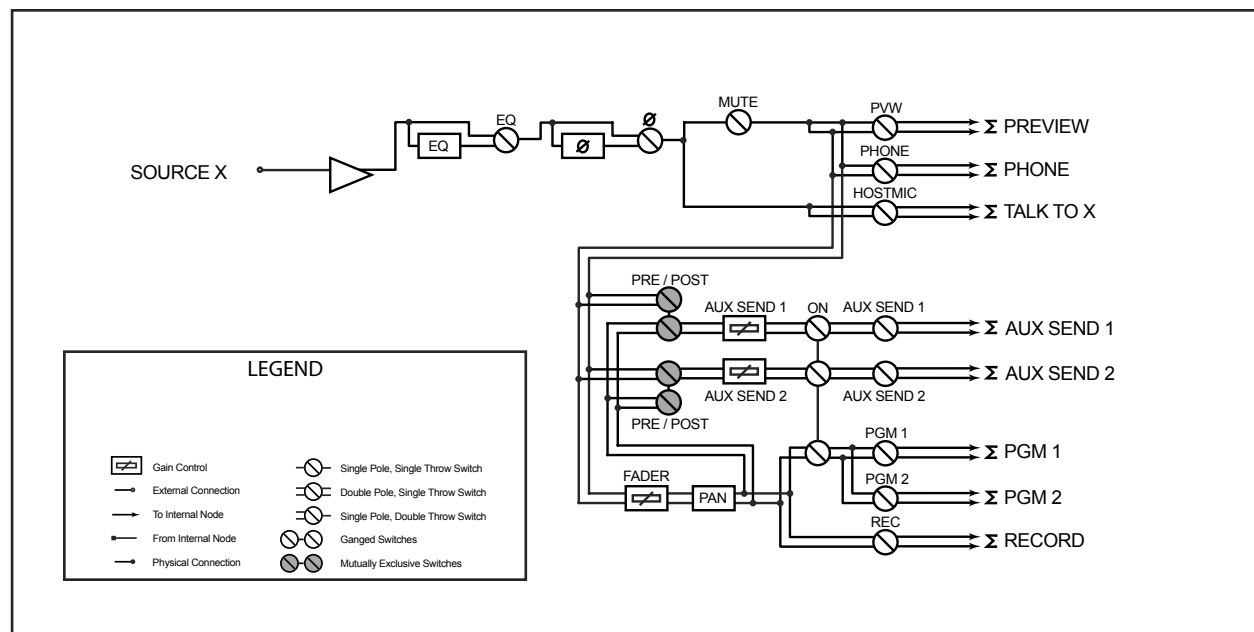


Figure 4-8: Operator Microphone channel block

Control Room Guest Microphone Channel Operation

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

The Control Room monitor speakers and Preview speakers will mute when this channel is turned on, or when **Preview** or **Record** are selected. The Control Room speakers will also be muted when this channel is assigned to the **Phone** bus.

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 **Preview** key will be lit and the Status Symbol will display the word **TALK**. When the user stops talking, the channel

returns to normal, and its **Preview** key flashes for three seconds, alerting the board operator to the guest who spoke to him, and enabling him to reply using the **Talk to Previewed Source** key.

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

Control Room Guest Microphone channels will mute the Control Room speakers and the Preview speakers whenever the channel is **On**, or if **Preview** or **Record** keys are selected. If only the **Phone** key is selected, the Control Room speakers will mute independently, leaving the Preview speakers un-muted.

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

The block diagram for this channel selection is shown in Figure 4-9.

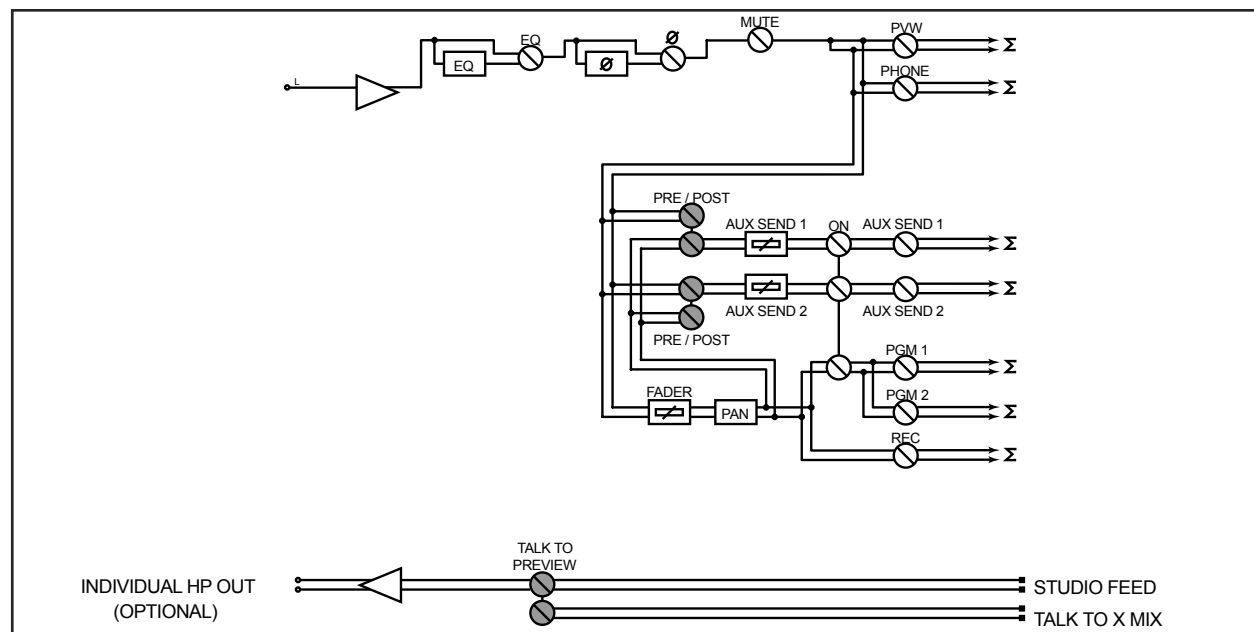


Figure 4-9: Control Room Guest Microphone channel block

Control Room Producer Microphone Channel Operation

The third type of microphone channel available with SmartSurface is the Control Room Producer Microphone. This mic type is intended for show producers who contribute to program content and are located in the control room. so activating a Producer mic channel affects the muting of the CR monitor speakers and the Preview speakers.

Associated GPIO outputs support several functions that can be controlled from a producer’s remote panel. Producer 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, as well as two remote talkback functions. CR Producer mics may also use the **Talk to Studio** and **Talk to Previewed Source** functions previously described in the “Operator Microphone Channel Operation” section of this chapter.

Remote Talk allows a producer, located in the Control Room, to talk to Studio guests or previewed sources with associated backfeeds. When **Talk to Studio** or **Talk to Previewed Source** is remotely activated on a CR Producer 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 **Preview** key will be lit and the Status Symbol will display the word **TALK**. When these functions are no longer active, the channel outputs return to normal.

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

The Producer microphone channel will mute the Control Room speakers and the Preview speakers whenever the channel is **On**.

The block diagram for this channel selection is shown in Figure 4-10.

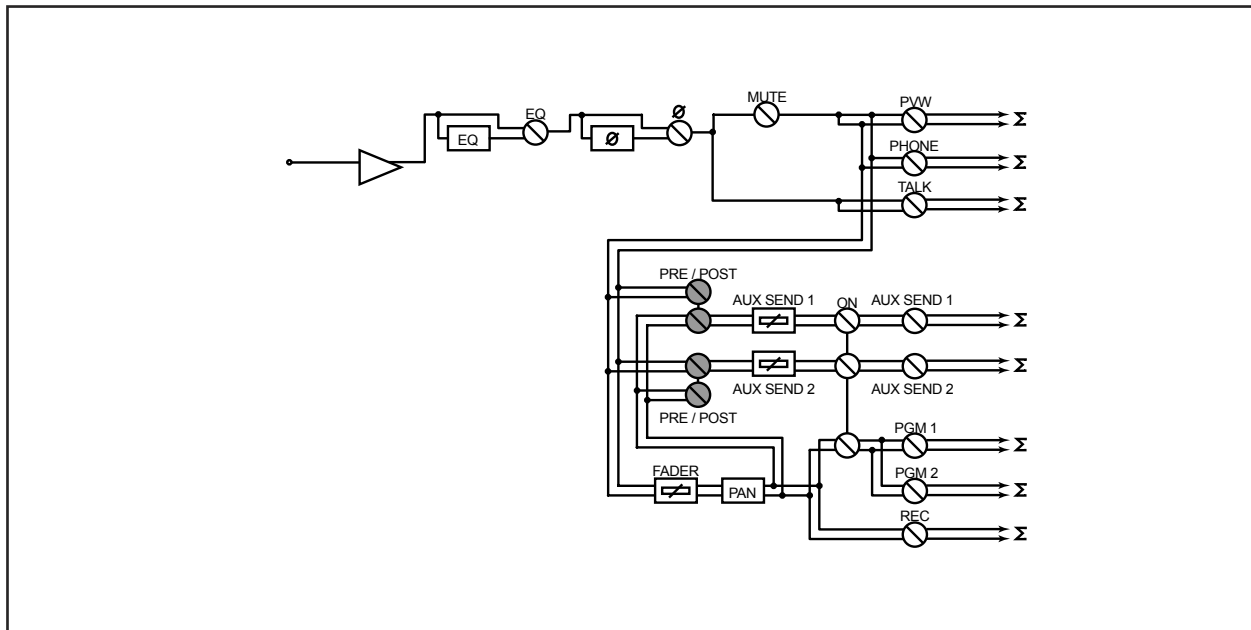


Figure 4-10: Studio Guest Microphone channel block

Studio Guest Microphone Channel Operation

The fourth type of microphone channel available with SmartSurface is a Studio Guest 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 Guest Microphone channels is identical to that of the Control Room Guest Microphone channel. Studio Guest Microphone channels can be turned on and off remotely and will 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 enables talent and guests to communicate with the board operator. When **Talk** is remotely activated on a channel, outputs to program buses for that channel are muted and the source audio is fed to the Talkback bus. While this is active, the **Preview** key will be lit and the Status Symbol will display the word **TALK**. When the user stops talking, the channel immediately returns to its previous state, and its **Preview** key flash-

es for three seconds, alerting the board operator to the guest who spoke to him, and enabling him to reply using the **Talk to Previewed Source** key.

When the **Remote Mute** function is activated, the Status Symbol will display the word **MUTE**, and will mute all the channel outputs until the command is released, at which time the channel is again active.

The Studio Monitor speakers will mute when a Studio Guest Microphone channel is turned on, or if it is assigned to either the **Record** or **Phone** bus.

The Studio Guest Microphone channel can be associated with an individual Headphone feed. All Studio Guest Microphone-associated feeds will monitor the audio selected on the **Studio Monitor** selector, but the board op can communicate with individuals privately using the **Talk To Previewed Source** function.

The block diagram for this channel selection is shown in Figure 4-11.

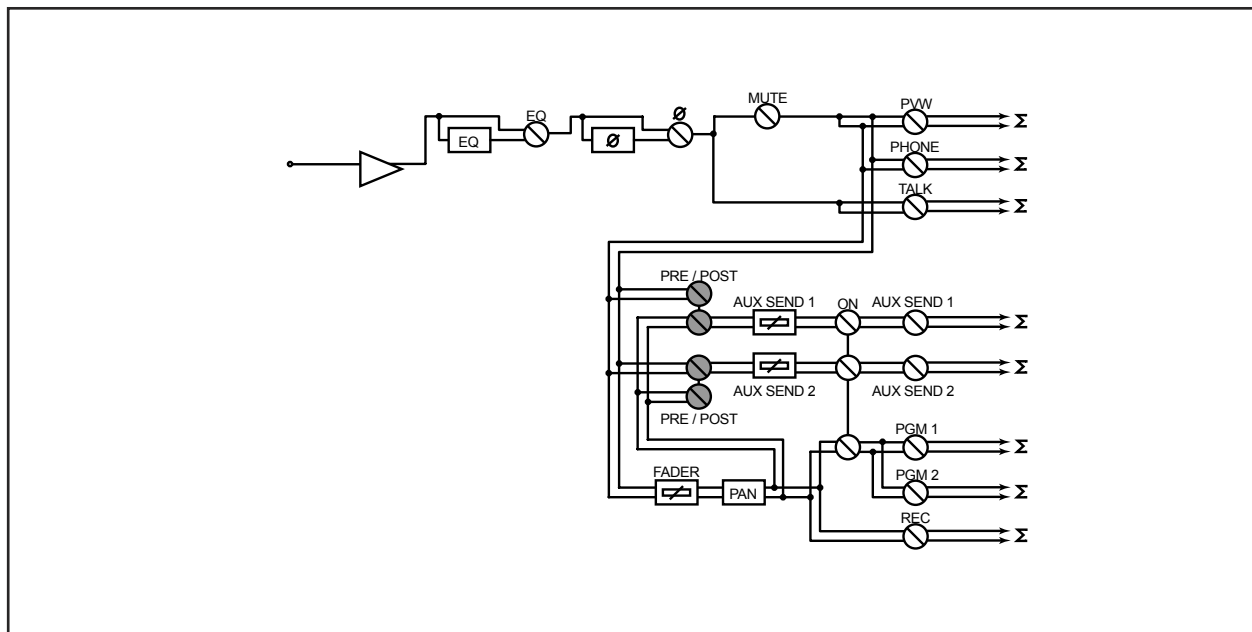


Figure 4-11: Studio Guest Microphone channel block

Line Channel Operation

Line-level devices like CD players, delivery system outputs, tape players, etc., use the Line Channel profile. The Line Channel is stereo, but the user can change the input mode to mono, left-only or right-only by using the **Mode** function in the **Channel Options** menu.

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

A **Reset** function is provided to remotely turn the channel **Off** while suppressing the **Stop** command, 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 SmartSurface 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 4-12.

Phone Channel Operation

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

SmartSurface can accommodate up to 16 callers at a time, and unique mix-minus feeds to callers are generated automatically. If Feed-to-Source is set to **Auto** mode the caller hears the output of the **PGM-1** bus, minus himself, when a Phone Channel is **On**. When the caller's channel is **Off**, he will hear anything assigned to the Phone bus, minus himself. This auto-mode switching can be disabled in the **Channel Options** menu (see Page 9). The Phone bus is fed by all channels before the fader, independent of on/off status.

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

In **Auto** mode, the feed-to-source output will normally toggle between **PGM-1** and **Phone**, but it can also be locked on **PGM-1**, **PGM-2** or **Phone**, using the options provided when constructing the Source Profile for

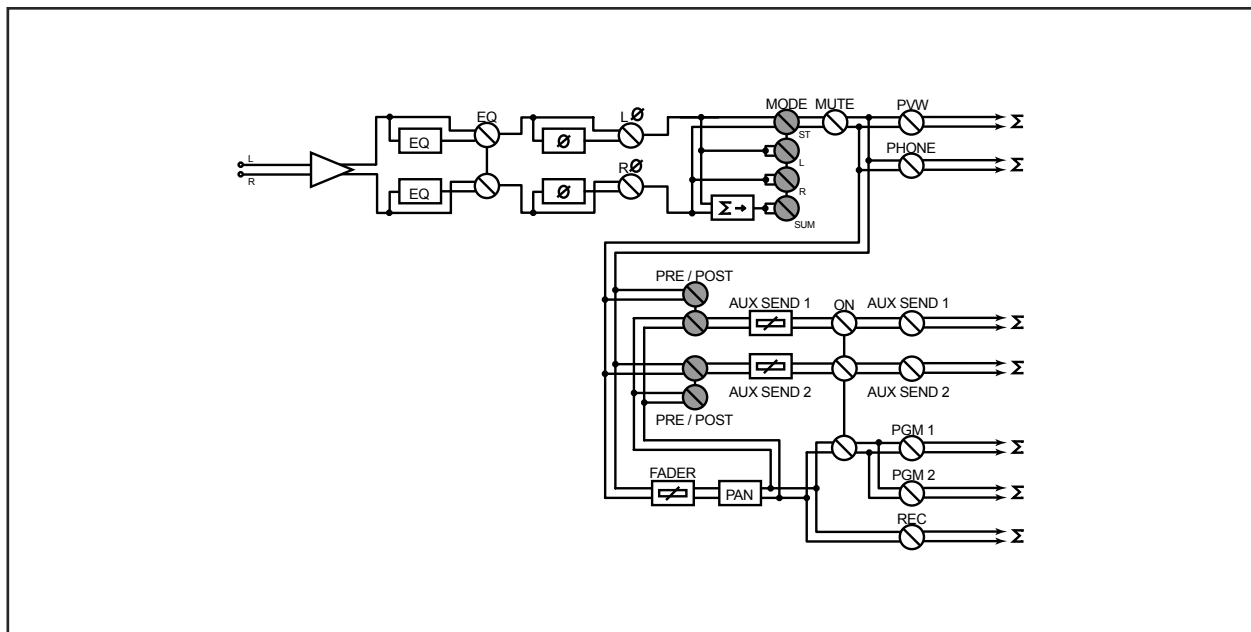


Figure 4-12: Line channel block

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 for details on Source Profile options).

If the board operator assigns a Phone Channel to the Preview bus and presses the **Talk To Previewed Source** key, the Status Symbol icon will change to the word **“HOST”** to indicate that the talkback channel is active.

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 infor-

mation interfacing to GPIO inputs and output.

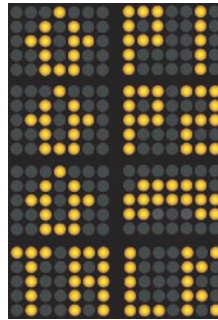


Figure 4-13: Status Symbols

Figure 4-13 shows Status Symbol displays used for Phone and Codec Channels. From top to bottom, the Status Symbols indicate that the caller/remote is hearing:

1. PGM-1 mix-minus,
2. PGM-2 mix-minus,
3. Phone bus mix-minus,
4. Board-op’s talkback mic.

The block diagram for this channel selection is shown in Figure 4-14.

Note: A complete listing of Status Symbols is found at the end of this chapter.

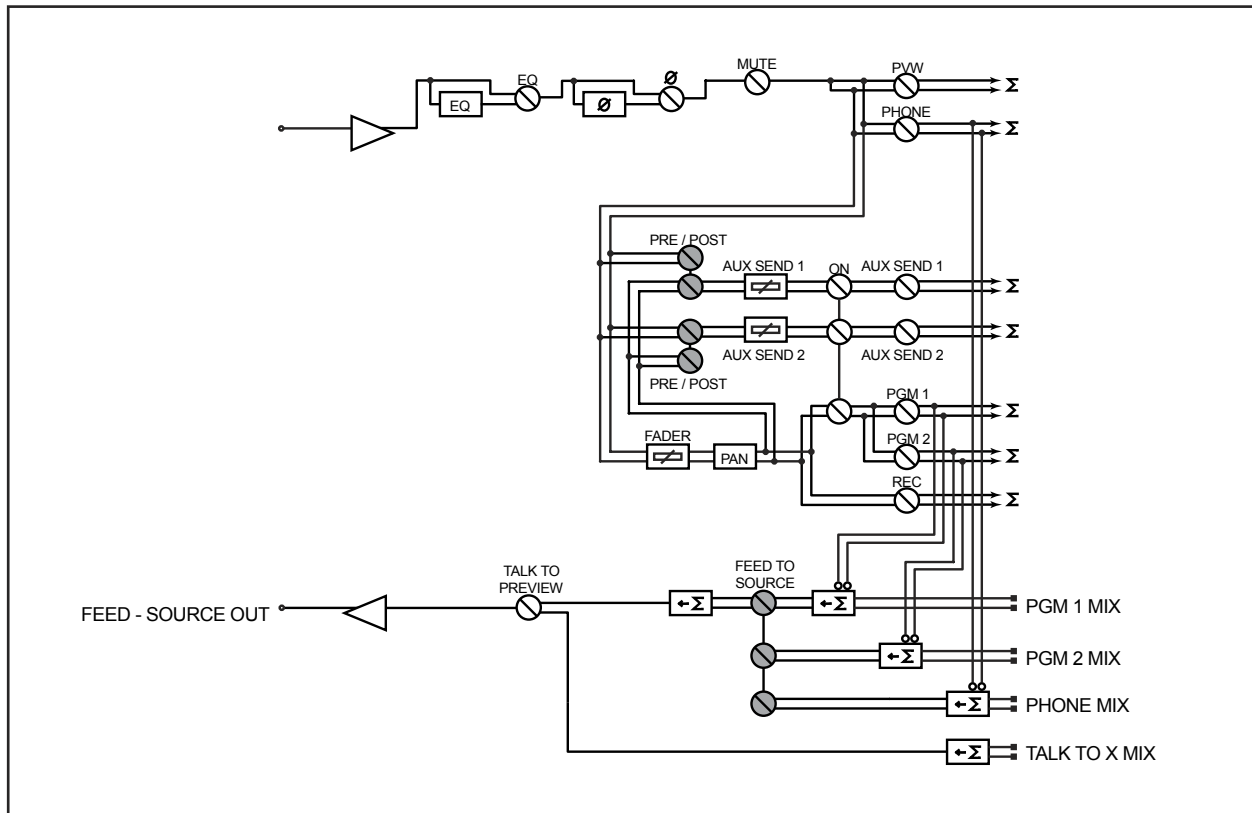


Figure 4-14: Phone channel block

Codec Channel Operation

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

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

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

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

The mix-minus output normally feeds **PGM-1** audio to the remote, but can be switched to **PGM-2** or **Phone**. 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 **Talk To Previewed Source** audio when the board op talks to the remote. The Right output sends uninterrupted program audio for use as a PA feed.

Feed-to-Source may also be set to **Auto** mode, which operates identically to the Phone channel function described on Page 38.

Status Symbol displays for Codec Channels are the same as for Phone Channels. The block diagram for this channel selection is shown in Figure 4-15.

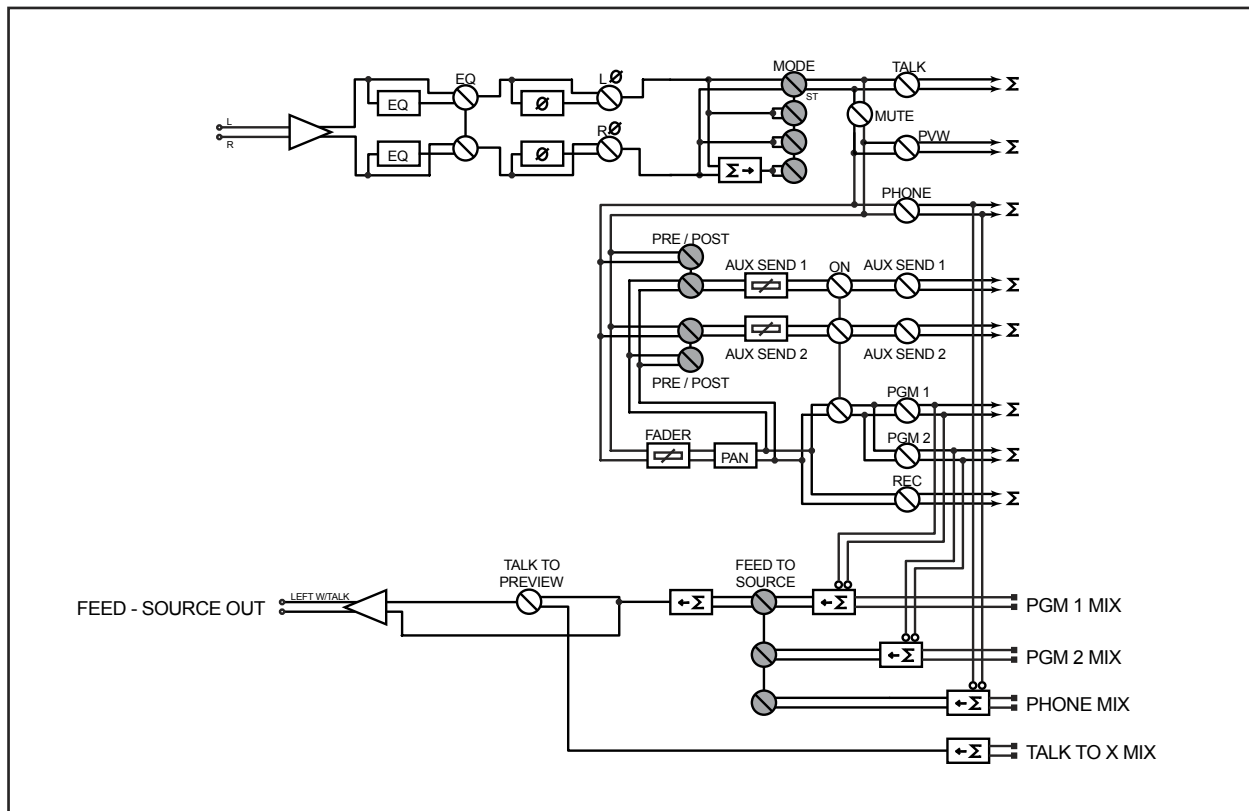


Figure 4-15: Codec channel block

Monitor Section Controls

The top of the SmartSurface center section contains the controls for selecting monitor sources and adjusting their levels, and for the board operator's **Talk To** controls. We'll examine the controls one-by-one.



Figure 4-16: Monitor Control Section

Talk To... Keys

The **Talk To** section controls one of SmartSurface's most powerful functions. When these keys are pressed, the board operator's microphone signal is temporarily re-routed to a particular destination. If the board operator's microphone is **On** and assigned to any bus, the outputs will be muted while the **Talk To** keys are depressed.

- Pushing **Talk to Studio** will route the board op's microphone audio to the studio's monitor speakers (unless muted by an open mic in the studio) and headphones. There is a GPIO logic output associated with the **Talk To Studio** output, which is active while the button is depressed.
- Pressing **Talk To External** will feed the board op's microphone signal to a dedicated audio output you can use for a variety of purposes. There is a GPIO logic output associated with the **Talk To External** output, which is active while the button is depressed.
- The **Talk To Previewed Source** option allows the board operator to easily talk to any channel presently in the preview mode, *providing that channel has a corresponding audio output*. For example, a channel configured as a hybrid will have a Send-to-Source (mix-minus) output associated with it. Placing that channel in Preview and pushing the **Talk to Pre-**

view key will allow the board op to talk directly to the caller. Channels configured for remote (codec) operation also have associated Send-to-Source outputs which can be talked to .

Control Room Guest microphone channels and Studio microphone channels can be equipped with individual headphone outputs. Using these individual headphone feeds, SmartSurface allows the board operator to speak to an individual talent or guest privately by simply selecting that channel's **Preview** key and depressing the **Talk To Previewed Source** key.

Sometimes, a talent or guest will initiate a conversation with the board operator using their Remote **Talk** key (GPIO function), which routes their audio to the Preview speakers. As long as they push this button, their channel **Preview** key will illuminate, and its Status Symbol will display the word **TALK**. When they release the Remote **Talk** button, the channel's **Preview** key will flash for three seconds before extinguishing. If the board operator presses **Talk To Previewed Source** while the button is flashing, the board op will be able to talk to that source. The **Preview** key will continue to flash as confirmation until the board op releases the **Talk To Previewed Source** key.

Note: The Control Room Monitor speakers are muted while the **Talk To...** function is active, but the Preview speakers will not be muted by this function. They may, however, already be muted by another mute source.

Monitor Volume / Selection Knobs

Just below the **Talk To...** keys are four control knobs. These are rotary encoders that function as volume controls when rotated and, when pushed, source selection controls for the four monitor buses. All four controls have 10 character alpha displays, which show the selected source, and, when the volume is being adjusted, graphically display the volume level. From left to right, these controls are:

- **Control Room Monitor Volume.** Normally, rotating this knob adjusts the level of the main control

room monitor speakers. Pushing the knob enters selection mode, allowing the user to select between monitor sources. In selection mode, the right-hand display screen will show a list of available monitor sources, such as **PGM-1**, **PGM-2**, **Record**, **External** (generally fed by a tuner or studio processing chain), etc.. Users can also monitor any source present on the Livewire network. The user rotates the knob to select the desired source and pushes it again to “take” that source. If the user enters selection mode but doesn’t make any changes within four seconds, selection mode exits.

To help protect the user from setting the monitor volume too loudly when the Control Room Monitor is dimmed (for example, when the **Talk To...** feature is in use) and subsequently being blasted when the dim ends, SmartSurface remembers the previously set Monitor volume, and restores it as soon as the dim ends.

The Control Room Monitor Volume knob’s alpha display shows the user the name of the source currently being monitored, and prefixes it with “CR_” to help avoid confusion. For instance, if the **PGM-1** bus is being monitored, the alpha display will read **CR_PGM1**.

- **Preview Volume.** This knob controls the level of the studio’s Preview speakers. These speakers will mute when any Control Room microphone is turned on, or is assigned to either the **Preview** or **Record** bus.

Preview volume affects all signals routed to the Preview speakers, including talkback signals from external sources.

The Preview Volume knob’s alpha display dynamically changes to indicate the source and mode of the preview speakers. If a single channel is selected either for **Preview** or **Talk To...**, the alpha display will name the source, prefixed with “PV_” to identify the Preview control. For example, if the board op assigns a source named “CD-1” to the Preview bus, the Pre-

view Volume knob display will read **PV_CD-1**. If multiple channels are in **Preview**, the display will read **PV_MULTIPLE**.

Pushing the Preview Volume control has no effect.

- **Studio Monitor Volume.** This knob works the same way as the control for the Control Room Monitor Volume, and allows the board op to select the source fed to the Studio Monitor speakers, and adjust its volume.

The Studio Monitor Volume knob’s alpha display shows the name of the source currently being monitored, and prefixes it with “ST_”. For instance, if the **Phone** bus is being fed to the Studio monitors, the alpha display will read **ST_PHONE**.

- **Headphone Volume.** This knob controls the volume of the *board operator’s* headphones. Headphones are highly configurable, since operators have different monitoring preferences. For example, the board op’s headphones can be set to monitor an external air feed, while the Control Room Monitor speakers monitor **PGM-1**. Configuration of monitoring modes is established by pressing the **Monitor Options** key (discussed in the next section) and selecting the configuration menu.

Normally, the operator’s headphones will listen to whatever has been selected for the Control Room Monitor speakers, but a separate source can be fed to the headphones if **Use Headphones Source** has been selected from the Monitor Options menu screen.

If the **Use Headphones Source** option has been selected, pushing the knob enters selection mode, allowing the user to select between headphone sources. In selection mode, the right-hand display screen will show a list of available headphone sources. The user rotates the knob to select the desired source and pushes it again to “take” that source. If the user enters selection mode but doesn’t make any changes within four seconds, selection mode exits.

If **Follow Monitors** is enabled in the headphones (see the “Monitor Options” section for details on how to do this) and **Auto** mode is active, the Preview signal will be summed to mono and fed to the right ear of the headphones; Control Room Monitor selection will be summed to mono and fed to the left ear. Deselecting **Preview** restores the normal stereo mode. If **Auto** mode is inactive, **Preview** audio will feed the headphones in normal stereo.

The Headphone Volume knob’s alpha display shows the name of the source currently being monitored, and prefixes it with “HP_”. For instance, if the **Record** bus is being fed to the Studio monitors, the alpha display will read **HP_RECORD**.

The block diagram for Control Room Monitor section is shown in Figure 4-17.

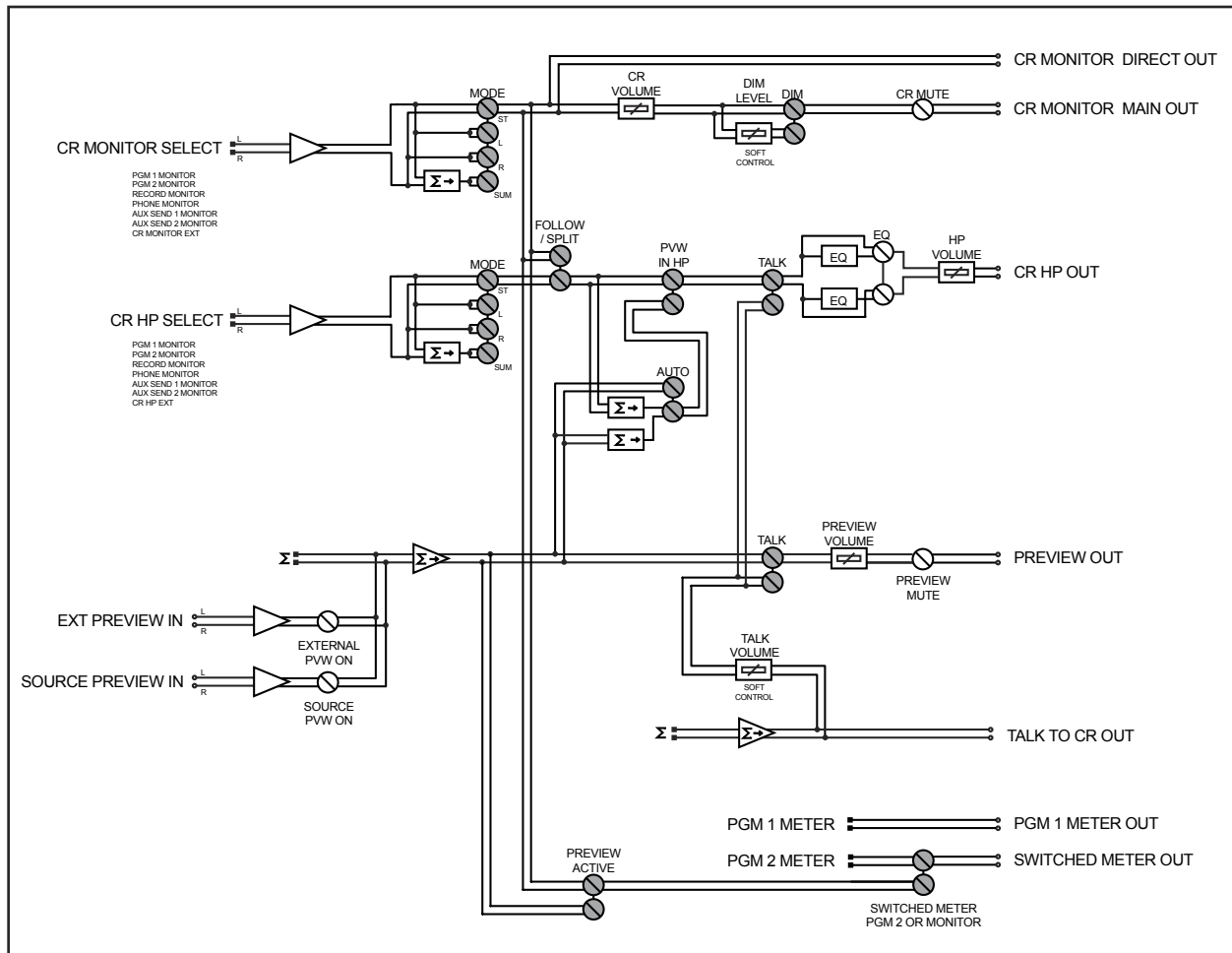


Figure 4-17: Control Room Monitor block

Monitor Options Key



Pressing this key causes the right display to show the **Monitor Options** screen shown in Figure 4-18.

Note: Activating this or any option screen will cause the context-sensitive center Soft Keys and the two center Monitor Control knobs (**PVW** & **STUDIO**) to mirror the functions shown in the display screen.

- **Exit** returns you to the meter screen.
- **Talkback Gain** lets the operator boost or cut the volume of the talkback signal coming to the board from a microphone or remote source. The alpha display beneath the Preview Volume Control knob flashes **TALK_GAIN**; rotate it to adjust.
- **Dim Gain** controls how much volume reduction will



Figure 4-18: Monitor Options screen.

be applied to the Control Room Monitor speaker feed during talkback operations. The alpha display beneath the Studio Monitor Volume Control flashes `DIM_GAIN`; rotate it to adjust.

- **MTR** (Meter) controls the source for the meters displayed on the right screen. Pressing the top left Soft Key toggles this setting between metering the **PGM-2** bus (`MTR PGM-2` is displayed), and metering whatever source the operator has assigned to the Control Room monitors (`MTR MONITR` is displayed).

Note: If `MTR MONITR` is chosen, the right-hand meter will meter the **Preview** selection, overriding the previous meter choice, whenever **Preview** is activated. When **Preview** ends, the previous selection will again be metered.

- **SCRUB** determines whether the scrub wheel at the bottom of SmartSurface's center section will be used for adjusting Control Room Monitor volume or the board operator's Headphone Volume. Pressing the top-middle left Soft Key toggles this setting between `SCRUB HP` (headphone volume), `SCRUB CR` (Control Room Monitor volume) and `SCRUB OFF`.
- **HP** toggles the board operator's Headphone feed between **Follow** and **Split** modes. If **HP Follow** is selected, the board op's headphones mirror the selection of the Control Room Monitor feed. If **HP Split** is selected, the board op may choose a headphone feed independent of the Control Room Monitor selection using the push-and-select feature of the Headphone Volume Control knob. Pressing the top right Soft Key toggles this setting between `HP SPLIT` and `HP FOLLOW`.

- **HP PVW** (Headphone Preview) toggles the Headphone Preview mode between **Normal**, **Auto** and **Off**. If Headphone Preview is 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 (as explained above). 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*. Pressing the top-middle right Soft Key toggles the adjacent alpha display between `HP PVW OFF`, `HP PVW NORM` and `HP PVW AUTO`.
- **HP EQ** allows three-band parametric equalization to be applied to the board operator's headphones to suit personal preference. Pressing the Soft Key labeled `HP EQ` changes the right-hand screen to show the Headphone EQ screen (Figure 4-19).

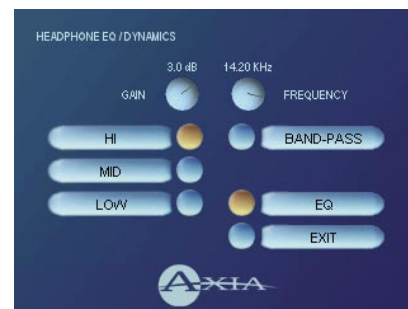


Figure 4-19: Headphone EQ screen.

Note that the Soft Keys mirror the functions shown on-screen.

- » Pressing the Soft Key marked **HI** activates high-band (1 kHz - 16 kHz) adjustments:
 - * The top right Soft Key toggles the high-band EQ method between **Band-pass** and **Shelf** modes.
 - * The Preview Volume Control knob now flashes `HI GAIN`. Rotating it adjusts the high-band EQ gain.
 - * The Studio Monitor Volume Control now flashes `HI FREQ`. Rotating it adjusts the cen-

ter frequency of the band-pass filter if high EQ is set to **Band-pass** mode, or the top frequency of the shelf if in **Shelf** mode.

- » Pressing the Soft Key marked **MID** activates mid-band (100 Hz - 1 kHz) adjustments. Gain and center frequency controls now read **MID GAIN** and **MID FREQ**, and are adjusted as described above. **Band-pass / Shelf** is not available for the midrange EQ band.
- » Pressing the Soft Key marked **LOW** activates low-band (25 Hz - 400 Hz) adjustments. Gain and center frequency controls now read **LOW GAIN** and **LOW FREQ**, and are adjusted as described above. **Band-pass / Shelf** is not available for the low EQ band.
- » Pressing the Soft Key marked **EXIT** returns you to the **Monitor Options** screen.

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

Studio Options Key

The **Studio Options** key activates a menu of choices relating to the Studio Monitors (Figure 4-20).



Figure 4-20: Monitor Options screen.

- **Dim Gain** controls how much volume reduction will be applied to the Studio Monitor speaker feed during talkback operations. The alpha display beneath the Studio Monitor Volume Control flashes **DIM_GAIN**; rotate it to adjust.
- **Exit** returns you to the meter screen.

Transport and Other Controls

The middle portion of SmartSurface's center section contains the Soft Keys, controls to operate an external dedicated record/playback device, and other control options.



Figure 4-9: Center Control Section

Transport Keys

The **Transport Keys** can be enabled to control a dedicated record/playback device. The selection of this device is defined using SmartSurface's Show Profiles setup screen, which is constructed in a similar method to that of a Source Profile, using SmartSurface's HTTP interface. (See Chapter Five, "Show Profiles," for instructions on assigning these keys to a recording device.)

The five **Transport Keys** are, from left to right, **Re-wind**, **Fast Forward**, **Stop**, **Play** and **Record**.

Soft Keys

The eight **Soft Key** buttons and displays just below the **Transport Controls** are context sensitive, and provide the user with controls appropriate to the task being performed.

As various modes are invoked, the **Soft Keys** will change to provide access to menu items and commands. The 10-character alpha displays adjacent to the **Soft Keys** indicate menu choices. For example, when the **Monitor Options** key is depressed, the **Soft Keys** display commands that follow the on-screen display, allowing the board operator to change various user-definable settings.

Record Mode Key

The **Record Mode** key is a special “one-touch” button. **Record Mode** settings are unique to each **Show Profile** and are defined using the Show Profile setup screen via SmartSurface’s HTTP interface (see Chapter 4, “Show Profiles” for details). This special mode can be set to automatically start a designated record device or application, assign pre-defined sources to the Record bus, and automatically change Monitor sources so that talent can record phone or remote interviews with a single button-press. The mode is terminated by pressing the **Record** key a second time.

Control Options Key

Pressing this key causes the right-hand display screen to change to the **Control Options** menu (Figure 4-21). This menu contains configuration options for the



Figure 4-21: Control Options screen.

Aux buses. This is also where the board operator loads Show Profiles, custom surface configurations that can be loaded as needed to reconfigure SmartSurface for different shows and talent.

As in all **Options** modes, SmartSurface’s Soft Keys dynamically display control options appropriate to the on-screen menus.

- **Exit** quits **Control Options** mode and returns SmartSurface to normal operation. You can also push the **Control Options** key again to exit.
- **Load Profile** changes the right-hand display screen to a list of show profiles available to load. The **Load Profile** screen shows the profile that’s currently running, and lists profiles you can load. When this menu is displayed, the alpha display under the Preview Speaker Volume Control flashes **PROFILE**. Turning the knob scrolls through the list; press the

knob to “take” the new profile. You can also use the Soft Keys labeled **UP**, **DOWN** and **SELECT** to navigate the list; choose **EXIT** to return to the **Control Options** screen.

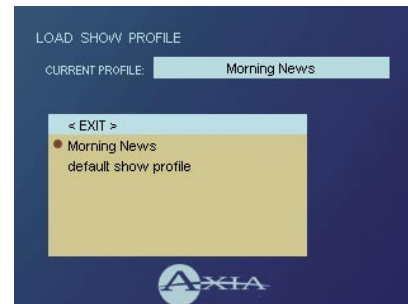


Figure 4-22: Load Show Profile screen.

As soon as you **SELECT** or “take” a new show profile, SmartSurface begins loading it. During the profile loading process, a status line at the bottom of the left display screen will read **Loading (Show Profile Name)** and displays a progress bar to inform the operator of the operation in progress.

Note that 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.

Note: SmartSurface 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 **PGM-1**, fader up, channel **On**). The board op loads a new **Show Profile** to ready SmartSurface 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, SmartSurface queues the new source and displays a scrolling message on the channel alpha display: **TURN OFF CHANNEL TO LOAD SOURCE**. When the board op is finished with the satellite feed, he turns the channel **Off** and the CD player automatically loads.

- **Help** displays a list of help topics, providing a “quick start” guide to help new users through SmartSurface operations, and installed software version.

- **Aux Sends** contains settings for the master send levels of the two stereo Aux 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).



Figure 4-23: Aux Sends master screen.

SmartSurface has two stereo Aux buses that can be used as utility buses for mixing, for constructing custom IFB mixes, or as effects buses for production. The Soft Keys and Volume Control knobs mirror the options onscreen. This menu contains many of the same controls as the individual **Channel Options** **Aux Sends** menu (see p. 17), but affects global settings for the Aux buses.

The controls for **Send 1** and **Send 2** are arranged in columns onscreen. **Send 1** controls are on the left, **Send 2** on the right.

- » **Send 1/2** switches the master Aux 1 Send or Aux 2 Send bus on and off. The top left Soft Key toggles this function for **SEND 1**, the top right Soft Key toggles **SEND 2** on and off. Onscreen, the indicators next to the **Send 1** and **Send 2** options will turn gold when these feeds are on, blue when they are off.
- » **Preview** lets the board op hear the audio being fed to the Aux 1 or Aux 2 Sends using the Preview channel. Press the **PREVIEW** Soft Key to preview send audio; release the Soft Key to exit preview.
- » **Send 1 Level:** the Preview Volume Control's alpha display is now flashing **SEND_1**. Turning it adjusts the gain of the Aux 1 send level.
- » **Send 2 Level:** the Studio Monitor Volume Control's alpha display is now flashing **SEND_2**. Turning it adjusts the gain of the Aux 2 send level.

- » **Exit** returns you to the **Channel Options** menu. Choose the bottom right Soft Key to **EXIT** this menu.

- **Aux Rtn 1 / Aux Rtn 2** menus contain the master gain, pan and bus assignment controls for the two Aux buses. **Aux Return** settings can be pre-defined for each show using Show Profiles (see Chapter Five, "Show Profiles"), but can be modified using these menus. Press the top-middle right Soft Key, labelled **AUX RTN1**, or the bottom-middle right key, labelled **AUX RTN2**, to access the **Aux Return Master** menus (Figure 4-24).



Figure 4-24: Aux Return master screen.

There are several options on these screens:

- » **Exit** (the bottom right Soft Key) returns you to the Return Master menu.
- » **Return 1/2** (the upper left Soft Key, labelled **RETURN**) is the master on/off switch for the Aux 1 or Aux 2 Return, respectively. Pressing this key toggles the Aux Return between **Off** and **On**; the indicator next to the **Return** option onscreen will glow gold when enabled.
- » **Preview** lets the board op hear the audio being fed to the Aux 1 or Aux 2 Return using the Preview channel. Press the **PREVIEW** Soft Key to preview audio; release the Soft Key to end preview.
- » **Pan/Mode** lets the operator adjust the pan/balance of the return source, and correct signal phase errors. Pressing the lower-middle left Soft Key (marked **PAN/MODE**) causes the right-hand screen to display the Aux Return Pan/Mode Menu (Figure 4-25). The alpha display beneath the Studio Monitor Volume Control knob now reads **PAN/BAL**; the

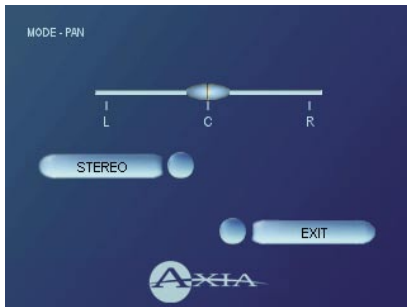


Figure 4-25: Aux Return Pan/Mode screen.

board op can use this to pan a mono source or adjust the balance of a stereo source.

- * **Stereo/Left/Right/Sum:** this control lets the board op switch source the selected Aux Return 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). This command appears on the left upper-middle Soft Key.
- * **Exit** (the bottom right Soft Key) returns you to the meter screen.
- » **PGM1, PGM2** and **Record:** these keys control which bus(es) the Aux Return is folded into. The top three right-hand Soft Keys (marked **PGM1**, **PGM2** and **RECORD**) control whether this Aux Return is sent to any or all of these three buses. Choosing their Soft Keys toggles the assignment **On** and **Off**; the onscreen indicators glow gold when an assignment is enabled.

Timer Mode Key

Pressing the **Timer Mode** key gives the user access to three different timer and clock modes.

The current Timer Mode is indicated just below the meters on the right-hand display screen. The modes are:

- » **Auto Reset.** The Event Timer resets to zero and counts upward whenever source that is set to start the timer is turned **On**. (Sources are programmed to **Auto-Start** the timer during Source Profile construction. See the section in Chapter Two en-

titled “Source Profile Setups” for details.)

- » **Auto Add.** For timing the total length of a program, regardless of sources being turned on or off. When **Auto Add** mode is active, the Event Timer will stop but not reset to zero when **Auto-Start** sources are turned **Off**, and will resume counting when those sources are turned back **On**.
- » **Manual.** In this mode, the right-hand column of Soft Keys become controls for manual **START**, **STOP** and **RESET** of the Event Timer.

Clock Options: The **Timer Mode** key is also used to set SmartSurface’s clock options. Press and *hold* the **Timer Mode** key for 5 seconds to enter Clock Options mode. The clock display on the left-hand meter screen turns red and reads **Clock Set**.

Important: Keep the Timer Mode key depressed during all clock set operations!

In **Clock Options** Mode, there is no on-screen menu; all functions are performed using the Soft Keys:

- » **FAST** advances clock time in ½-hour increments.
- » **SLOW** advances clock time in 1-minute increments.
- » **BACK** regresses clock time in 1-minute increments.
- » **HOLD** stops the clock’s advance while depressed, so that you can synchronize seconds. Releasing **HOLD** resumes normal clock advance.
- » **NTP:** If you entered an IP for an NTP (Network Time Protocol) server during initial setup, pressing this Soft Key toggles the connection to your NTP server **On** and **Off**. The Soft Key illumi-

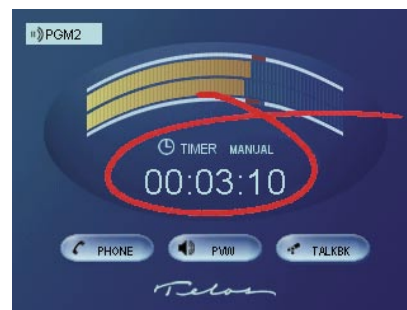


Figure 4-25: Right-hand meter screen with current Timer Mode circled.

nates when the NTP connection is **On**.

- » **12 H MODE / 24 H MODE:** Pressing this Soft Key toggles the clock between 12-hour and 24-hour (military-style) timekeeping.
- » **GMT:** Allows you to set your location relative to Greenwich Mean Time. This setting is necessary for correct time display when slaving the clock to an NTP server. To set, press this Soft Key repeatedly until the setting for your time zone is displayed.

Scrub Wheel

The **Scrub Wheel**, located at the bottom of the SmartSurface center section, can be used as a shortcut for monitor or headphone volume adjustments according to operator preference. The operator can change between functions by choosing **SCRUB** from the **Monitor Options** menu (see p. 28).

The Scrub Wheel can also be turned off completely.

Note: informal testing by Axia personnel has also determined that the **Scrub Wheel** can be used as a candy holder, paperclip depository, rubber-band receptacle, or as a spin-toy for nervous types.

Axia has achieved less-than-satisfactory results when attempting to use the **Scrub Wheel** as a drink-holder or ashtray, and we strongly recommend that you discourage your airstaff from attempting these uses.

Status Symbol Displays

Located in the same window as, but directly above the alphanumeric displays at the top of each SmartSurface channel, is a second display that contains Status Symbols. These icons inform the board operator of the presence and type of backfeed being sent to a particular channel.

The primary set of Status Symbols, shown in Figure 4-26, is shown when feeding phone hybrids and codecs. From top to bottom, the symbols indicate that the caller or remote is hearing:

1. A mix-minus of PGM-1,
2. A mix-minus of PGM-2,
3. A mix-minus of the PHONE bus, or

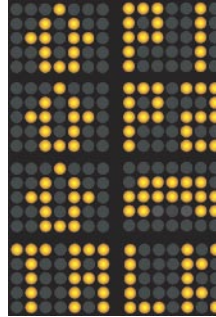


Figure 4-26:
Primary
Status Symbols

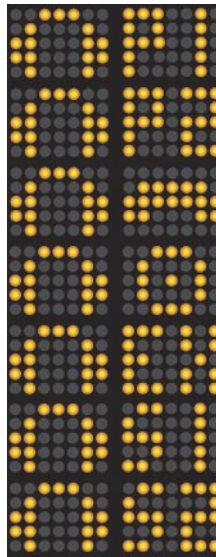


Figure 4-27:
Headphone
Status Symbols



Figure 4-28:
Remote Mute/Talk

4. The board op's talk channel.

A second set of Status Symbols (Figure 4-27) appears when the backfeed is being sent to a user's headphones using an Individual Headphone Feed. From top to bottom, they indicate:

1. PGM-1 mix-minus,
2. PGM-2 mix-minus,
3. PHONE mix-minus,
4. RECORD mix-minus,
5. EXTERNAL monitor,
6. AUX SEND 1, or
7. AUX SEND 2.

Finally, the Status Symbols shown in Figure 4-28 are displayed above a Mic channel when that Mic user has activated, via GPIO, the **Remote Talk** or **Remote Mute** functions described earlier in this chapter.

What's Next

One of SmartSurface'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. 🔄

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 SmartSurface 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 SmartSurface behaves, and which sources are loaded, and on which channels they appear.

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.

Best of all, since SmartSurface operates in a networked environment, these Show Profiles can be loaded in any studio in your facility, making studios truly interchangeable at last.

Creating A Show Profile

Build A Show

SmartSurface's Web interface, which you've used to construct Source Profiles, is also used to build Show Profiles.

When you first plug in your SmartSurface, it is equipped with a default Show Profile. This loads the first time you turn on your SmartSurface; it provides

the "blank slate" you'll use to construct custom Show Profiles. You might want to load the Default Show Profile from the Control Options menu located beneath SmartSurface's Soft Keys to make sure you're in "ready mode."

The first step in building a Show Profile is to set up SmartSurface using the Channel Options, Monitor Options and Control Options menus described in Chapter Four, "SmartSurface Operations." Get started by assigning a source to each SmartSurface 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 Keys or Studio Monitor Volume knob (flashing SOURCE_1) to select a source from the list; "take" the source and then press the **Options** key on Channel 2. You'll see the Studio Monitor Volume knob alpha change to read SOURCE_2 to confirm that you're now programming Channel 2.

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

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

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

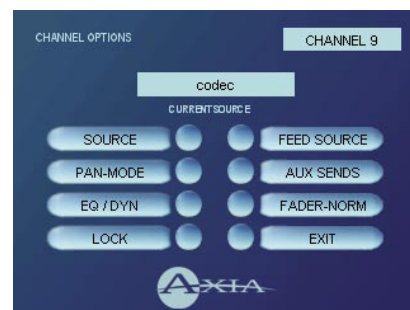


Figure 5-1: Channel Options screen.

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

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

Capture It!

It's now time to save your SmartSurface configuration.

Enter SmartSurface's IP address in your Web browser and, if you set a password during initial setup, enter it.

When the Main Menu appears, choose "Show Profiles." This is where you'll come whenever you need to capture, construct or administer a Show Profile.

In the Show Profiles screen, you'll find four options:

- » **Default Show Profile.** This is the "starter" profile that comes with SmartSurface. 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 SmartSurface to a "blank slate" condition.
- » **New Show Profile.** Choose this option when you want to construct an entire Show Profile completely from scratch.
- » **Capture Show Profile.** Takes a "snapshot" of the current board state and saves it for recall. This is the option we'll use in just a moment.
- » **Sort by Time.** Sorts saved Show Profiles in order of creation date / time.

Since you've already set up your SmartSurface, all that's left is to capture your settings for further use. Choose **Capture Show Profile** from the menu, and you'll see a new Show Profile appear in the list next to the Default profile. 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 SmartSurface itself. Let's go a little deeper into Show Profiles to find out exactly what's available to you.

Suggestion: As part of SmartSurface 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.

Let them set up the board exactly the way they want it. You can then use the Capture command to save their work. This way, talent gets the satisfaction of personalizing their own SmartSurface configuration — and saves you the work!

(Sneaky, aren't we?)

Note: Talent cannot save changes to shows themselves. This means that unless the Engineer deliberately "captures" changes, they will be discarded when loading any profile.

Show Profile Options

Although the ability to create Show Profiles using SmartSurface's "capture" function is powerful, there are more options available exclusively via the SmartSurface Web interface.



Figure 5-2: Show Profile Settings Menu

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

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

You'll see the menu shown in Figure 5-2. You can type a more descriptive name into the **Show Profile**

Name box at the top of the page – “Morning Drive”, perhaps – and click the “Save Changes” button to apply the new name.

Notice that there is an entry for each of SmartSurface’s 16 channels, as well as a link for **Auxiliary Send/Return Data, Monitor Section Data,** and **Record Mode.**

The Channel Description Screen

Click on any of the **Channel** links to examine the settings for a fader. Many of the options will seem familiar to you, and indeed you’ve seen some of 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 SmartSurface’s **Options** menus, prior to capturing the Show Profile.

Note: in addition to “active” options – choices that are actively made to SmartSurface settings when loading a new Show Profile – there is also a “passive” option for many items: **Retain Source Setting.** Choosing **Retain** allows the attributes for this option to load from the Source Profile you previously defined for this source.

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

- **Source ID:** The name of the source you assigned to this fader. You can change the source using the drop-down box.
- **Fader Mode:** Set channel to **Normal** or **Fader Start**, or **Retain Source Setting.**
- **Feed To Source Mode:** Specify **PGM-1 mix-minus, PGM-2 mix-minus, PHONE mix-minus** or **AUTO** for this source, or **Retain Source Setting.**
- **Auto-Start Timer:** Determines whether turning

this fader on will start/reset the Event Timer. You can **Enable, Disable** or **Retain Source Setting.**

- **Signal Mode:** Choose from **Stereo, Left** (left channel fed to both sides), **Right** (right channel fed to both sides), **Sum** (sum L+R to mono and feed to both sides) or **Retain Source Setting.**
 - **Signal Mode Locked:** **Enable** or **Disable** the user’s ability to change the Signal Mode, or **Retain Source Setting.**
 - **Panorama Position:** Trim Pan/Balance for this source, or **Retain Source Setting.**
 - **Phase:** Adjust for phase errors. May be set to **Normal, Invert Left, Invert Right, Invert Left And Right,** or **Retain Source Setting.**
 - **EQ Active:** Set source EQ for **Bypass, Active** or **Retain Source Setting.**
 - **EQ High Mode:** Switches high-frequency EQ mode between **Shelf** and **Bandpass,** or **Retain Source Setting.**
 - **EQ High Frequency:** You can specify the top of the High Frequency Shelf, or the middle frequency of the Bandpass filter (depending upon which mode was chosen for EQ High Mode), or **Retain Source Setting.**
 - **EQ Mid Frequency / EQ Low Frequency:** Choose the middle frequency of the Low and Middle EQ bands, or **Retain Source Setting.**
 - **EQ High Gain / EQ Mid Gain / EQ Low Gain:** dial in boost or cut for the three EQ bands, or **Retain Source Setting.**
-
- Note:** For details on SmartSurface EQ operations, please refer to Chapter Two, “Working With Sources.”
-

- **Assign to PGM1 / PGM 2 / RECORD / PHONE /**

AUX1 / AUX 2: Checking **ON** for any of these buses assigns the channel to that bus when the Show Profile is loaded. You can also choose **OFF** or **Retain Source Settings**.

- **Aux1 Pre/Post Fader:** Allows you to send this channel to the Aux Send 1 bus either **Pre-Fader**, **Post-Fader** or **Retain Source Settings**.
- **Aux2 Pre/Post Fader:** Allows you to send this channel to the Aux Send 2 bus either **Pre-Fader**, **Post-Fader** or **Retain Source Settings**.
- **Aux Send 1 Gain:** Turn this channel's feed to the AUX1 bus **Off**, specify boost or cut, or **Retain Source Settings**.
- **Aux Send 2 Gain:** Turn this channel's feed to the AUX2 bus **Off**, specify boost or cut, or **Retain Source Settings**.

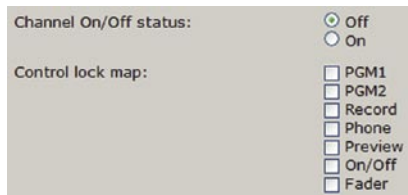


Figure 5-3: Channel lock controls.

- **Channel On/Off Status:** Allows you to turn this channel **On** or **Off** when the Show Profile is loaded.
- **Control Lock Map:** This option works similar to the **Lock** command in the SmartSurface **Channel Options** menu (see Chapter Four for details). Place a check mark in any of the boxes to “lock” the selections for this channel's bus assignments, On/Off state and fader position (Figure 5-3).

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

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

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

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

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

The Auxiliary Send Description Screen

In the same way that the Show Profile Channel Description screens mirror SmartSurface's **Channel Options** menus, the Auxiliary Send Description Screen captures the global Aux Send settings found in SmartSurface's **Control Options** menus.

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.



Figure 5-4: Detail of bottom-of-page links from SmartSurface Show Profile Settings page.

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

- **Aux Send 1 Master Gain:** Selecting **Off** turns off master gain adjustment for the Aux Send 1 bus. Select **Use** and enter a gain value between -25 dB and 10 db if desired, or select **Retain Existing Setting**.
- **Aux Send 1 On/Off Status:** Choose **Off** to turn Aux Send 1 off completely, **On** to turn it on, or **Retain Existing Setting**.
- **Aux Send 2 Master Gain:** Operates as described above, but for Aux Send 2.
- **Aux Send 2 On/Off Status:** As above, but for Aux Send 2.
- **Aux Return 1 Master Gain:** Selecting **Off** turns off master gain adjustment for the Aux Return 1 bus. Select **Use** and enter a gain value between -25 dB and 10 db if desired, or select **Retain Existing Setting**.

Note: The **Retain Existing Setting** option should be used carefully. If a Show Profile uses **Retain Existing Settings**, re-loading that profile will not overwrite the board's current settings! This means that if a board op gets themselves into trouble, loading a profile may not clear the problem!

Because of this, it's safest to specify values for every function when constructing a Show Profile – even though it means a little extra work in setup – unless you have a known need for the **Retain...** setting.

- **Aux Return 1 On/Off Status:** Choose **Off** to turn Aux Return 1 off completely, **On** to turn it on, or **Retain Existing Setting**.
- **Aux Return 1 Signal Mode:** Choose from **Stereo**, **Left** (left channel fed to both sides), **Right** (right channel fed to both sides), **Sum** (sum L+R to mono and feed to both sides) or **Retain Existing Setting**.
- **Aux Return 1 Panorama Setting:** Trim Pan/Balance for Aux Return 1, or **Retain Existing Setting**.
- **Aux Return 1 Assign to PGM1 / PGM2 / Record:** Choose **On** to fold Aux Return 1 into the PGM-1, PGM-2 and/or Record bus(es), respectively, or choose **Retain Existing Setting**.

- **Aux 1 Source ID:** If this drop-down box looks familiar, 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 device that will be providing source audio for the Aux 1 Return.
- **Aux Return 2 Master Gain:** As above, but for Aux Return 2.
- **Aux Return 2 On/Off Status:** As above.
- **Aux Return 2 Signal Mode:** As above.
- **Aux Return 2 Panorama Position:** As above.
- **Aux Return 2 Assign to PGM1 / PGM2 / Record:** As above.
- **Aux 2 Source ID:** As above.
- **Save Changes** button applies any changes made to the Auxiliary Send Description and returns you to the Show Profile Settings menu.
- **Back to Show Profile** link takes you back to Show Profile Settings without saving changes.

Monitor Section Data

The Monitor Section Data screen captures the Monitor settings found in SmartSurface'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 SmartSurface 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.

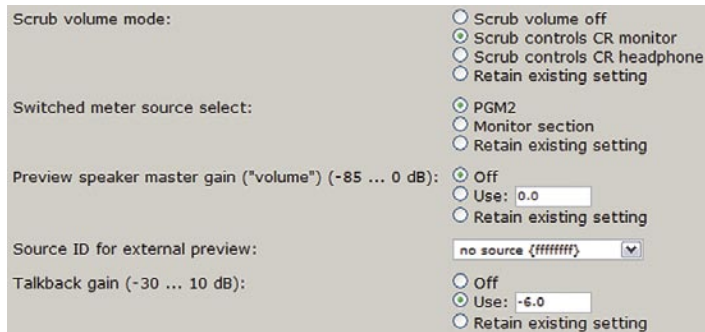


Figure 5-5: Monitor section general options

General Monitor Options

- **Scrub Volume Mode:** Assign the Scrub Wheel to control **Control Room Monitor** volume or **Control Room Headphone** volume, choose **Scrub Volume Off** to have the Scrub Wheel assigned to nothing, or choose **Retain Existing Setting**.
- **Preview Interlock Mode:** By default, SmartSurface’s **Preview** key acts like a latching switch, *Momentarily* pressing any channel’s **Preview** key assigns that channel to the Preview bus; pressing it again removes the channel from Preview. To enable an operator to quickly listen to various sources, the Preview function is interlocked: pressing **Preview** on any channel will remove any other channel from Preview mode. Operators can momentarily disable this interlock to add more than one source to the **Preview** mix (see the description of the Channel Preview Key in Chapter 4, “SmartSurface Operations”, for details).

However, you may wish to disable the Preview Interlock on a global basis. Choosing **Enabled** for this setting retains the default operating style; choosing **Disabled** shuts off the interlock, so that multiple sources may be added or removed from the **Preview** mix without additional button presses.

- **Switched Meter Source Select:** Choose whether the right-hand screen’s meters will display levels for the Program-2 bus (**PGM2**), the source feeding the Control Room Monitors (**Monitor Section**) or **Retain Existing Setting**.
- **Preview Speaker Master Gain:** Lets you **Use** up to -85 dB of attenuation on the Control Room Pre-

view speakers, turn gain adjustment **Off**, or **Retain Existing Setting**.

- **Source ID for External Preview:** This drop-down box lets you specify the source that will be fed to the monitors when External is selected for monitoring (typically an off-air monitor or a dedicated headphone processing loop).

- **Talkback Gain:** Apply up to -30 dB attenuation or 10 dB gain to the level of the Talk To... channels. You can also turn gain adjustment **Off** or **Retain Existing Setting**.

Control Room Monitor Options

- **Monitor Assignment:** Choose to load **PGM1**, **PGM2**, **Record**, **Phone**, **AUX1**, **AUX2** or an **External Source** to the Control Room Monitor 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 Monitor Assignment list above.

Note: The word “External” is used for setup purposes only; when a source is assigned to the **External** input, the name of that source will appear in SmartSurface’s Monitor Selection Screen.

- **CR Monitor Master Gain:** Allows you to **Use** as much as -85 dB of gain reduction on the audio signal sent to the Control Room Monitor Speakers. You can also turn gain control **Off** or **Retain Existing Setting**.
- **Signal Mode: CR Monitor** lets you set the audio feed to the Control Room Monitor Speakers to **Stereo**, **Left**, **Right**, **Sum** or **Retain Existing Setting**.
- **CR Monitor Dim Gain:** Specify the amount of vol-

ume by which the Control Room Monitors 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 the Control Room Monitor a channel number, effectively making it a “source” to which logic commands – to activate Mute, Dim, tally lights – can be associated. You’ll enter this channel number in the GPIO Setup web page to assign a GPIO port for these logic functions. See Chapter Three, “GPIO Configuration,” for more information.

Control Room Headphone Options

- **Monitor Assignment:** Choose to load **PGM1**, **PGM2**, **Record**, **Phone**, **AUX1**, **AUX2** or an **External Source** to the Control Room headphones 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 headphone channel.
- **Source ID for EXTERNAL Input:** Use this drop-down box to choose what source will be auditioned in the Control Room headphones when **External** is chosen from the Monitor Assignment list above.
- **CR Headphone Master Gain:** Allows you to **Use** as much as -85 dB of gain reduction on the audio signal sent to the Control Room Monitor Speakers. You can also turn gain control **Off** or **Retain Existing Setting**.
- **Signal Mode: CR Headphone:** Lets you set the audio feed to the Control Room headphones to **Stereo**, **Left**, **Right**, **Sum** or **Retain Existing Setting**.
- **CR Headphone Independent:** Lets you choose whether, for this Show Profile, the Control Room headphones should be fed the same audio as the Control Room Monitor Speakers (**Follow Monitors**) or if the board op is allowed to select a source for headphones independent of the Monitor Speakers (**Use Headphone Source Select**). You may also **Retain Existing Setting**.
- **Preview-in-Headphone Mode:** Choices are **Off**, **Normal**, **Auto** and **Retain Existing Setting**. If set to **Off**, sources assigned to the Preview bus will only be heard in the Control Room Preview speakers, leaving the board operator’s headphones to continuously monitor the selected source. In **Normal** mode, audio from the Preview bus will feed the board op’s headphones, *in stereo*, whenever **Preview** is active, interrupting the selected source. **Auto** mode sums Preview audio to mono and feeds it to the *right headphone*; the selected source audio is also summed to mono and fed to the *left headphone*.
- **CR Headphone EQ Active:** Set headphone EQ for **Bypass**, **Active** or **Retain Existing Setting**.
- **CR Headphone EQ High Mode:** Switches high-frequency EQ model between **Shelf** and **Bandpass**, or **Retain Existing Setting**.
- **CR Headphone EQ High Frequency:** You can specify the top of the High Frequency Shelf, or the middle frequency of the Bandpass filter (depending upon which model was chosen for EQ High Mode), or **Retain Existing Setting**.
- **CR Headphone EQ Mid Frequency / EQ Low Frequency:** Choose the middle frequency of the Low and Middle EQ bands, or **Retain Existing Setting**.
- **CR Headphone EQ High Gain / EQ Mid Gain / EQ Low Gain:** dial in boost or cut for the three EQ bands, or **Retain Existing Setting**.

Studio Monitor Options

- **Monitor Assignment:** Choose to load **PGM1**, **PGM2**, **Record**, **Phone**, **AUX1**, **AUX2** or an **External Source** to the Studio Monitor 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 Studio monitor.
- **Source ID for EXTERNAL Input:** Use this drop-down box to choose what source will be heard when

External is chosen from the Monitor Assignment list above.

- **Studio Monitor Master Gain:** Allows you to **Use** as much as -85 dB of gain reduction on the audio signal sent to the Control Room Monitor Speakers. You can also turn gain control **Off** or **Retain Existing Setting**.
- **Studio Monitor Dim Gain:** Specify the amount of volume by which the Control Room Monitors will dim when **Talk To...** or **Preview** is in use. **Use** as much as -30 dB of cut, turn the option **Off**, or **Retain Existing Setting**.
- **GPIO Channel for Studio Monitor:** In this box, you assign the Studio Monitor a channel number to associate a logic port. You'll enter this channel number in the GPIO Setup web page to assign a GPIO port for these logic functions. See Chapter Three, "GPIO Configuraton," for more information.

Save and Exit

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

Record Mode

SmartSurface's Record Mode (described in Chapter Four) acts as a one-button "macro," assigning channels to the Record bus, muting the monitors, starting a recording device.

However, unlike most other SmartSurface options,

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

Figure 5-6: Record Mode Mixing section

Record Mode cannot be configured from the surface itself. You must define Record Mode here, in the Show Profiles web page.

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

Record Mode Configuration

- **Record Mode Activation:** If **Disabled** is chosen, the board op will not be able to use **Record Mode** while this Show Profile is loaded. Choose **Enabled** to allow Record Mode to be used.
- **GPIO Channel for Recorder Control:** Enter the channel number of the GPIO port assigned to your chosen recording device, if you want SmartSurface's **Transport Keys** to be active during **Record Mode**. See Chapter Three, "Configuring GPIO," for a complete reference of SmartSurface GPIO functions.
- **GPIO Channel for Hybrid Control:** Enter the channel number of the GPIO port assigned to your telephone hybrid, if you want **Record Mode** activation to control your hybrid's **Answer** and **Drop** functions, among others. Refer to Chapter Three for a complete reference of SmartSurface GPIO functions.

Mixing

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

CR Monitor

CR Monitor Source: Usually, this is set to monitor the Record bus during **Record Mode**. You can also choose to load **PGM1**, **PGM2**, **Phone**, **AUX1**, **AUX2** or an **External Source** to the Control Room Monitor channel when **Record Mode** is activated. You can also choose **No**

Source and **Retain Existing Source**. The Control Room Monitor source will be returned to its previous selection when **Record Mode** is ended.

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

CR Headphones

- **CR Headphone Source:** Choose to load **PGM1, PGM2, Record, Phone, AUX1, AUX2** or an **External Source** to the Control Room headphones channel when **Record Mode** is activated. You can also choose **No Source** and **Retain Existing Source**. The Control Room headphone source will be returned to its previous selection when **Record Mode** is ended.
- **CR Headphone External Source ID:** Use this drop-down box to choose what source will be heard when **External** is chosen from the CR Headphone Assignment list above.

Studio Monitor

- **Studio Monitor Source:** Choose to load **PGM1, PGM2, Record, Phone, AUX1, AUX2** or an **External Source** to the Studio Monitor channel when **Record Mode** is activated. You can also choose **No Source** and **Retain Existing Source**. The Studio Monitor source will be returned to its previous selection when **Record Mode** is ended.
- **Studio Monitor External Source ID:** Use this drop-down box to choose what source will be heard when **External** is chosen from the Studio Monitor Assignment list above.

Save and Exit:

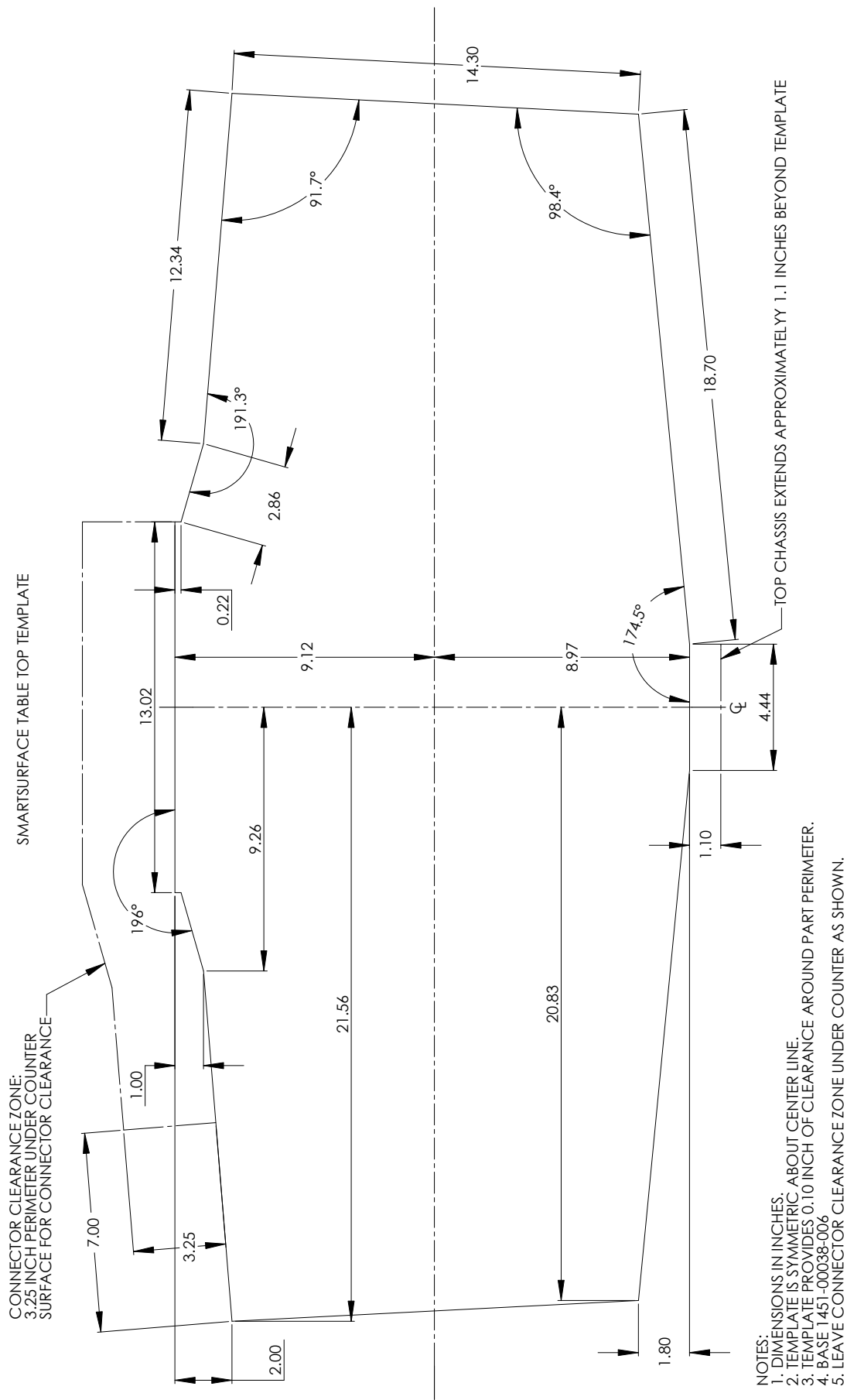
- **Save Changes** button applies any changes made to the Record Mode Setup and returns you to the Show Profile Settings menu.
- **Back to Show Profile** link takes you back to Show

Profile Settings without saving changes.

splicing block, razor

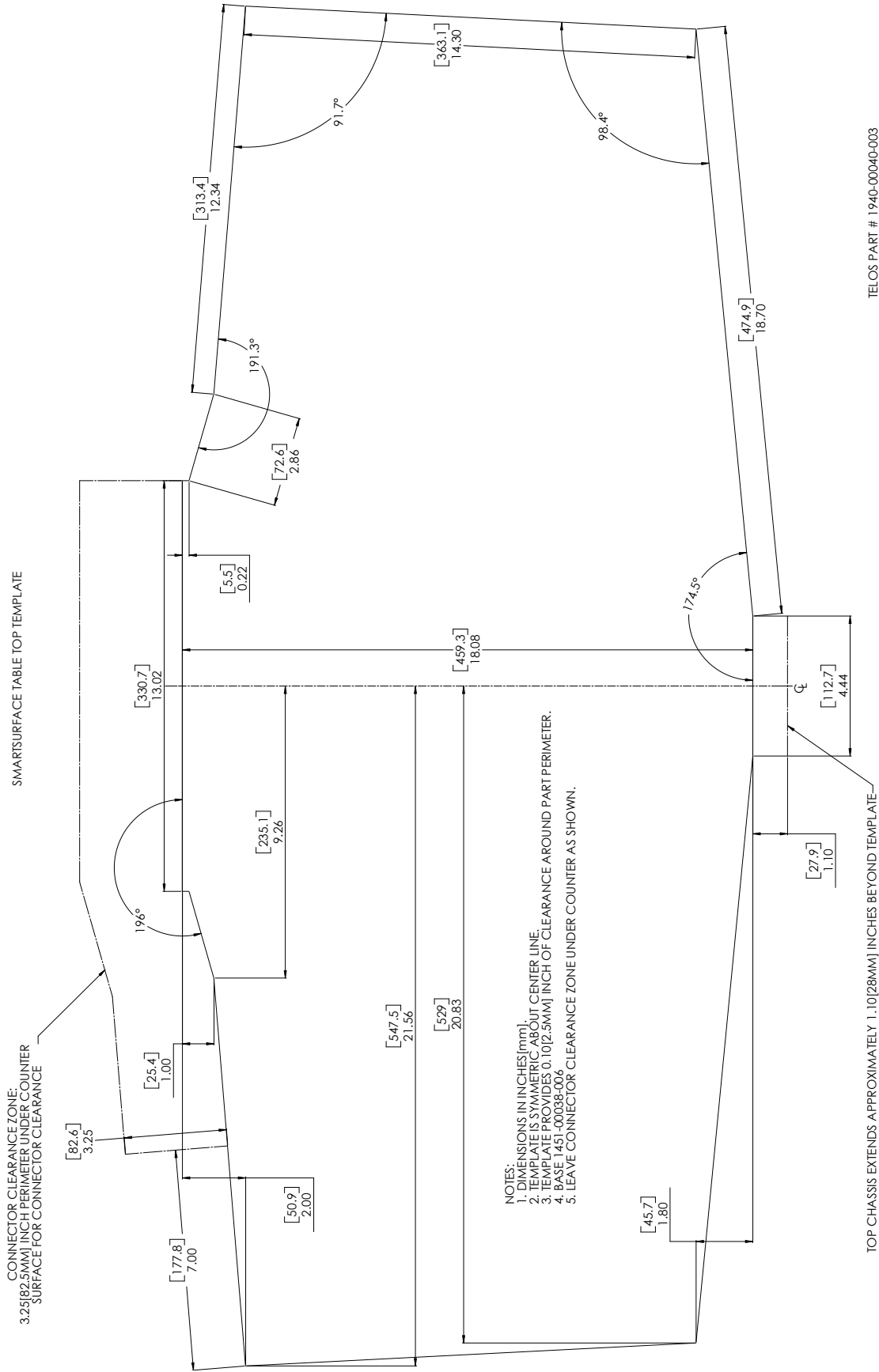
old ways are long surpassed now

Ampex nevermore.



SSTEMPLATE006_2, REVISION 26JAN04

Cutout Dimensions, In Inches



TELOS PART # 1940.00040-003
REVISION 18AUG05

Cutout Dimensions, In Centimeters

Appendix A: Table of Inputs and Outputs

<u>Main Outputs</u>	<u>Type</u>	<u>Comments</u>
Program-1	Stereo	Main stereo bus (Usually referred to as Program)
Program-2	Stereo	Second stereo bus (Sometimes referred to as Audition)
Record	Stereo	Third stereo bus (Sometimes referred to as Utility), signal post fader, before on/off.
Phone	Stereo	Fourth stereo bus (Sometimes referred to as Auxiliary), signal pre-fader, before on/off.
<u>Aux Inputs and Outputs</u>		
<u>Aux Inputs and Outputs</u>	<u>Type</u>	<u>Comments</u>
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
<u>Monitor-related Outputs</u>		
<u>Monitor-related Outputs</u>	<u>Type</u>	<u>Comments</u>
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) headphones, source same as main, fixed level output, with talkback
Studio Headphone Guest	Stereo	Studio (guest) headphones, source same as main, fixed level out, 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
<u>Monitor-related Inputs</u>		
<u>Monitor-related Inputs</u>	<u>Type</u>	<u>Comments</u>
CR Monitor External Input	Stereo	Allows an external source to be monitored in the CR monitors
CR HP External Input	Stereo	Allows an external source to be monitored in the CR headphones
Studio Monitor External Input	Stereo	Allows an external source to be monitored in the Studio monitors
External Preview Input	Stereo	Allows an external path into the Preview speakers
Source Preview Input	Stereo	Allows any audio source systemwide to be assigned to the Preview speakers
<u>Source Inputs</u>		
<u>Source Inputs</u>	<u>Type</u>	<u>Comments</u>
Microphone Input <i>n</i>	Mono	
Analog Line Input <i>n</i>	Stereo	
Digital Line Input <i>n</i>	Stereo	
<u>Source-related Outputs</u>		
<u>Source-related Outputs</u>	<u>Type</u>	<u>Comments</u>
Feed-to-Source A <i>n</i>	Mono	Mono mix-minus output feeds the left side of a stereo connection. "Talk to..." function enabled.
Feed-to-Source B <i>n</i>	Mono	Mono mix-minus output feeds the right side of a stereo connection. "Talk to..." function disabled.
Individual Hdphone Feed <i>n</i>	Stereo	Individual headphone feed (for talent and guest mics) with *Talk to x* function enabled.

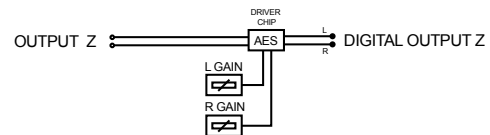
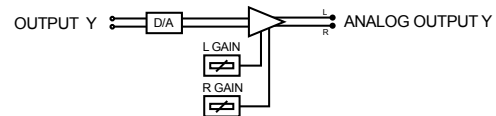
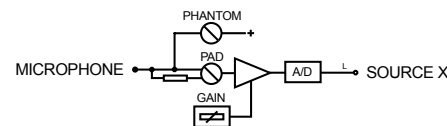
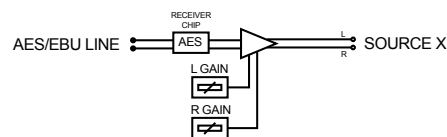
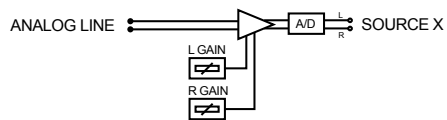
*Harsh bells destroy sleep:
the main transmitter is down.
“Raise the Aux, you fool!”*

Appendix B: Block Diagrams

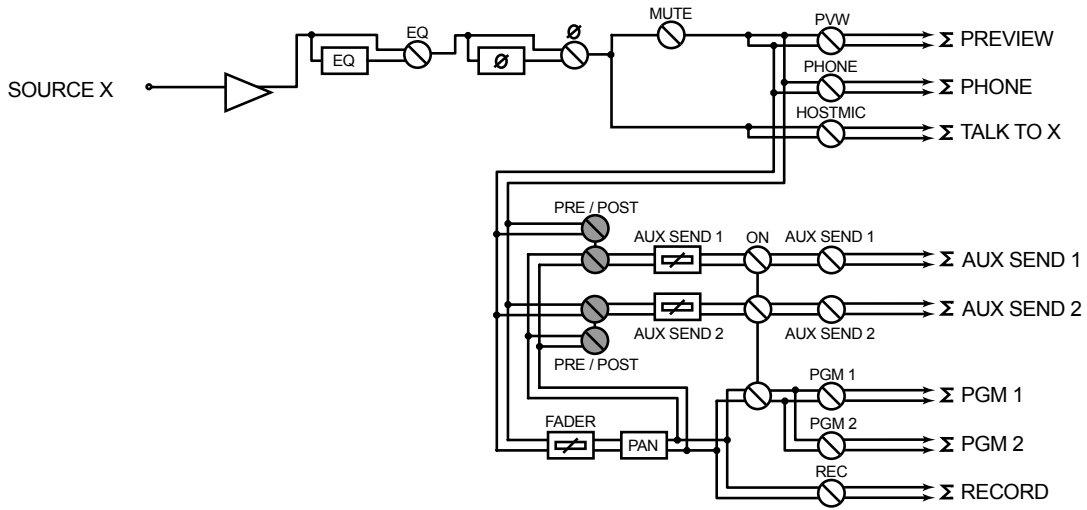
Legend



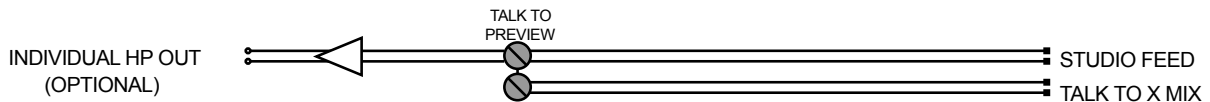
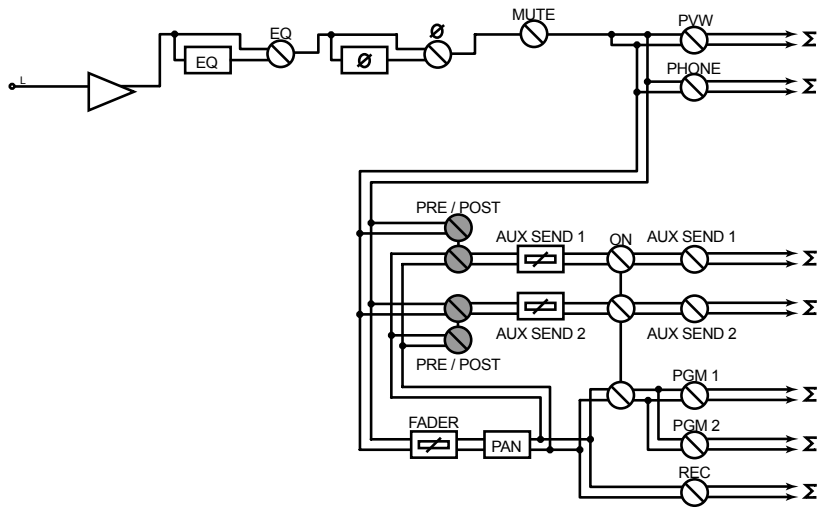
Input & Output Interface Blocks (External to Engine)



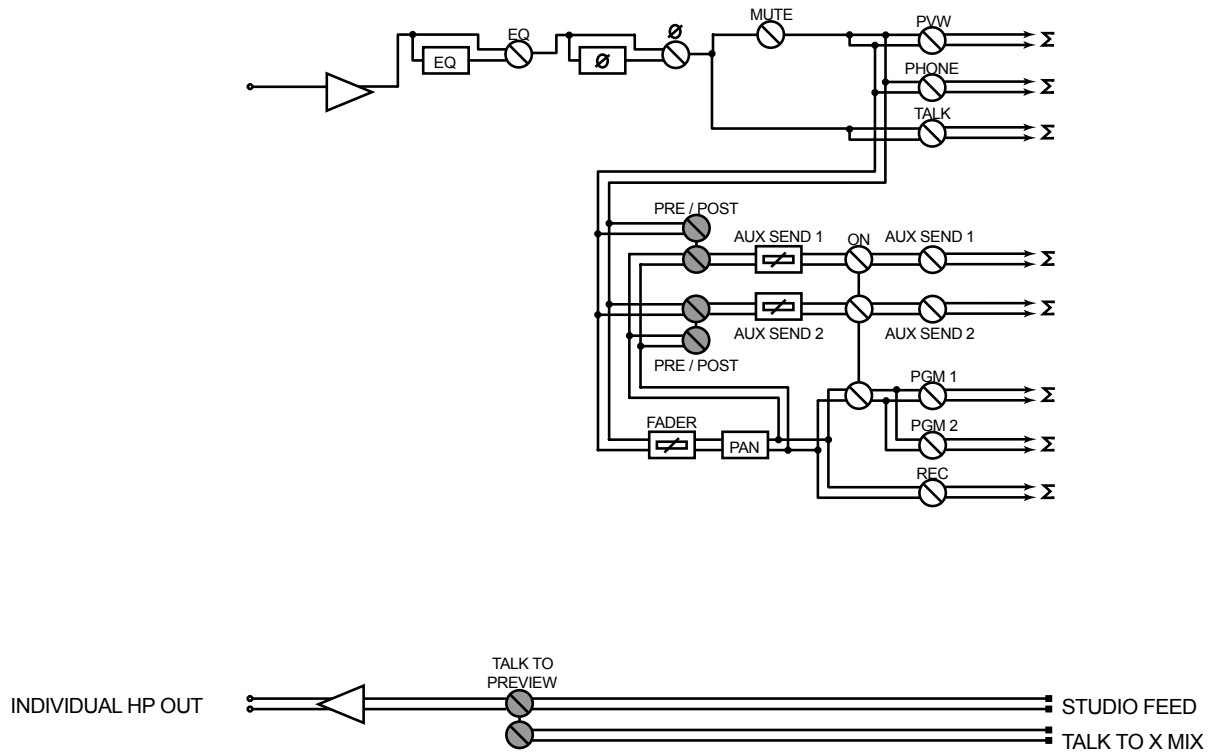
Operator / Control Room Producer Microphone Channel Block



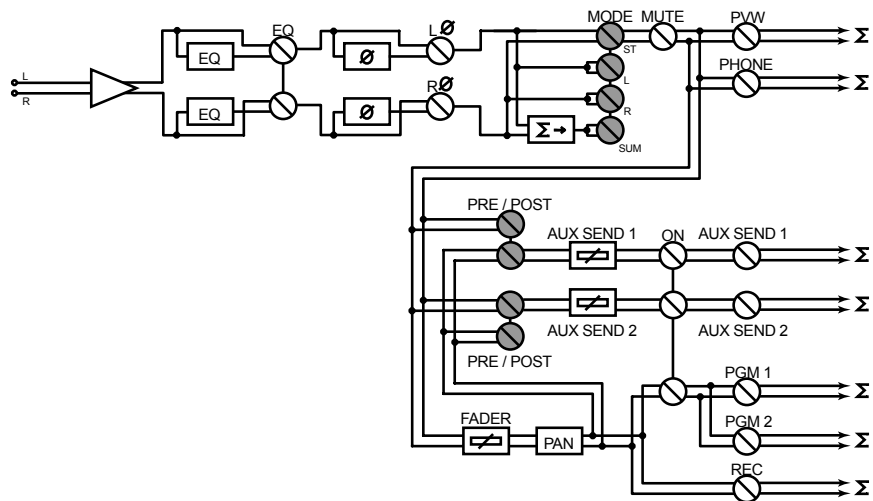
Control Room Guest Microphone Channel Block



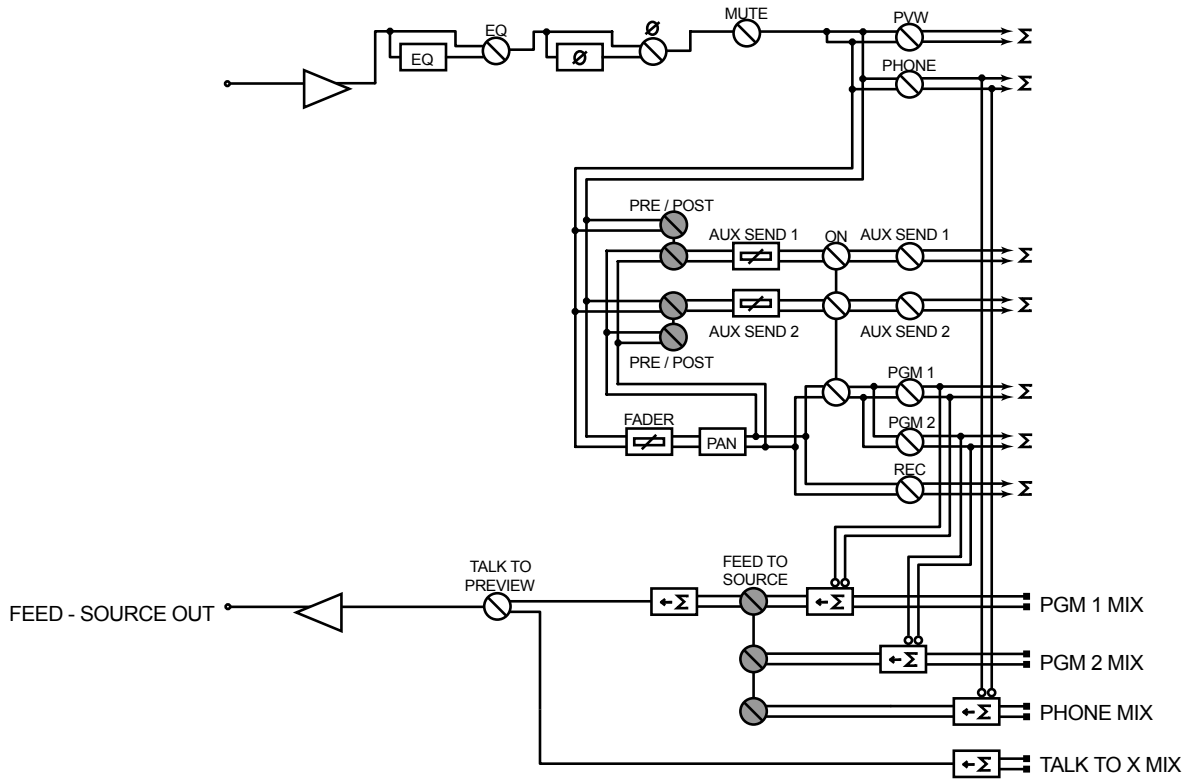
Studio Guest Microphone Channel Block



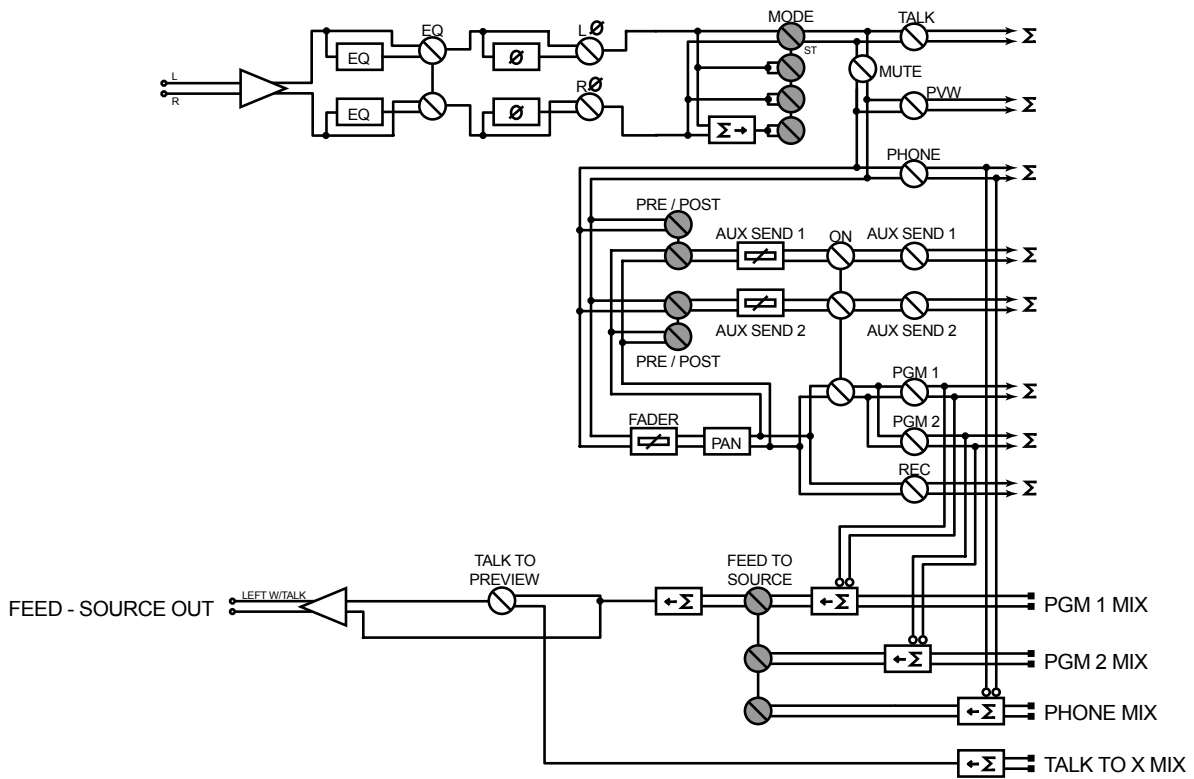
Line Input Channel Block



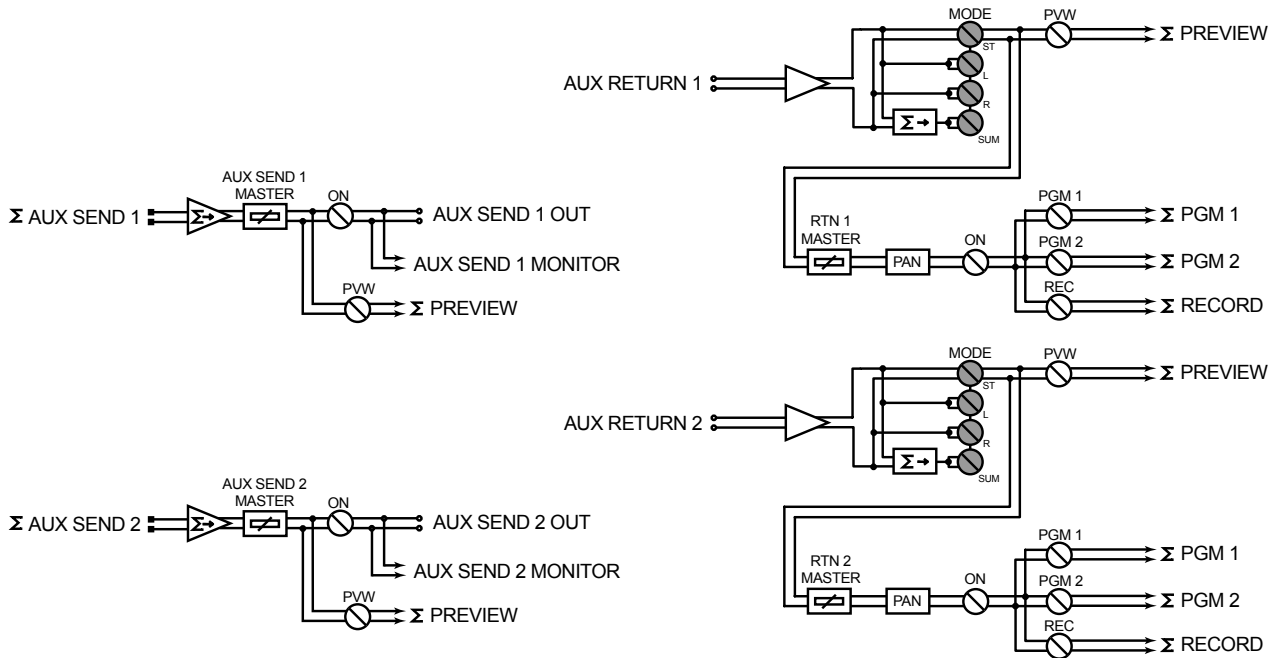
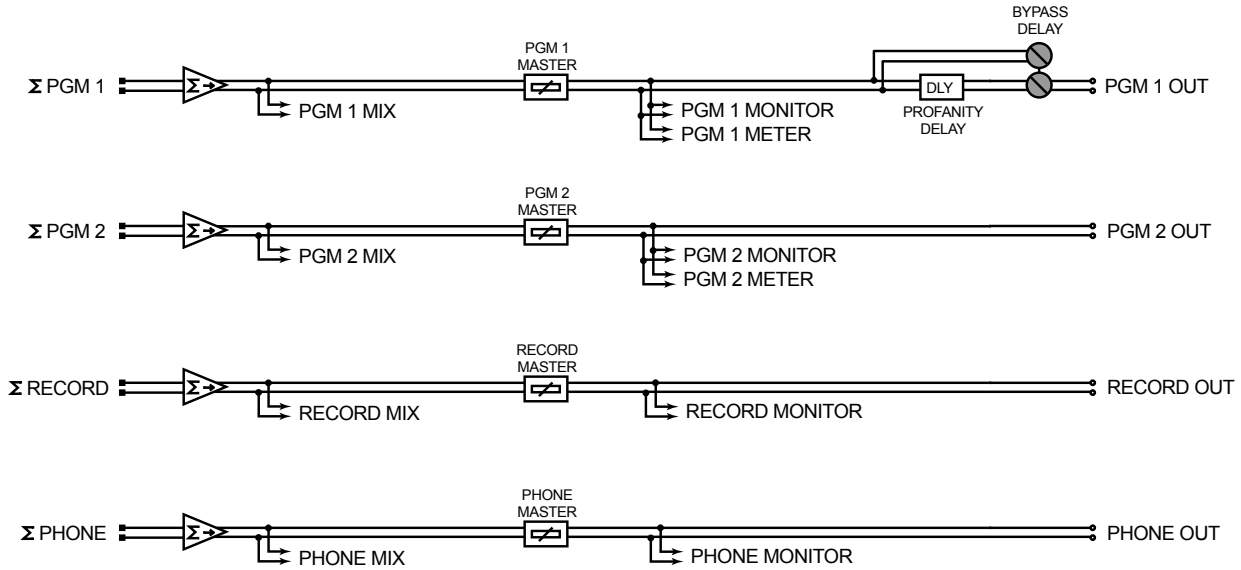
Phone Channel Block



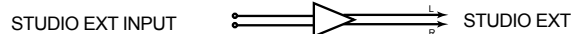
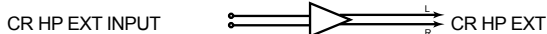
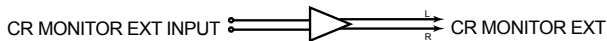
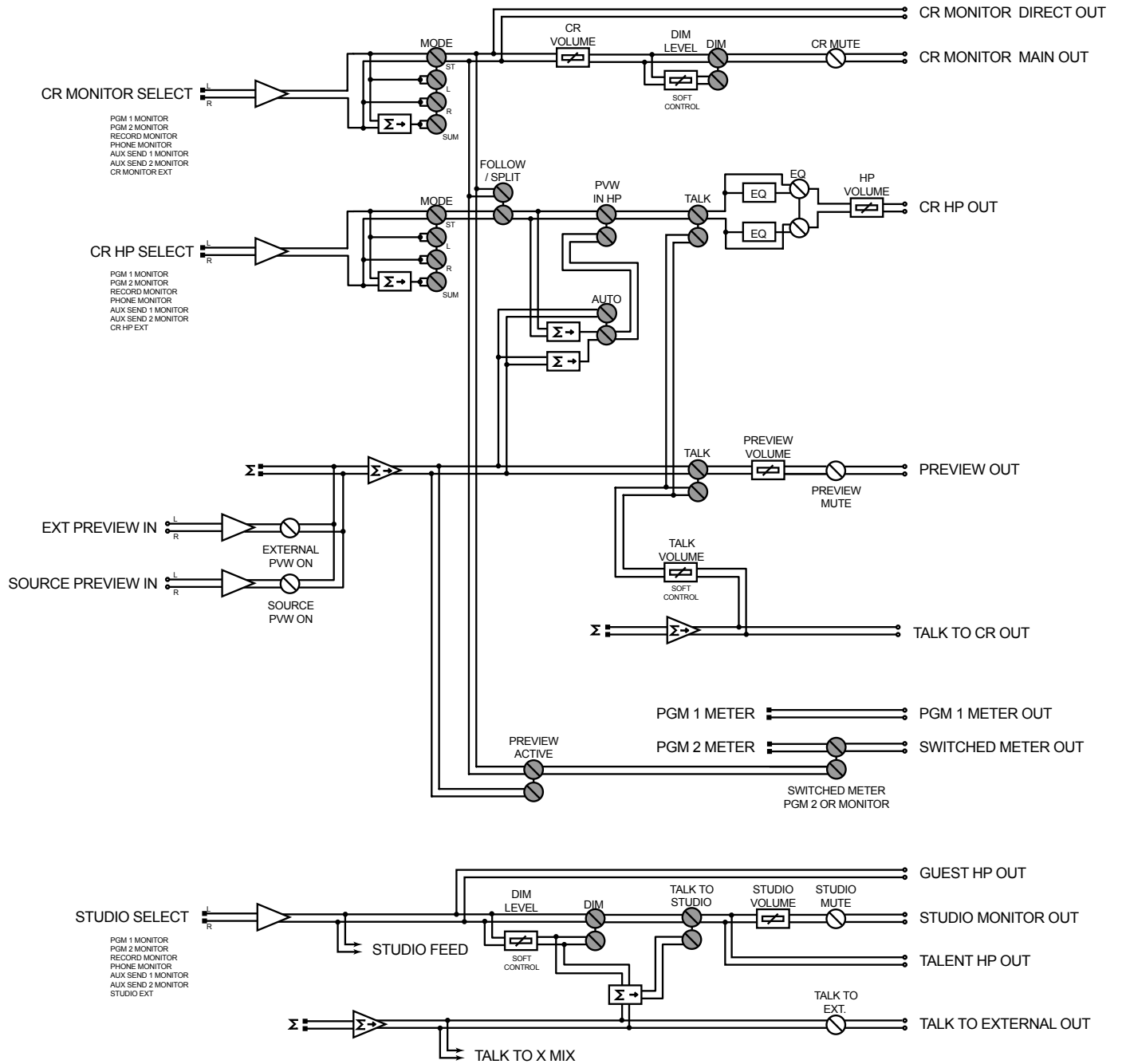
Codec Channel Block



Output Section and Aux Send Section Blocks



Monitor Section Block



Appendix C: FAQ / Diagnostics / Maintenance

This appendix contains answers to some frequently-asked SmartSurface 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 SmartSurface?

Press and hold the **Control Options** key found in the center section of SmartSurface just below the Soft Keys, then press and hold the three **Talk To...** keys just below the meters. Hold these keys for 5 seconds; the right display screen will change to the IP Setup screen.

Q: The SmartSurface web interface asks me for a password, but I haven't set one.

SmartSurface 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 "user" (without quotes, of course).

Q: I get a "Page Not Found" error when I try to connect to SmartSurface 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 the 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 SmartSurface'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.

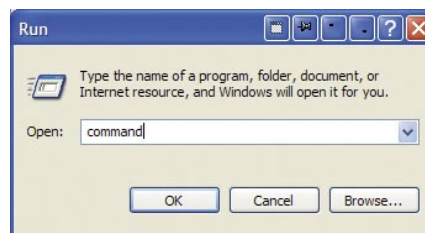


Figure C-1: starting a command box

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.
3. 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 SmartSurface.

Explanation: The **IP address** is usually expressed as four decimal numbers, each repre-

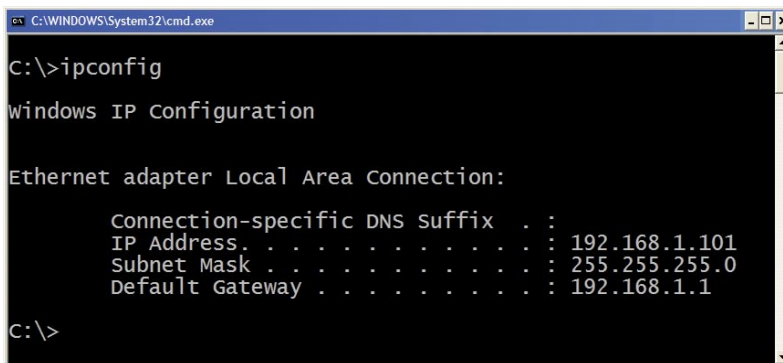


Figure C-2: IPCONFIG command displays computer's IP settings

sending 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 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](http://tinyurl.com/64bb5), available at <http://tinyurl.com/64bb5>.

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 SmartSurface; 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 SmartSurface. If they are not, change your computer’s Subnet Mask settings to match that of SmartSurface.

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 SmartSurface, you should be able to access your SmartSurface’s web page. If you’ve confirmed network settings and still cannot access your SmartSurface, “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 SmartSurface’s IP address).

2. If your SmartSurface is responding to commands, the **ping** command will produce a screen resembling Figure C-3.

```

C:\WINDOWS\System32\cmd.exe
C:\>ping 192.168.1.1

Pinging 192.168.1.1 with 32 bytes of data:

Reply from 192.168.1.1: bytes=32 time<1ms TTL=150
Reply from 192.168.1.1: bytes=32 time<1ms TTL=150
Reply from 192.168.1.1: bytes=32 time<1ms TTL=150
Reply from 192.168.1.1: bytes=32 time<1ms TTL=150

Ping statistics for 192.168.1.1:
    Packets: Sent = 4, Received = 4, Lost = 0 (0% loss),
    Approximate round trip times in milli-seconds:
        Minimum = 0ms, Maximum = 0ms, Average = 0ms
  
```

Figure C-3: Successful PING. SmartSurface is responding normally.

Results like these, showing packets sent and returned, indicate that your SmartSurface is active and responding to requests. However, if the **ping** command produces results like these...

```

C:\WINDOWS\System32\cmd.exe
C:\>ping 192.168.1.1

Pinging 192.168.1.1 with 32 bytes of data:

Request timed out.
Request timed out.
Request timed out.
Request timed out.

Ping statistics for 192.168.1.1:
    Packets: Sent = 4, Received = 0, Lost = 4 (100% loss),
  
```

Figure C-4: PING command showing unresponsive SmartSurface

...your SmartSurface is not responding and must be restarted. Disconnect the SmartSurface 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 SmartSurface on the same LAN or VLAN. Try connecting the two directly with an Ethernet crossover cable (for greater detail on cable diagrams, read the *Introduction to Livewire: System Design Reference and Primer*, available from www.AxiaAudio.com/tech/.)

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 SmartSurface?

Press and hold the **Timer Options** key near the Scrub Wheel. Don't let go. The right-hand display screen will change when it enters settings mode. Look at right display and soft keys for the various choices; keep the **Timer Options** key depressed while you make changes.

Q: How do I make my On-Air lamp illuminate when the Operator's mic is turned on?

You will use one of your GPIO Node Ports to activate the relay for your "on-air" lamp. Here's how:

1. Find the Channel Number previously assigned to your StudioEngine's **CR Monitor** output. If you did not use a Channel Worksheet to keep track of assigned channel numbers, you can find this number in your StudioEngine configuration web page. Use a Web browser on a computer connected to your Axia network, enter the IP address of your StudioEngine in the browser's address bar, and navigate to the "Program & Monitor Outputs" page. You'll find the **CR Monitor** entry about halfway down the page.
2. Enter the IP Address of your GPIO Node in your Web browser. Navigate to the "Configuration" page and choose the GPIO port you'll use to activate your on-air lamp relay. Enter the **CR Monitor** Channel Number in the field adjacent to that port's name.
3. Enter the IP Address of your SmartSurface in your Web browser and navigate to the "Show Profiles" page. Click on the name of any Show Profile and then choose "Monitor Data Section." When the Monitor Data page loads, find the section of the page titled "CR Monitor", enter the **CR Monitor** channel number in the field labeled

CR GPIO.

4. Repeat Step 3 for each Show Profile to ensure "on-air" lamp operation no matter which Show Profile is loaded.

Refer to the GPIO pinout diagrams in Chapter 3, "Configuring GPIO", to determine which pins on your GPIO port to connect to your lamp relay.

Lamp & Display Diagnostics

This section provides instructions for testing SmartSurface lamps and screens. Follow these instructions:

1. Enter SmartSurface's IP Setup menu. Press and hold the **Control Options** key, then press and hold the three **Talk To...** keys just below the meters. Hold these keys for 5 seconds; the right display screen will change to the IP Setup screen.
2. Notice that the upper left Soft Key display reads **TEST**. Press this to enter Test Mode.

Note: Test Mode has six cycles, which you advance by using the Soft Key labeled **NEXT**. Note that you can exit Test Mode at any time by pressing the upper right Soft Key, labelled **EXIT**.

3. When **TEST** is depressed, all lamps on SmartSurface are turned on to verify working status. These include all fader **On** and **Off** keys and all program bus assignment keys and **Options** keys. The alpha-numeric displays adjacent to the Soft Keys also flash.
4. In between flashes, you'll see that the upper left Soft Key is now labelled **NEXT**. Depress this key to move to the next test cycle.

The cycles are:

- » Lamp Test - all bus assignment keys, **Options** keys and fader channel **On** and **Off** keys illuminate.
- » Status Symbols display test - all Status Symbol display segments flash on and off to verify operation.

- » Fader Channel / Monitor alpha test - all segments of the alpha-numeric displays above each fader channel and below each Monitor Volume knob flash on and off to verify operation.
- » Right Screen Test - the right QVGA display screen completely red, green and blue to test color circuitry. This pattern repeats until you advance to the next test cycle.
- » Left Screen Test - same as previously described for the right display screen.
- » Soft Key Alpha Test - the alpha-numeric displays adjacent to the Soft Keys flash all segments on and off.

This is the last test cycle, You may repeat the cycles by using the **NEXT** key if you wish, or return to normal operating mode using the **EXIT** key.

Fader Cleaning Procedures

SmartSurface’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 SmartSurface. 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.

SmartSurface faders can be serviced “hot.” If you disconnect a fader, the StudioEngine will retain the fader’s most recent settings until the hardware is reconnected.

Tools and supplies you will need to remove and service a fader:

- » A $\frac{1}{16}$ ” hex wrench
- » A jeweler’s screwdriver ($\frac{5}{64}$ ” or smaller)
- » Cotton swabs
- » Dow Corning 510 lubricating oil

Fader Disassembly and Cleaning

1. With SmartSurface closed, identify the fader you wish to service and open SmartSurface by applying gentle upward pressure on each front corner. If the surface is locked closed, insert a #00 Philips-head screwdriver in the lock holes located in the front corners of SmartSurface’s bumper and rotate counter-clockwise to unlock. After a small amount of travel, the internal lifting mechanism will assist you in opening SmartSurface and will keep it open while you work.



Figure C-5: Locating fader connections

2. Locate the three-wire cable soldered to the terminals of the fader you’re going to service, and trace it to its connection point on the surrounding circuit board (Figure C-5).

Using gentle pressure, pull the red cable connector from the circuit board pins. Lift the white retaining tab slightly to “unlock” it.

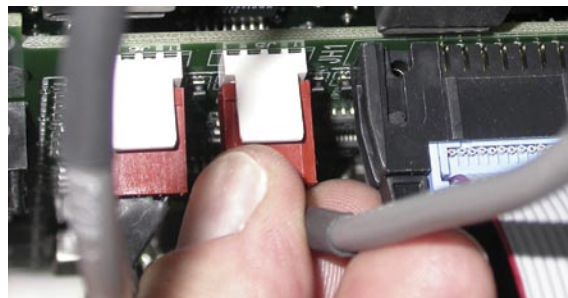


Figure C-6: Unplugging the fader connecting cable

3. Now that the fader is disconnected, close the surface and locate the two $\frac{1}{16}$ ” hex screws at the top and bottom of the fader slot. Use a hex wrench to loosen these screws, taking care that they do not

drop into the fader slot.

4. Raise the surface partway. Hold the fader body from below while gently pulling up on the black fader knob. When the fader knob detaches, the fader assembly will drop out into your hand from below.



Figure C-7: Loosening hex screws

5. Lay the fader assembly on your work surface, label-side up. Remove the snap-on fader assembly 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-8). Use a jeweler's screwdriver to gently pry off the cover.

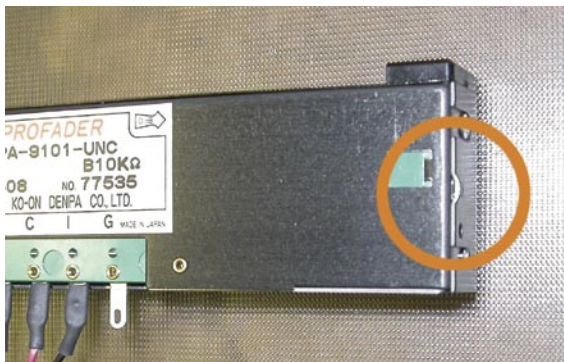


Figure C-8: Pry point on fader cover

6. 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.

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.

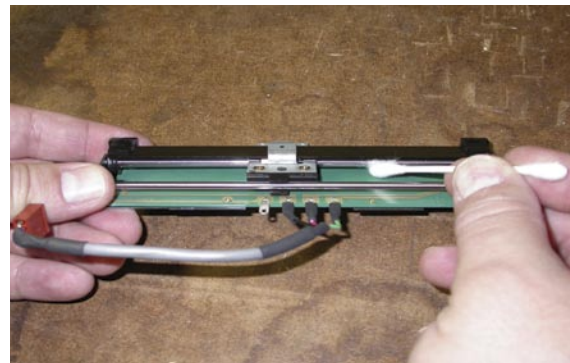


Figure C-9: Cleaning the fader glide rails

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 lead to its mating pins on the circuit board inside the Surface.
6. Close the SmartSurface by pulling gently downward on the front corners of the surface; take care that the lifting assist mechanism does not “snap” the surface shut. If desired, use a #00 Philips-head screwdriver to engage the closure locks located in the forward corners of the SmartSurface bumper.

Appendix D: Phone Controller Installation

If your station uses a Telos Talkshow System, you may have purchased an optional SmartSurface Console Director for on-the-board control of your studio phone system. This Appendix will walk you through installation. You don't need to power off the Surface during installation.

Tools and supplies you will need to complete this installation:

- » A 2 mm. hex wrench
- » A pair of diagonal cutters
- » Two or more small wiring tie-strips
- » #0 Philips-head screwdriver (optional)
- » A CAT.5 Ethernet cable of sufficient length to connect your SmartSurface with your phone system.

If you have not opened the shipping box containing your Console Director, please do so now. Inside you should find:

- » (1) Console Director, static-wrapped
- » (1) Ethernet connection cable

If any parts are missing or appear damaged, please contact Telos Systems Support immediately using the contact information found on Page iii of this manual.



Figure D-1: Removing the expansion bay blanking panel

Installation

First, decide whether you will install your Console Director in the left- or right-hand expansion bay of your SmartSurface. (We've chosen to use the right-hand bay.)

Use a 2 mm. hex wrench to remove the six round-head hex screws at the edges of the blanking panel, as shown in Fig-



Figure D-2: Installed Console Director panel

ure D-1. When removed, place the screws and blanking panel in a safe location.

Remove the Console Director from its protective static wrap, taking care to ground yourself (to discharge static electricity) before doing so.

Gently lower the Director into the empty SmartSurface bay, taking care not to catch any exposed components on the bay's metal edges. Use the 2 mm. hex screws saved from removal of the blanking panel to fasten the Director panel to the SmartSurface.

Now open the SmartSurface by applying gentle upward pressure on each front corner. If the surface is locked closed, insert a #00 Philips-head screwdriver in the lock holes located in the front corners of SmartSurface's bezel and rotate counter-clockwise to unlock.

After a small amount of travel, the internal lifting mechanism will assist you in opening SmartSurface and will keep it open while you work.

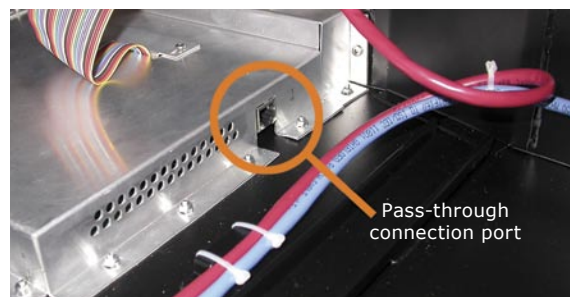


Figure D-3: Locating the internal connection point

In the bottom of the SmartSurface frame, locate the Master Control module (the large, aluminum-shielded rectangle in the middle of the frame). On the right-hand side, near the rear, is an unmarked RJ-45 port (shown

in Figure D-3). This port is a “pass-through” connector, which allows connection of your Console Director to your phone system using the RJ port on SmartSurface’s connection panel marked “Phone System.”

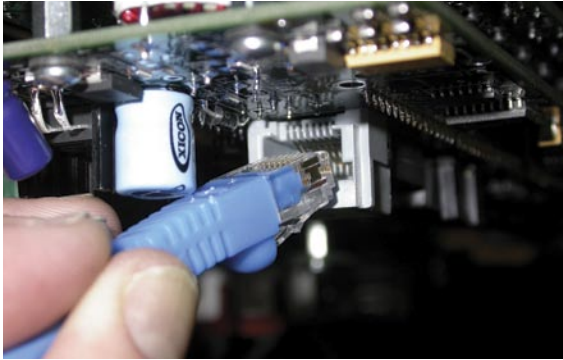


Figure D-4: Console Director RJ-45 port

Connect one end of the Ethernet cable supplied with your Console Director to this port, and the other end to the RJ-45 port on the Director itself (you’ll find this port on the front edge of the Director, facing you when the enclosure is in the “open” position - see Figure D-4).

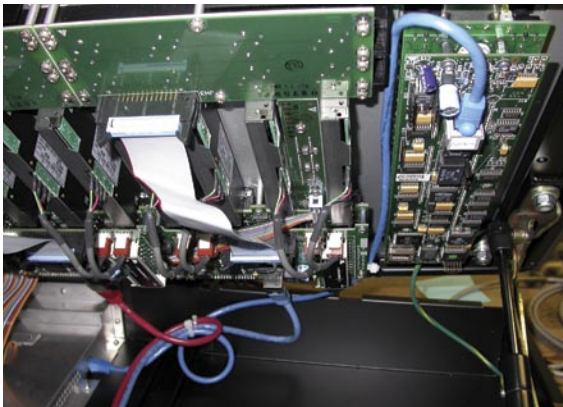


Figure D-5: Completed Console Director cable installation

Once the cable is connected to both RJ ports, position it next to the other cable bundles adjacent to SmartSurface’s fader assemblies, and use small tie-strips to attach it to one of these bundles. Do not leave the cable loose inside the enclosure, as it may interfere or become entangled in the moving parts within. A small loop next to the Master Control module is acceptable if there is excess cable length.

Close the SmartSurface by pulling gently downward on the front corners of the surface; take care that the lifting assist mechanism does not “snap” the surface shut. If

you wish, you can use a #00 Philips-head screwdriver to engage the closure locks located in the forward corners of the SmartSurface bezel to lock the surface closed.

The last step in this procedure is to connect your SmartSurface to your phone system.



Figure D-6: “Phone System” pass-through port

Route an appropriate length of CAT.5 Ethernet cable from your phone system to your SmartSurface. Connect it to the port marked “Phone System” on the SmartSurface connection panel, as shown in Figure D-6. Connect the other end to the appropriate port on your phone system (see the User’s Manual specific to your Telos phone system for details).

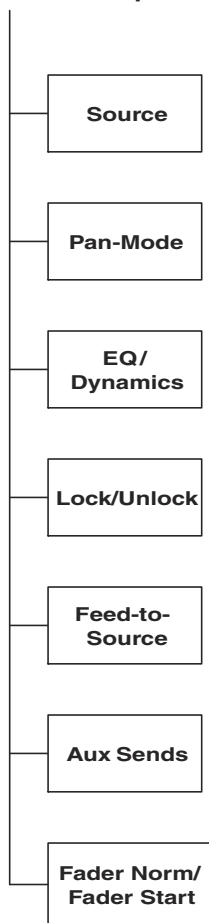
The installation is now complete. You may begin using your Console Director immediately - no restart of SmartSurface is necessary.

Appendix E: Menu & Screen Reference

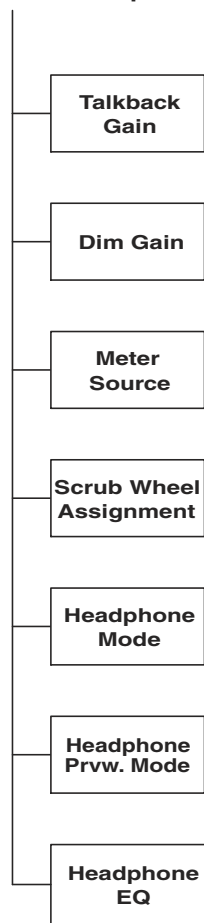
The charts in this appendix are provided as a “speed reference” to be used in locating commands within the SmartSurface command structure.

SmartSurface Operator’s Option Menus

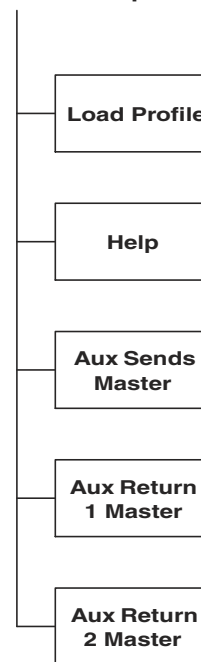
Channel Options Menu



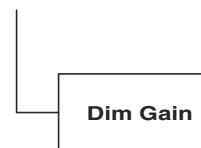
Monitor Options Menu



Control Options Menu

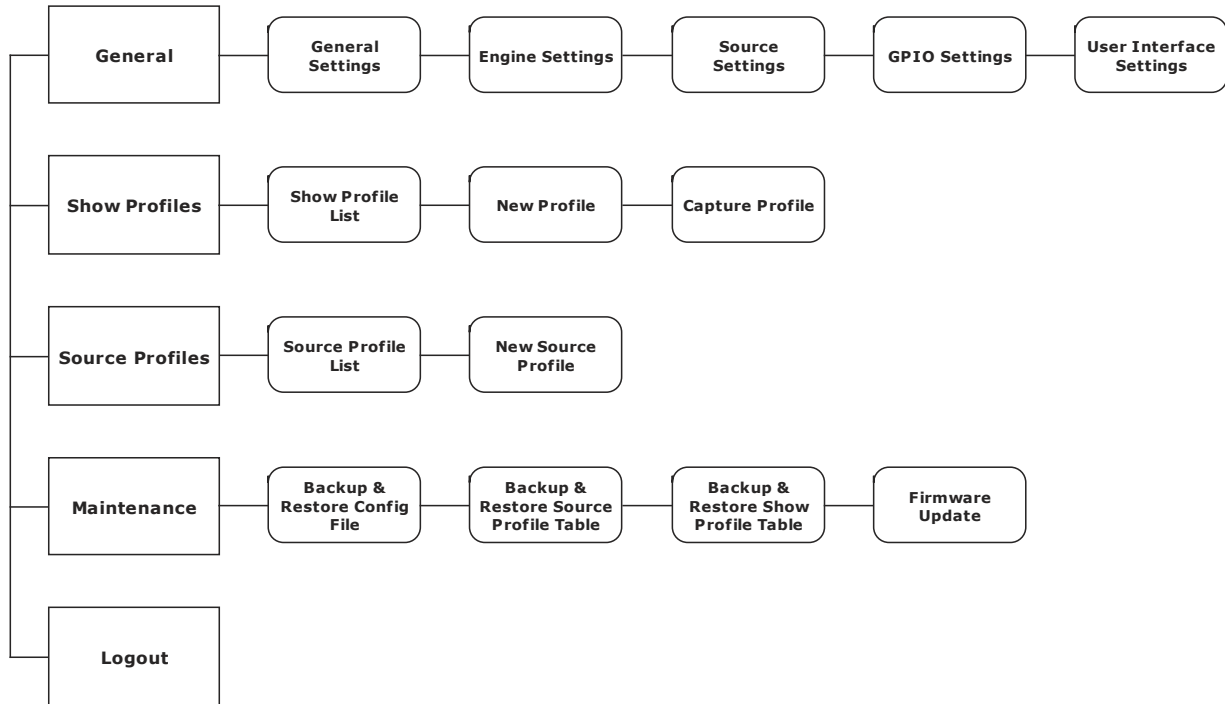


Studio Options Menu



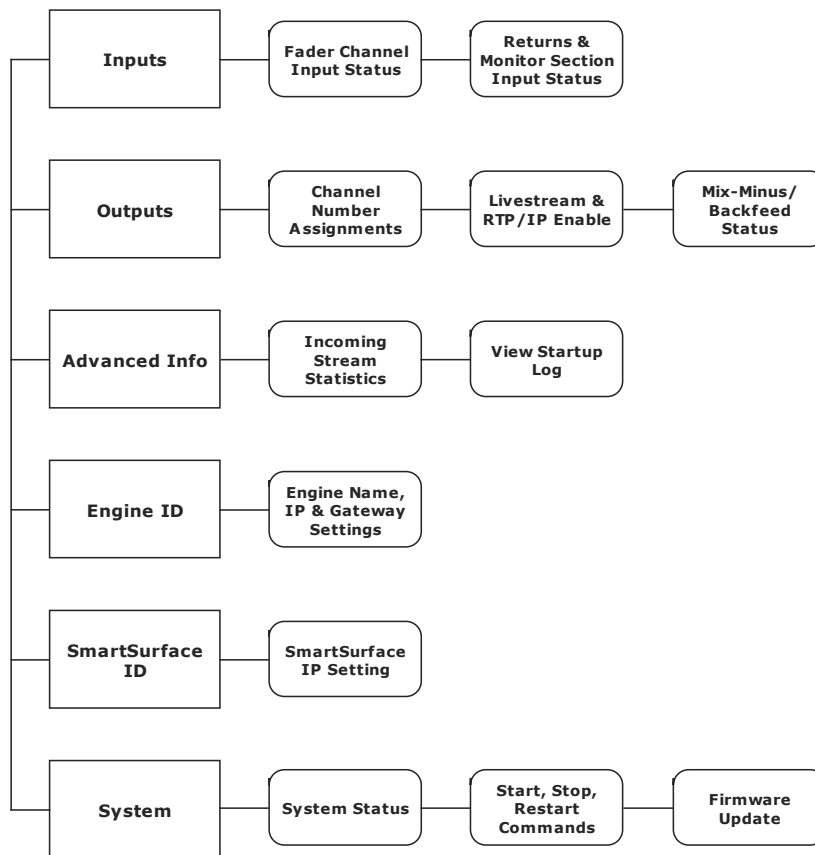
Web-based Option Menus

SmartSurface HTTP Configurator

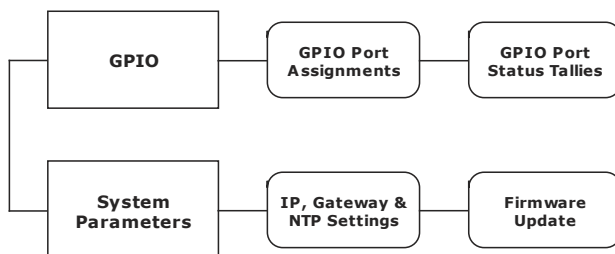


Web-based Option Menus

SmartEngine HTTP Configurator



GPIO HTTP Configurator



Meeting coffee, black.

Less Powerpoint, I beg you.

Must keep eyes open.

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, SmartSurfaces, 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 can be 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*.

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.

Fader moves roughly

Must disassemble and clean.

No Cokes at the board!



Axia Audio, a Telos Company • 2101 Superior Ave. • Cleveland, Ohio, 44114, USA • +1.216.241.7225 • www.AxiaAudio.com