

SoftSurface

Installation & User's Guide



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SoftSurface Version 1.04

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Feedback

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

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This product is covered by a 90-day limited warranty, the full text of which is included in Appendix E of this manual.

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If you call Axia Support with a software question, we strongly recommend being near your computer when you call, so our Support Engineers can verify information about your configuration and the conditions under which the problem occurs. Please have your software license key handy.

Feedback is welcome

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

About This Manual

This manual covers setup and use of Axia SoftSurface software. It is assumed in this document that you are familiar with Livewire's basic concepts, as outlined in the companion *Introduction to Livewire: Systems 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.

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

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A Note From The Vice President of Axia

2013 marks a banner year for Axia — this year marks the first decade of the Connected Studio.

In retrospect, it's hard to believe it's been 10 years since a start-up division of Telos grabbed everyone's attention with a radical idea about building radio studios using Ethernet. At a time when "state of the art" meant discrete digital audio, and a routing switcher was something only the richest stations could afford, Axia shocked everyone with the idea that studio peripherals from the console to the audio processor could be networked, and that everyone could have the benefits of a routing switcher — for about a third of the cost of traditional technology.

When we launched Axia, Telos founder Steve Church made a prediction. "Three things will happen," Steve told us. "First, people will say that 'it will never work.' When they're proven wrong, they'll say 'It works, but you don't need that.' And finally, as they see Axia becoming successful, they'll say 'IP-Audio? We do that too!'"

Steve, as always, was right on the money. In 10 years, Axia Livewire has become the world's most popular IP-Audio networking technology, and our mixing consoles are the world's best-selling — they're on-air in over 4,000 studios, and counting. You'll find Axia equipment everywhere: at privately-owned stations, and large clusters run by big conglomerates. At established public broadcasters, and at newly-licensed International stations. At government broadcast facilities, and podcast studios. Needless to say, we're humbled by the trust broadcasters place in us — and by your enthusiasm!

Being first with new technology is nice, but we've never been content to rest on our laurels. So the Telos R&D team continues to innovate and expand the scope

of IP-Audio. Axia consoles are the first and only IP-Audio consoles with a pre-configured network switch built in, to save broadcasters the effort and expense of procuring and programming third-party switches. Our compact xNode AoIP interfaces feature one-button setup, and can run on mains power or Power over Ethernet (PoE) for flexibility and redundancy. We've even developed our own zero-configuration Ethernet switch for Livewire — xSwitch — to make deploying IP-Audio easier than ever.

More than 45 Livewire partners believe in the vision of the Connected Studio too. Collectively, they make dozens of hardware and software products (ranging from telephone systems to audio processors to program automation) that interoperate directly with Axia networks via Ethernet — making the connections "smart", and eliminating the need to purchase audio conversion devices. We've also partnered with the high-performance audio standard RAVENNA to expand broadcasters' networking options even further. And there are a lot more great new ideas in the cooker — so watch this space!

In the final analysis, Axia's success isn't due to our efforts. It's you, the broadcast professional, who has embraced our vision, used our technology in ways we never dreamed of, and raved about Axia products to everyone who'd listen. We owe you a huge debt of thanks. And we promise to never stop trying to amaze and delight you.

So here's to the next 10 years.
Onward and upward!

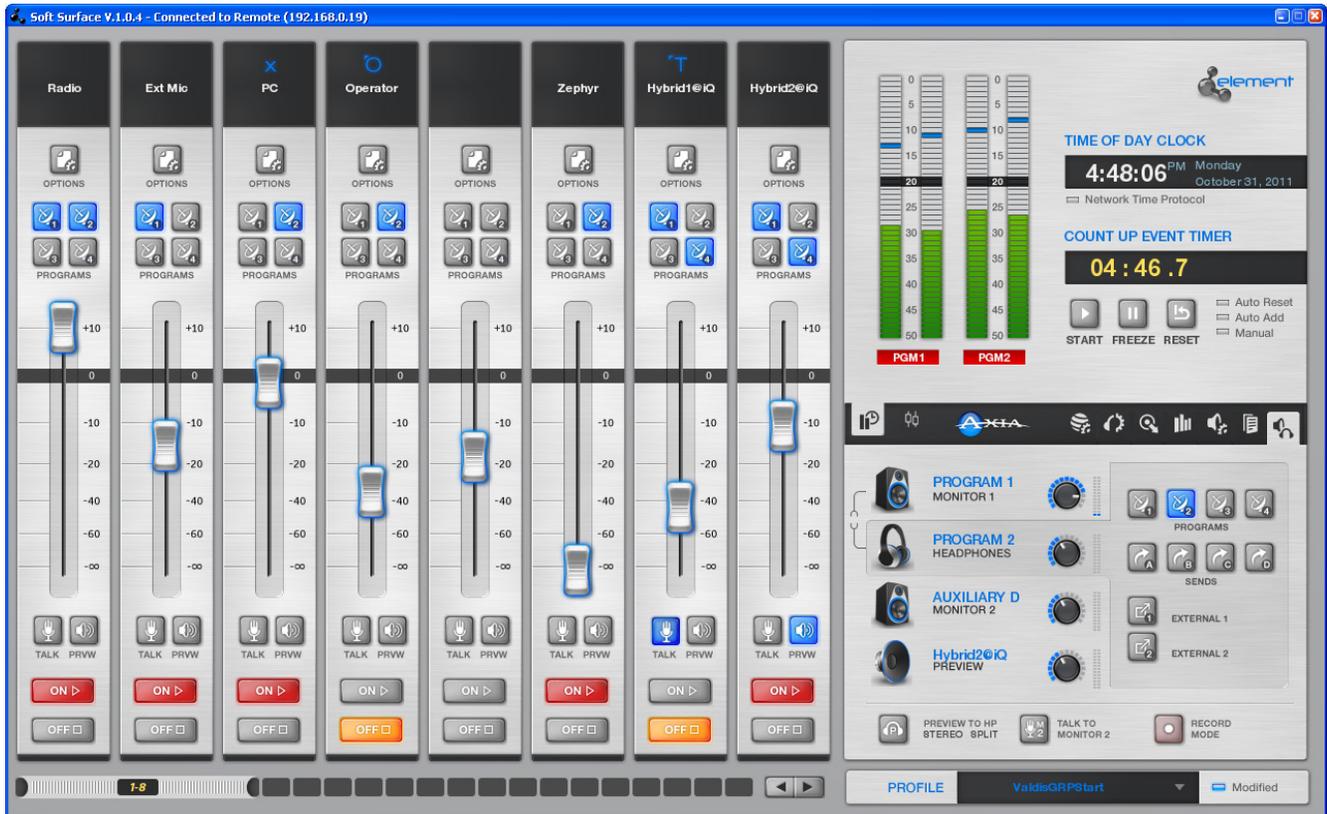
Marty Sacks
Vice President, Axia Audio



*CAT.6, small and light
yet sound and data pour forth.
What magic is this?.*

Chapter One:

What is SoftSurface?



Introduction

Axia SoftSurface is a Windows application that can be used in either of two ways:

- 1) As a remote control for an Axia Element control surface, or
- 2) As a stand-alone mixing surface controlling an Axia StudioEngine.

SoftSurface software requires a monitor with a minimum resolution of 1280x1024. It can function on Windows XP and Windows 7, 32-bit or 64-bit versions. SoftSurface provides channel controls similar to an Axia Element surface.

SoftSurface Interface

The application can be divided into two areas: the channel section and the control section. The channel section's onscreen width can vary based on the amount of channels you wish to display; the minimum fader count is 4. The control section is fixed in size and what is displayed varies based on which tabs are selected.

Channel Section



Figure 1-1: Onscreen faders

The number of faders displayed is defined by the “Faders visible” option in the control section. If the number of visible faders is less than the active faders, the scroll bar and Left/Right arrow buttons under the channel section allows the user to access those other faders by scrolling left or right. The scroll bar has numbers on it to indicate which ranges of faders are displayed.

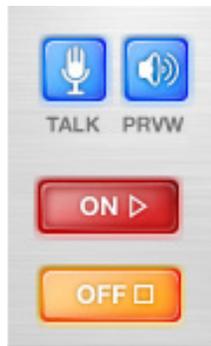


Figure 1-2: Channel controls

The OFF button indicates an OFF state when “illuminated” orange. Pressing the button will engage the OFF state.

The ON button indicates an ON state when “illuminated” red. Pressing the button will engage the ON state.

If the source loaded to the channel has Fader Start enabled, the ON and OFF states are controlled by the fader position. The ON button will only engage the ON state if the fader is above the bottom position.

The TALK button will “illuminate” blue when pressed and the appropriate source is loaded to the channel. The TALK function is an IFB (interrupted feedback), also known as Talkback, and opens an audio channel from the operator microphone to the dedicated return audio for the source.

The Preview (PRVW) button illuminates blue when the channel is in the preview mix (also known as a cue mix). Pressing the button assigns the channel to the mix.

NOTE: mouse left click is a momentary press of the button when the pointer is over the button. A mouse right click is a press and hold of the button which the pointer was resting above.

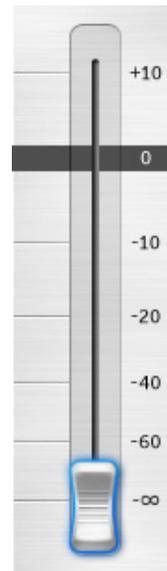


Figure 1-3: Fader control

The fader controls the audio gain of the source loaded to the channel. If the profile loaded to the channel has Fader Start enabled, the channel automatically turns ON when the fader is raised from its lowest position. Once the fader is in the ON state, the ON and OFF buttons may be used to control it. Fader in the lowest position is always OFF state for the channel.

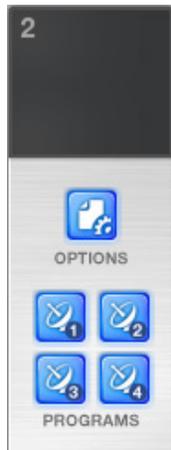


Figure 1-4: Bus assignment buttons

Above the fader are four mix channel buttons and an option button. The mix channel buttons represent Program buses 1, 2, 3 and 4 and are represented with the same number on the button. For example, will be “illuminated” blue when the channel is feeding the Program 1 mix.

The OPTION button opens other functions that can be applied to the channel. These functions are found in the Control section of the application. The Source Select function is one option that is available; it allows the user to load an audio source to the channel. When a source is loaded, the name of that source will appear above the channel. The top area of the channel shows the channel number in the top left, the source name, and may also show a status indication above or below the name based on some state operation.

Examples of these additional displays:

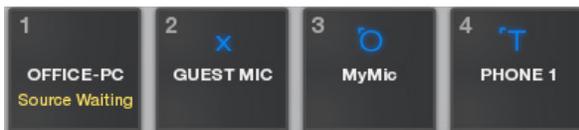


Figure 1-5: Channel status displays

“Source Waiting” indicates that a source is waiting for Channel-1 to turn OFF before loading.

The blue “X” on Channel-2 indicates that the source is loaded but in “Listen Only” mode. This means that some other control surface has ownership of the source (or that there is a possible error in source ownership advertisement information).

The blue “O” on Channel-3 indicates that this is the Operator mic channel, and that this channel is the source audio in Talkback being sent to another destination. The destination is indicated by a...

Blue “T” on Channel-4, which indicates that this channel is the receiving end of talkback audio.

Control section



Figure 1-6: Assigning a source

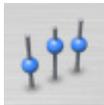
When the Option button is pressed on a channel, the top half of the control section will change to display the channel source select options. The 6 icons represent different options that are available for a channel.



Source select allows the user to load a source to a fader. First column shows the name of the source, 2nd column shows the type of source, the 3rd column shows the Livewire system name given to the source, and the 4th column shows ownership information of the source. Highlighting a source and pressing the select button will load the source to a channel.



Feed to Source is the unique mix-minus that has been generated for the source. This mix-minus (backfeed) can be changed from the SoftSurface control interface. This option is only valid for phone or codec sources.



Equalizer provides the ability to modify the frequency response. This is three bands with adjustable center frequencies.



Microphone processing provides the option to add/adjust a noise gate, a compressor, and a de-esser. This option is only valid for microphone, phone, or codec sources.



Pan option provides the ability to adjust the stereo presence of a source between the Left and Right channels. Also provided is a mode control and inverse polarity. The mode function allows the user to select the stereo channels from the source or one channel and create a dual mono source.



Aux send option allows sending a channel's output to any of 4 auxiliary mix buses, beyond the 4 program mixes. The control provides gain adjustment and pre- or post-channel fader configuration.

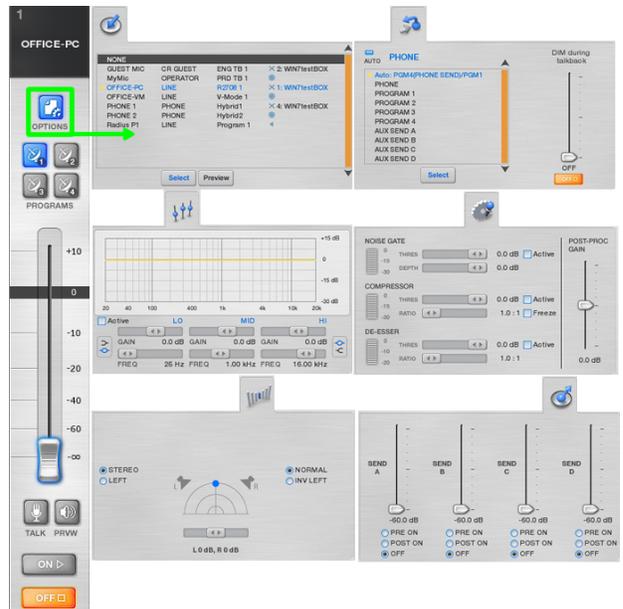


Figure 1-7: Options screens



Those are the options available through the Channel Options  button. While the Option function is enabled, there are confidence meters present in the top right corner which show presence of audio levels from the source (and returning to the source, if the appropriate source type is selected).

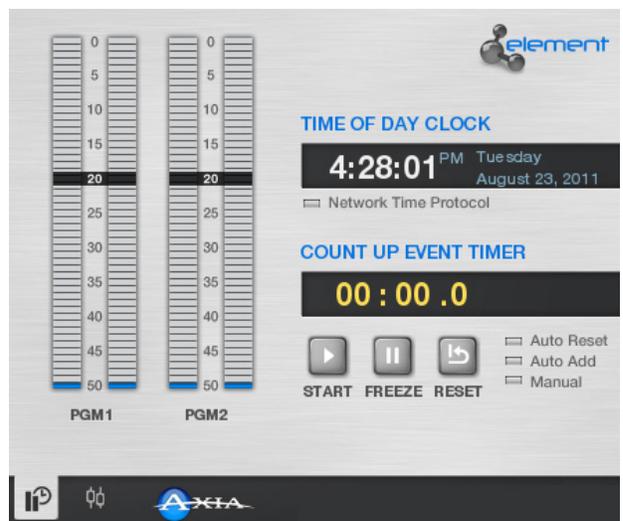


Figure 1-8: Meters, clock and timer section

When none of the channels have the Options button selected, the top half of the control section displays Program 1 and 2 meters, along with the time and date (provided by the host PC) and a count up timer.

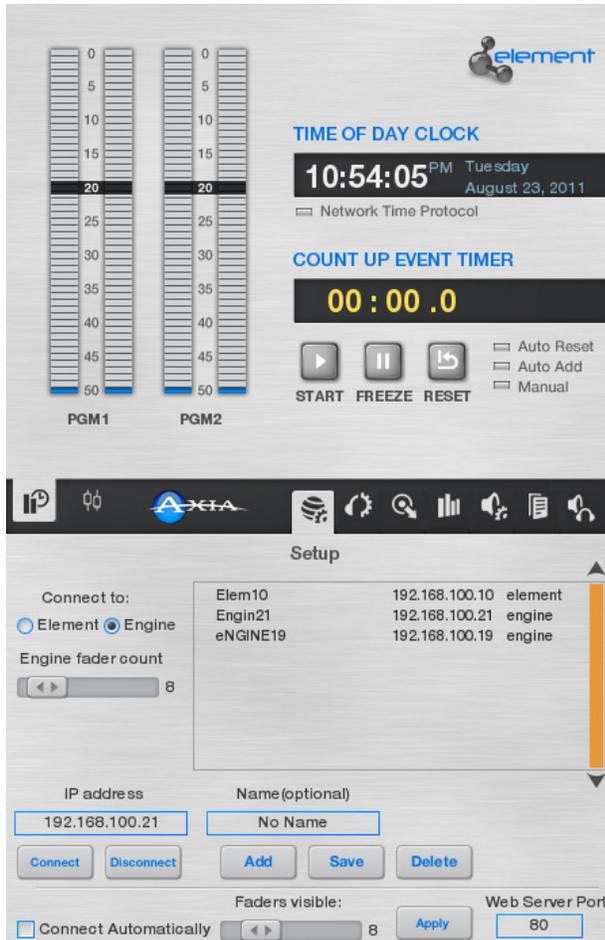


Figure 1-9: Connecting to Element or Engine

On startup, the control section will default to showing the Setup tab in the lower section, if no connections are defined or no auto-connect is configured. If the application is configured to connect automatically, then the default view will have the lower section set to the main monitor tab. (More about this in Chapter 2, which covers installation and setup of the SoftSurface application.)

The Setup tab is where connections are configured, the number of faders displayed on the screen, and an option to change the configuration port is presented, in cases where some other application installed on your PC also uses port 80.



The Main Monitor view provides control of the monitor speakers, operator headphones, external monitor speakers, and preview volume.



Show Profiles tab shows the profiles that have been configured and allow the user to select a show and load it.



Monitor options provide control over dimming values and control of the channel feeding the monitors and headphone.



Meter options provide control over the presentation of the meters.



Auxiliary Send and Return provides control of the final AUX send mix as well as two auxiliary returns as defined in the Show Profile. The returns can be assigned to a program bus. These options are defined in a Show Profile and controlled as needed by the user.



The Headphone (HP) processing tab shows the frequency curve of the processing and allows for control of the processing.



At the bottom of the control section is a status lamp and two functional buttons. The status lamp provides indication on the preview to operator headphone option. There are three options: OFF, SPLIT, and STEREO.

There is a Talk to Monitor 2 button which, when engaged, routes the operator microphone to the Monitor 2 source allowing communication with anyone monitoring that source.

A Show Profile may have a Record Mode option enabled. If enabled, this button is used to enter and exit the record mode.

Below this is the Show Profile status information, which indicates the name of the Show loaded to the surface.

Chapter Two:

Installation and Setup

SoftSurface can be installed on Windows machines with a minimum monitor resolution of 1280x1024. The computer needs a static IP address within the network scheme of the Axia LAN. If you are not clear on the IP address to use, please contact your network administrator. If you are the administrator, please call Axia support if you need help resolving this issue. The computer also needs a network connection to the Axia LAN.

If you will be using SoftSurface to remote-control an Axia Element, the PowerStation paired with that Element needs to be version 1.1.3 or higher; if your Element is paired with a StudioEngine and Element Power Supply /GPIO, the PS/GPIO needs to be version 2.5-series or higher. If you are using SoftSurface in stand-alone mode, you will need an Axia StudioEngine, version 2.5.4e or higher.

System Requirements

Windows XP or later
Video Resolution 1280x1024
Static IP
PowerStation 1.1.3 (or higher)
Element PS/GPIO 2.5 (or higher)
StudioEngine 2.5.4e (or higher)

Installation

Navigate to and double click on the SoftSurface Windows Installer Package and follow any on-screen instructions. A status bar will indicate the process completion. If the process appears to be stuck three quarters of the way, check to see if there is a Register window hidden from view.

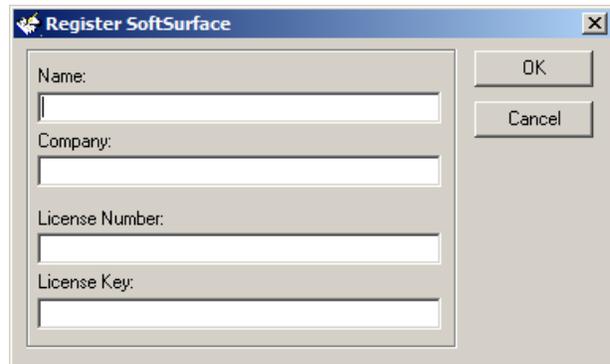


Figure 2-1: Registration screen

Enter the values requested. Once completed, the status bar's progress will continue and installation of SoftSurface is finished. If you're using the SoftSurface as a remote control for an Element, make sure the Element is already configured and proceed to SoftSurface Setup (next page) for setup of the SoftSurface. If you plan to use SoftSurface in stand-alone mode, you need to configure the Axia StudioEngine.

StudioEngine setup

Connect power and a network connection to the StudioEngine. The network connection needs to come from an approved network managed switch (see www.AxiaAudio.com/switches/) with the appropriate configuration. If assistance is needed with the configuration of the network switch, please contact Axia support.

Also required is an Axia Audio Node which provides audio input and output and network sync source. If you're not familiar with Axia audio nodes, please refer to the documentation that comes with the product.

After power has been applied, the StudioEngine will boot; once completed you will use the front display and rotary knob to give the device an IP address. This will prompt the device to reboot when done.

- Press the knob (it acts as a button too)
- Rotate the knob until "4. Engine IP settings" is highlighted
- Press the knob to select

- Rotate the knob to highlight the “Net Addr” field
- Press the knob to select
- Rotate the knob to move the cursor. Press the knob to highlight the character under the cursor. While highlighted, you can rotate the knob to edit the value. Press the knob when the desired value is shown. Continue this process until the desired IP value is shown. When done, select the check mark to exit editing.
- Adjust the net mask if required for your network. 255.255.255.0 is likely a suitable netmask value in most installations.
- Adjust the gateway if required for your network. You may leave this blank.
- When the IP values have been entered, select the [OK]

If a change was made, the system should ask for a reboot, confirm for reboot. If not, you may select “6. System” to select a restart option.

Now, open a web browser with access to the Axia LAN and type the IP address of your StudioEngine.

The first order of business is to make sure the Program and Monitor channels have unique numbers in your network. These numbers equate to audio streams within your network. If another device is generating a network stream with the same number, this will cause problems with your audio.

Select the **Program and Monitor** link. Authentication is requested; the default **username** is *user* with **no password**.

Review the channel numbers and change as needed. Note that all StudioEngines come with the same default numbers. For this reason it is suggested that you change these numbers if you plan to add additional StudioEngines to the network.

SoftSurface Setup

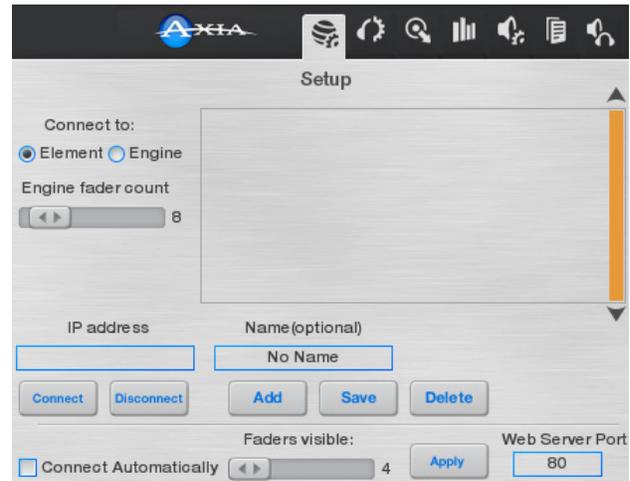


Figure 2-2: Specifying a connection

SoftSurface offers the ability to connect to an Element for remote control, or to assume dedicated control of a standalone StudioEngine. If connecting to a StudioEngine, you will need to define the number of faders to be enabled with the “Engine fader count” setting.

- Select the device you are connecting to. Select “Element” if connecting to a PowerStation.
- Move the computer mouse over the “IP Address” text box to type the IP address of the device you will control.
- Move the mouse over the “Name” text box to give a logical name to the connection.
- Press the **Add** button to add the settings to your connection list.
- Use the **Connect** button to establish a connection to the desired device.

You can also use the **Save** button to save changes made to a previous item, or **Delete** to remove a configuration from the list.

The **Faders Visible** control is used to define the width of the SoftSurface displayed onscreen. After setting a value and pressing the **Apply** button, you need to close and restart the application.

Use the **Connect Automatically** option to connect to a device when the application starts without user input.

A web server port setting is available if you need to change the port to access the application configuration HTML interface. In most cases port 80 is suitable. If there is another application using port 80, this option allows you to change the SoftSurface port.

When using SoftSurface to remote-control single or multiple installed Element control surfaces, define and save the settings in the setup section for quick access to those devices. Select the radio button for Element under “Connect To” and add the IP address of your Element-PS/GPIO device(s) and/or PowerStation(s) into the connection list. Once added, select the device in the connection list and press **Connect**. More information on remote control is found in Chapter 3.

Using SoftSurface in stand-alone mode is an easy way to save on countertop space in a small production suite or voice over booth. Select the Engine radio button under “Connect To”, select the amount of Engine faders, and type in the IP address of a dedicated StudioEngine. Press the **Add** button and consider selecting the Automatic Connection option to immediately setup the connection when the application starts. Chapter 3 covers topics for stand-alone mode.

*Do digital streams
dream of electric DJs?
Just Bradbury knows.*

Chapter Three:

Using SoftSurface

Using SoftSurface for Element remote control

SoftSurface can act as a remote control to an existing Element surface. This requires either PowerStation version 1.1.3 (or higher) or Element-PS/GPIO version 2.5.0 (or higher) with a StudioEngine.

In the setup tab of the control section of SoftSurface,

- Select the “Element” option under “Connect To:”
- Enter the IP address of either the PowerStation or Element-PS/GPIO
- Press the Connect button

You may also consider entering in a name and pressing the Add button to enter the connection into your connection list which appears in the Setup section.

SoftSurface provides access to the channels in the Element surface as well the monitoring section. The functions you would have at the channel are available remotely on the SoftSurface. The user of the SoftSurface may turn channels ON and OFF, adjust the gain of a channel, change the assignment to Program mixes, as well as adjust the channels per the Options function. However, some Element options are not currently represented in SoftSurface; modules not available for control via SoftSurface are the Call Control Module, Intercom module, or SmartSwitch accessory modules.

The Element control surface will change as you make changes in SoftSurface. As a channel is turned on from the SoftSurface, Element’s ON lamp for the same channel will light up.

If a fader is moved with SoftSurface, the corresponding Element motorized faders will move. If motorized faders are not installed, the Element control surface

will provide an indication in the Status area above the fader, showing that the physical fader position does not match the channel’s actual output value. The channel number will be replaced with either a downward or upward arrow, which indicates the direction the physical fader must move in order to reach actual output value. If someone were to move the physical fader at this time, no volume change would occur until fader position matches the value established by SoftSurface. Once the logical and physical fader values match, the channel number reappears and control is given back to the physical fader.

The SoftSurface channel Options button will provide control of the various options available to the user of the SoftSurface, but will not change the video output of the remote controlled Element as if the option button was pressed. The Options buttons on the control surface and the SoftSurface are independent of each other.

TIP: In the unlikely event of hardware failure within an Element console, you can use SoftSurface in Stand-alone remote control mode for the duration of time it may take to resolve the hardware issue. Refer to the next chapter on using the SoftSurface in stand-alone mode.

Using SoftSurface with StudioEngine

SoftSurface can act as a control surface for an Axia StudioEngine, even if there is no hardware control surface present. This requires a StudioEngine, version 2.5.4e (or higher) which is not associated with any other Element.

In the setup tab of the control section of SoftSurface,

- Select the Engine option under “Connect To:”
- Select a value under the Engine fader count (the amount of channels you wish this “soft” control surface to have)
- Enter the IP address of the StudioEngine
- Press the **Connect** button

If SoftSurface will *always* be connecting to this Engine for use in Stand-alone mode, consider selecting the “Connect Automatically” option so that when the application starts, it connects without user input.

Adjust the “Faders visible” option to configure the desired width of the application on the computer desktop. If the visible value is less than the configured Engine fader number, you will use the lower scroll bar and arrows under the channel section to move the view of the faders.

In Stand-alone mode, you will need to configure Source Profiles and Show Profiles for the user of the surface. To do this you will need to open a browser and type in the IP address of the PC that SoftSurface is installed on, and then proceed to Source Profile configuration. (From the same PC you may also type 127.0.0.1 in the browser, which is a loopback address for the PC.) When selecting a link for the first time, authentication may be required. The **default username** and **password** is the same as other Axia devices: Username: *user*, Password: (no password).

Source Profile Configuration

First connecting to the configuration user interface of the SoftSurface, you will see various links to the left hand side and configuration view in the center.

First step is to select the **Configuration** link; this opens a view of the Show Profile list and provides access to the Source Profiles configuration.

Next, select the **Source Profile** link. At first there will be no sources shown in the center, only the **Add New Source** link. You will need to define the sources that will be used on the SoftSurface. These definitions will determine how the sources are treated and how the studio will perform when the sources are used.

Select the **Add New Source** link. If you are familiar with Element Source Profile configuration, you should be able to get going on your own. If you have some questions on some of the details of Source profiles, refer to Appendix A – Source Profile Details; this will lead you through some examples to get you started. After that you should be able to build the remainder on your own and refer to Appendix-A for additional options on the profiles.

CD player (Line type)

The following is an example of how to configure a CD player Source Profile. It assumes that a CD player has been connected to an Axia Audio Node (either Analog or AES/EBU) using the instructions found in the Node’s product manual.

The “Primary Source” option is where you define the Livewire Channel that is unique to the source. Select the **Browse** button next to the text field. A pop up will list sources advertised on the network. Find the source for your CD audio and select the **Channel Number** link. If you do not see the desired source, it’s probably not set up correctly; refer to the Node manual for correct setup procedures for node sources.

Enter in a user friendly name in 10 characters or less in the Source profile name field, i.e.: “CD 1”.

By default all the “Source availability” check boxes are checked. If you desire that this source not be available on a specific channel, deselect the channel’s corresponding box.

By default the “Source type” is set to Line which is the appropriate option for this type of source. Other types are explained in Appendix-A.

The remainder of the options are also appropriate in their default setting, so the next step is to press the **Save Changes** button at the bottom of the screen.

You have now created a Source Profile which can be loaded to a channel as described in section 1-2-2. Press the **Options** button of a channel, highlight the source shown in the Control section, and press the **Select** button to do so.

Your Microphone (Operator Mic)

The following is an example of how to configure a Source Profile for the user’s microphone. It assumes that a microphone has been connected to an Axia audio node with a microphone input or the output of a microphone preamp is connected to an Axia audio node that accepts the output signal from the preamp. Begin first by clicking **Add a new source**.

Define the Livewire Channel for your microphone input as the Primary Source. Select the **Browse** button next to the text field. In the pop up, select the source.

Enter in the user friendly name in 10 characters or less in the Source profile name field – let’s name it “MY MIC”.

Under “Source Type,” you must select the **Operator** option to identify the function of this Source Profile. “Operator” designation is always used for console operator’s mic position, and helps to identify source audio for talkbacks and determine which monitors to mute.

The remainder of the options will be acceptable at their default value. One option to consider is the Signal Mode setting, halfway down the page. If the mono microphone source was connected to a stereo input, the signal mode can be set to Left to create a dual mono source based on the Left channel of that Livewire source. If a microphone input was used, this source is already set as dual mono and the default stereo setting will be suitable. Save changes to end.

Guest Microphone (Guest Mic)

The following is an example of how to configure a Source Profile for a guest microphone next to the control operator. This may be considered the second microphone in the room. It assumes that a microphone has been connected to an Axia Audio Node with a microphone input or the output of a microphone preamp is connected to an Audio Node that accepts the output signal from the preamp. From the Source Profile page, select the **Create Source Profile** link to begin.

In the **Primary Source** field, you can enter in the Livewire channel number in the text box (if you know it) or you can select the **Browse** button next to the text field and search for the desired source. Within the pop up, select the source and the channel number will load into the text field.

Enter a user friendly name in 10 characters or less in the **Source Profile Name** field; something like “Guest MIC”.

Under “Source Type,” select the **CR guest** option. This designation is for a microphone in the Control Room (the studio space where the control surface is located – SoftSurface PC) and will mute the CR monitors when the microphone is turned on.

The remainder of the options will be acceptable at their default value. One option to consider is the signal mode setting. If the mono microphone source was connected to left side of a stereo input, the signal mode should be set to Left to create a dual mono source based on the Left channel.

Saving the changes will complete the profile creation.

Caller (Phone)

The following is an example of how to configure a Source Profile for a phone hybrid. It assumes that a single hybrid, similar to a Telos Hx1, has been connected to an Axia Audio Node, and that GPIO logic has been wired between the hybrid GPIO port and an Axia GPIO Node per the manuals of these products.

The Livewire channel number which represents the source should be selected in the Primary source field, by manually entering in the known number or searching for the source with the use of the browse button.

In 10 characters or less, the Source profile name should indicate the audio source. We’ll use “PHONE”.

Under “Source Type,” select the **Phone** option. The Phone source type indicates that a mix-minus will be produced for this source.

“Hybrid Answer Mode” set to **Channel ON answers hybrid** will trigger GPIO ports so that when the channel assigned to PHONE source is turned ON, the hybrid will be activated, seizing the line.

“Feed to source mode” is the option that selects the audio that will be sent to the caller. If the **Program 1** option is selected, the caller will hear Program 1 audio.

The **PHONE** option is a special audio mix that is similar to Program 4, with the exception that the mix is pre-fader and pre-ON/OFF. In this mode, any channel with Program 4 selected will be heard by the caller, regardless if the channel is OFF or the fader is DOWN. The default option is the **Auto** setting, which toggles the mix-minus audio based on the state of the channel the hybrid is loaded to. If in the OFF state, the Feed to Source is the PHONE mix. If the channel state is ON, the Feed to Source is the Program 1 mix. Select the mode that will fulfill your operational requirements; if you're not sure which one is right, keep the default setting — but be aware that the mix-minus audio can change based on whether the caller fader is in the ON state or the OFF state!

“Signal Mode” should be set to **Left** assuming the hybrid audio has been wired into the Left channel of a stereo input.

The remainder of the options will be acceptable at their default value. Press the Save changes button to complete the profile.

The function of this source uses GPIO to signal to the hybrid to seize the line. SoftSurface is responsible for signaling this ON and OFF state to the network. In order for the GPIO node that is wired to the GPIO port of the digital hybrid to activate the hybrid, the port needs to be associated with the hybrid source. To do this, using the GPIO Node interface, simply select the Livewire channel number that corresponds to the hybrid audio source. This associates the GPIO port to the logic as defined by the Phone source type just created. Since the Livewire channel number is tied to the Source Profile, so is the number tied to the logic of a Phone type. Refer to the GPIO tables in Appendix-B to see the logic associated with each source type.

Continue adding the remainder of your Source Profiles and refer to Appendix-A for details on the other options that are available in setting up a profile. Appendix-B shows the GPIO logic that is associated with each source type.

Show Profiles

Show profiles are control surface layouts. A Show profile defines sources loaded to channels and the monitor settings. Once you have Source Profiles configured, you can build Show Profiles to quickly load all the channels at once.

In the configuration screen above, the only Show is the Default Show profile, which is a permanent profile. You can edit the Default show, create a new show, or capture the current settings and make a show.

To make a basic show:

- Select the **New Show Profile** link
- Select the first channel
- Select a Source Profile in the Source ID field (if there are no sources, than follow the instructions in section 4-1)
- Locate the “Assign to PGM1” option and select ON
- Select the **Save Changes** button at the top or bottom of the page
- Repeat steps c, d, and e for each channel
- Select the **Monitor section** link
- Locate the “Control Room Monitor Options” heading and select **PGM1** for the source option.
- Locate the “Control Room Headphones Options” heading and select **PGM1** for the source option.
- Locate the “Studio Monitor Options” heading and select **PGM1** for the source option.
- Select **Save Changes**
- Give the Show a name and press **Save Changes**.

This is a basic show. You can select the Show Profile tab in SoftSurface, introduced in Chapter 1, and find the newly created show. Select the show and observe the show load to the surface.

You probably noticed during configuration that there are a lot of other options for Show Profiles. We'll look at some of these now.

Each channel link is primarily used to define which Source Profile will be loaded to the channel. Most of the options available in the Channel Edit screen override the settings that have been defined in the Source Profile. For

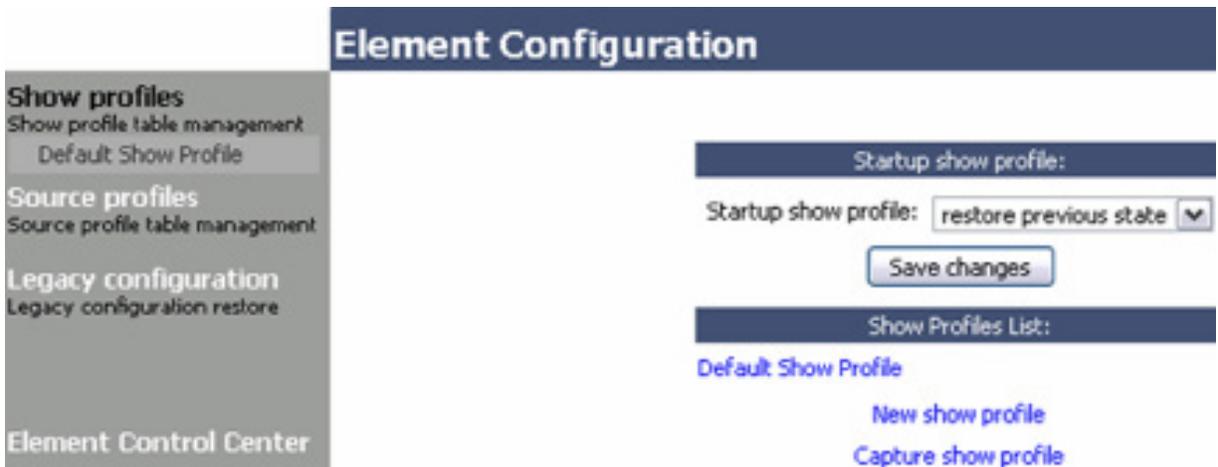


Figure 3-1: Configuration options

example, the first option is the “Feed to Source” option, which you can modify or choose “Retain source setting”, which preserves the settings in the Source Profile.

The most important option to define in the Channel page is the “Assign” option — “Assign to PGM1” for example. In most channels you will want to assign PGM1 to On.

The **Monitor Section** link provides important settings that define which source is monitored in the control room monitors, control room headphones, and studio monitor. Depending on the usage of the SoftSurface, monitoring Program 1 or an External input would be appropriate.

Setting up ONAIR light logic with a Show profile

In order to get an ON AIR light trigger out of a GPIO node, you must define a number in the “GPIO channel for CR monitor” option. This number is also assigned to a GPIO port. Doing so will tie the GPIO port to the CR monitor logic as defined in Appendix-B.

From the configuration screen, select a Show Profile and:

- Select the Monitor Section link
- Locate the “Control Room Monitor Options” heading
- Locate the “GPIO channel for CR monitor” option
- In the text box, enter a number that doesn’t conflict

- with any numbers that your sources use
- Save changes
- Load the Show
- When configuring the GPO port on a GPIO node that you’ll use to trigger the ON AIR light relay, enter the same number that you entered in the “GPIO channel for CR monitor” box in the step above. For example, if you used channel 2011 in the text box, enter the same value in the channel field for the GPIO port.
- Save and test your new ON AIR trigger using the Operator’s mic.

More information on Show Profile options are available in Appendix-C

StudioEngine Setup

The StudioEngine which is required to operate the SoftSurface in stand-alone mode is a multi functional device. This section will walk you through configuring SoftSurface for use as a software-only controller for your StudioEngine.

Configuring an IP address

Once the Engine has completed its boot cycle, the flashing screen on the front display will be replaced with system operating information. Follow these steps to configure the Engine with an IP address:

- Press the knob (it acts as a button too)
- Rotate the knob until “4. Engine IP settings” is highlighted
- Press the knob to select
- Rotate the knob to highlight the “Net Addr” field
- Press the knob to select
- Rotate the knob to move the cursor. Press the knob to highlight the character under the cursor. While highlighted, you can rotate the knob to edit the value. Press the knob when the desired value is shown. Continue this process until the desired IP value is shown.
- When done, select the check mark to exit editing.
- Adjust the net mask if required for your network.
- Adjust the gateway if required for your network.
- When the IP values have been entered, select the [OK]

If a change was made, the system should ask for a reboot, confirm for reboot. If not, you may select “6. System” to select a restart option.

HTML user interface – Program and Monitor Sources

Once the StudioEngine has an IP address, you can access the HTML user interface from the PC. When you type the IP address in a browser, you are presented with the “home” page which provides the needed links to different functions of the Engine. After pressing a link for the first time, the computer will request authentication. The **default username** and **password** is the same as other Axia devices:

Username: *user*

Password: (no password)

Setting up the StudioEngine requires defining the Livewire channel numbers that the Engine will produce. To do this, select the **Program and monitor outputs** (Prog and mon out) link.

This page shows all the various audio mixes that are produced by the Studio Engine. These all are sources in the Livewire network which you can route to an Axia audio node or other device that requires the audio from the studio. Mixes that are commonly used are Program 1, CR Monitor, CR Headphone, and Preview. Each mix should be given a unique Livewire Channel number. The Mode of each source that you plan to use should be set to Live Stereo. Any sources you don’t need can be disabled to help reduce clutter in source listings if you desire.

Once you have given each source a unique channel number, press the **Apply** button to save the change.

Note that each Engine, out of the box, has identical channel numbers — so we highly recommend changing these numbers to a unique series. If you do not do this, you risk introducing a new Engine to your network and having source conflicts.

This completes the setup of the StudioEngine. Now that you have defined channel numbers for your monitor sources and Program mixes, you can log into an Axia Audio Node’s Destination section and assign these sources to the Node’s outputs, giving your ears access to the audio from the control surface.

The other links available in the StudioEngine are covered in the troubleshooting section or in Appendix-D – StudioEngine.

Main, Auxiliary and Monitor Outputs				
	Channel (1..32767):	Mode:	Status:	Audio:
Program 1	<input type="text" value="2001"/>	Live Stereo <input type="button" value="v"/>	<input type="button" value="OK"/>	<input type="button" value=""/>
Program 2	<input type="text" value="2002"/>	Live Stereo <input type="button" value="v"/>	<input type="button" value="OK"/>	<input type="button" value=""/>
Program 3	<input type="text" value="2003"/>	Disabled <input type="button" value="v"/>	<input type="button" value=""/>	<input type="button" value=""/>
Program 4	<input type="text" value="2004"/>	Disabled <input type="button" value="v"/>	<input type="button" value=""/>	<input type="button" value=""/>
Program 4 Record	<input type="text" value="2005"/>	Disabled <input type="button" value="v"/>	<input type="button" value=""/>	<input type="button" value=""/>
Aux Send 1	<input type="text" value="2006"/>	Disabled <input type="button" value="v"/>	<input type="button" value=""/>	<input type="button" value=""/>
Aux Send 2	<input type="text" value="2007"/>	Disabled <input type="button" value="v"/>	<input type="button" value=""/>	<input type="button" value=""/>
Aux Send 3	<input type="text" value="2008"/>	Disabled <input type="button" value="v"/>	<input type="button" value=""/>	<input type="button" value=""/>
Aux Send 4	<input type="text" value="2009"/>	Disabled <input type="button" value="v"/>	<input type="button" value=""/>	<input type="button" value=""/>
CR Monitor Direc	<input type="text" value="2010"/>	Disabled <input type="button" value="v"/>	<input type="button" value=""/>	<input type="button" value=""/>
CR Monitor	<input type="text" value="2011"/>	Live Stereo <input type="button" value="v"/>	<input type="button" value="OK"/>	<input type="button" value=""/>
CR Headphones	<input type="text" value="2012"/>	Live Stereo <input type="button" value="v"/>	<input type="button" value="OK"/>	<input type="button" value=""/>
Preview	<input type="text" value="2013"/>	Live Stereo <input type="button" value="v"/>	<input type="button" value="OK"/>	<input type="button" value=""/>
Talk to CR	<input type="text" value="2014"/>	Disabled <input type="button" value="v"/>	<input type="button" value=""/>	<input type="button" value=""/>
Guest Headphones	<input type="text" value="2015"/>	Disabled <input type="button" value="v"/>	<input type="button" value=""/>	<input type="button" value=""/>
Studio Monitor	<input type="text" value="2016"/>	Live Stereo <input type="button" value="v"/>	<input type="button" value="OK"/>	<input type="button" value=""/>
Talent Hdphones	<input type="text" value="2017"/>	Disabled <input type="button" value="v"/>	<input type="button" value=""/>	<input type="button" value=""/>
Talkback	<input type="text" value="2018"/>	Disabled <input type="button" value="v"/>	<input type="button" value=""/>	<input type="button" value=""/>

Figure 3-2: StudioEngine output channels screen

Chapter Four:

Troubleshooting

No Connection



Figure 4-1: No connection

If the SoftSurface has been set up and you have pressed the Connect button, the Channel section will be active. Buttons and faders will respond to user action. If the channel section appears to be grayed-out, then a good connection has not been established. This could be due to the device not being active on the network or the device running the wrong software. Element control requires PowerStation version 1.1.3 (or higher) or Element-PS/GPIO version 2.5.0 (or higher). A StudioEngine running software version 2.5.4e (or higher) is required for pure Engine control.

Diagnostic information with StudioEngine

Fader channels

Host name: Engine21

Fader and Headphones Inputs					Feed to Source Outputs			
	Channel:	Source Name:	Status:	Audio:	Channel:	Source Name:	Status:	Audio:
Fader 1	5201	PRD TB 1	OK					
Headphones 1		INTERNAL MON2						
Fader 2	150	SRC 1	OK					
Headphones 2		INTERNAL MON2						
Fader 3	200	R2708 1	OK					
Headphones 3		INTERNAL MON2						
Fader 4	10601	Hybrid1	OK					
Headphones 4		INTERNAL MON2			10601	To: Hybrid1	OK	
Fader 5	3301	Program 1	OK					
Headphones 5		INTERNAL MON2						

Figure 4-2: Engine diagnostics

The StudioEngine can help provide information on the status of sources loaded to channels and active return feeds. Figure 5-2 shows a portion of the Fader channels link. What is shown are the Livewire channels are loaded to a fader, Stream status, and the status of the Feed to Source stream for a Phone source type assigned to Fader 4. Becoming familiar with this page during normal operations may help identify where the error is when things are not working.

Appendix A: Source Profile Details

This appendix provides details on the options available to a Source Profile. Many of the default settings are suitable in most cases, but the functions are made available to customize to the needs of the customer.

Primary source:

This is the Livewire channel (the audio) which the profile defines. Enter in the channel number if known or use the browse button to the right of the text field to search for the desired source.

Source profile name:

This is the name which the user will know the source as. It is limited to 10 characters.

Source name override:

This is a feature that allows the source name to be changed. Three options are available:

- 1) Show sourcename [default]
- 2) Allow application to override
- 3) Show both

This feature requires LABL attribute support from the device which the source originates. Windows IP driver and Node software 2.6.15a or higher.

Source availability:

A checkmark indicates that the source can be loaded to the designated control point. Channels refer to the channels of the surface. CR Monitor input is for monitoring positions located in the control room. Studio monitor input is to the Monitor 2 section.

Source type:

Define the source as one of the options to give muting, mix-minus, and predefined GPIO logic to the profile. Appendix B covers the different GPIO behavior between source types.

- 1) Operator – designation for the microphone of the operator of the surface. Mutes CR monitor and used as source for talkback.
- 2) CR producer – a microphone source which mutes the CR monitors, causes the generation of a backfeed, and has specialized GPIO configuration for talkback options from the this source and items loaded to the surface and in preview.
- 3) CR guest – a microphone source that mutes the CR monitors, causes the generation of a backfeed, and general GPIO control options.
- 4) Studio guest – a microphone source that mutes Studio monitor source (Monitor 2), causes the generation of a backfeed, and general GPIO control options.
- 5) External microphone – a microphone source that does not mute any monitors, causes the generation of a backfeed, and general GPIO control options.
- 6) Line – a generic source which has general GPIO control options.
- 7) Phone – a source that enables phone control options, causes the generation of a mix-minus (backfeed) based on the Feed-to-Source option, and predefined GPIO control.
- 8) Codec – a source that causes the generation of a mix-minus (backfeed) based on the Feed-to-Source option (dual mono) which the Left return channel has IFB, and predefined GPIO control.

9) Computer player – similar to Line type from above but different GPIO control (refer to Appendix B).

Fader mode:

Defines the operational use of the fader. Normal is where the fader function is post the ON/OFF state, Fader Start is where moving the fader above –Infinity turns the channel state ON

1) Normal – The fader controls the gain of the channel

2) Fader Start – If the fader is moved up from the –Infinity position, the channel is turned ON. Moving down to –Infinity turns the channel OFF.

3) Fader Start with Arm – Fader start function with the inclusion of Arm option. Armed state is entered by pressing the ON Button when the fader is down. The armed state means that the channel will go to the ON state when the fader is moved up. When not in this state, moving the fader up has no effect. When in Arm state, the channel OFF lamp will be illuminated. The channel ON lamp will flash a “wink” - a long on-time and short off-time. If the fader is in the up position, using the ON and OFF buttons will act as in the Normal option.

For all fader modes, pressing the OFF button at any time will place the channel in the OFF state.

Phone hybrid ip:

EURO PHONE CONTROL MODE – single hybrid with single dedicated line

Enter in the IP address of the Telos unit

Phone line:

EURO PHONE CONTROL MODE – single hybrid with single dedicated line

Which dedicate line. (example, Nx12 has 12 possible lines to select)

Phone hybrid (for Nx12):

EURO PHONE CONTROL MODE – single hybrid with single dedicated line

Which hybrid is being used. If the Telos unit has multiple hybrids built in, you need to define which hybrid this is from the Telos unit.

Use 2nd show (split mode for Nx12):

EURO PHONE CONTROL MODE – single hybrid with single dedicated line

If using the split mode in an Nx12.

Phone Module ID:

PHONE CONTROLLER MODE – use of the Phone Controller module.

Define which phone module to use. Most cases the only option is 1, but if you had multiple controllers, this is how you would define which one.

Hybrid Nr.(0 - None):

PHONE CONTROLLER MODE – use of the Phone Controller module.

Define which hybrid is being used from the Telos Show Profile. Note that in the Nx12, there is Split mode option which is two shows running, Hybrid 1&2 and Hybrid 3&4. Each show has two hybrid. If connecting to the Hybrid 3&4 show and using the 3rd Hybrid from the Nx12, you are using the 1st hybrid of the show, so the correct entry would be 1 in this field.

Conferencing allowed:

PHONE CONTROLLER MODE – use of the Phone Controller module.

Allows for “button mashing”. This is where you assign multiple lines to a single hybrid. Note that hybrid performance suffers from this technique.

Fixed line (1 to 24, 0 - No):

PHONE CONTROLLER MODE – use of the Phone Controller module.

Dedicate a line to a hybrid.

Preview routing:

- 1) Pre-fader listen – the preview button assigns the source at unity gain to the preview mix.
- 2) After-fader listen – the preview button assigns the source to the preview mix after the gain stage defined by the fader position.

Preview switching mode:

Hopefully these are self explanatory, if not please let us know.

CHANNEL ON turns Preview OFF

Preview ON turns CHANNEL OFF

Hybrid answer mode:

Options intended for single hybrids which have GPIO control.

- 1) Normal, auto answer disabled – [default] no GPIO control
- 2) Channel ON answers hybrid – ON state will trigger GPIO tally
- 3) Channel ON or Preview ON answer hybrid – ON state or preview will trigger GPIO tally

Backward feed enabled/disabled:

By default this is set to enable. If you wish to not allow the profile to cause a backfeed (return feed) to be generated, then you may disable that with this option.

Backward feed dim gain (-30 ... 0 dB):

When a talkback is engaged to this source, the regular feed will be altered; this option provides the definition of the change. Off turns off the regular return feed (an IFB) while “use [value]” defines the gain stage the regular feed will pass through prior to being mixed with the talkback audio.

Logic port:

GPIO function control.

- 1) Disabled – disable GPIO functions associated with this profile
- 2) Exclusive mode – [default] GPIO functions are enabled and not shared if the source is loaded to another fader in the facility.
- 3) Shared mode – GPIO functions are enabled and not constrained by fader/source ownership. Only a single fader in the facility should have ownership of a source. Ownership defines that the fader has GPIO control of the source.

GPIO Ready enabled/disabled:

Ready function is a signal between the source and the surface that indicates the source is ready to play. This function was common back in the day of cart decks. If enabled, the Element will not illuminate the OFF button, while in the OFF state, unless the GPI-Ready pin for the source is active. Otherwise, the OFF button will not illuminate.

Feed to source mode:

Define the mix-minus for the source if the source type requires a mix-minus. The options are composed of the available mixes that are generated by the Surface. Two options require some further definition as they defer from the others.

PHONE – is a special mix that is anything assigned to Program 4 but is PRE fader and PRE ON/OFF (unity gain).

Auto (PGM1/PHONE) – [default] defines the backfeed to switch between two different sources based on the state of the channel. If the channel is ON, the backfeed will be Program 1. If the channel is OFF, the backfeed is PHONE (mix).

Signal mode:

- 1) Stereo – [default] the profile will be composed of the Left and Right channels
- 2) Left – the profile will be a dual mono source based on the Left channel of the source
- 3) Right – the profile will be a dual mono source based on the Right channel of the source
- 4) Sum – the profile will be a dual mono source based on the sum of Left and right channels

Signal mode lock:

Signal mode can be defined in the profile and also changed from the surface itself. If the desire is to not allow surface control of the mode, this option allows for the disabling (locked) of this control. Unlocked is the default.

Signal phase:

By default the phase of the source is not altered. If you have a need to change the phase of a channel or both channels, this option provides that control.

Panorama position (-24 .. 24):

Preset pan settings for the profile. Pan is variable in 49 steps, center being 0, far left -24 and far right +24. This setting can be adjusted on the fly by the operator if permitted by Show Profile settings.

Auto-start timer:

Determines whether the count up timer will reset to zero when this profile is turned on.

EQ status:

Enable or disable the EQ function for the profile. The EQ function uses SmartQ™ which 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.

EQ high mode:

Define the behavior of the higher band. Shelf creates a flat response from the 3 band to the end of the higher frequency. Peak creates a response which rolls off from the 3 band point to the end of the higher band.

EQ high frequency (1250 .. 20000Hz):

Sets the frequency for the high-band. Variable from 1.25 kHz to 20 kHz.

EQ mid frequency (125 .. 2000Hz):

Sets the center frequency of the midrange band. Variable from 125 Hz to 2.0 kHz.

EQ low frequency (20 .. 320Hz):

Sets the center frequency of the low band. Variable from 20 Hz to 320 Hz.

EQ high gain (-25 .. 15dB):

Sets the gain for boosting or cutting the band up to 40 dB of adjustment.

EQ mid gain (-25 .. 15dB):

Sets the gain for boosting or cutting the band up to 40 dB of adjustment.

EQ low gain (-25 .. 15dB):

Sets the gain for boosting or cutting the band up to 40 dB of adjustment.

Noise gate status:

Turns the voice processor's noise gate on and off. Only functional for voice based profiles (Microphones, Codecs, Phones).

Compressor status:

Turns the voice processor's compressor function on and off. Only functional for voice based profiles (Microphones, Codecs, Phones).

De-esser status:

Turns the voice processor's de-esser module on and off. Only functional for voice based profiles (Microphones, Codecs, Phones).

Noise gate threshold (-50 .. 0dB):

Determines the point at which the Noise Gate will open.

Noise gate depth (-30 .. 0dB):

Sets the amount of noise reduction that will be applied to the audio when closed.

Compressor threshold (-30 .. 0dB):

Determines the point at which the Compressor will reduce the level of the audio signal.

Compressor ratio (1:1 .. 16:1), use values 1.0 to 16.0:

The amount of gain reduction is determined by a ratio: the ratio of 16:1 means that if input level is 16 dB over the threshold, the output signal level will be 1 dB over the threshold. The gain (level) has been reduced by 15 dB. The ratio of 1:1 will have no reduction.

De-esser threshold (-20 .. 0dB):

Define the value at which the De-esser begins to reduce excess sibilant consonants of the human voice. Sibilance lies in frequencies anywhere between 2 kHz-10 kHz, depending on the source.

De-esser ratio (1:1 .. 8:1), use values 1.0 to 8.0:

Similar to the compressor ratio, the amount of reduction is determined by a ratio: the ratio of 8:1 means that if input sibilance is 8 dB over the threshold, the output signal level will be 1 dB over the threshold.

Compressor mode:

Freeze option allows for the a gain reduction to remain for the short pauses that are natural in voice communication.

Post-processing trim gain (-20 .. 20dB):

Some processing settings will result in a reduction of overall signal level. This setting allows you to adjust the amount gain after dynamic processing to compensate for any gain lost by compressor processing.

Record mode:

Define how the profile audio will enter into the Program 4 Record mix. A common desire is to have phone interviews separated into two different channels to simplify editing.

- 1) Stereo – No alteration, the Left and Right channels of the profile will enter into the Program 4 Record mix.
- 2) If Phone or Codec Left, otherwise Right – If the profile is defined as a PHONE or CODEC, the audio will be mono summed into the left channel of the Program 4 Record mix.
- 3) Summed-Mono to Left – The profile audio will be summed into the left channel of the Program 4 Record mix
- 4) Summed-Mono to Right - The profile audio will be summed into the right channel of the Program 4 Record mix

Appendix B: GPIO Tables

Operator's Microphone Logic and Producer's Microphone Logic

INPUTS	FUNCTION	PIN	NOTES
GPI-1	ON Command	11	Turn channel ON
GPI-2	OFF Command	12	Turn channel OFF
GPI-3	TALK Command (to Monitor 2)	13	Activates TALK TO MON2 function
GPI-4	MUTE Command	14	Mutes channel
GPI-5	TALK Command (to Previewed Source)	15	Activates TALK button on any source in Preview

OUTPUTS	FUNCTION	PIN	NOTES
GPO-1	ON Lamp	1	Active when ON_State unless TALK or MUTE are active
GPO-2	OFF Lamp	2	Active when OFF_State
GPO-3	TALK Lamp (to Monitor 2)	3	Active when TALK TO MON2 is active
GPO-4	MUTE Lamp	4	Active when MUTE is active
GPO-5	TALK Lamp (to Previewed Source)	5	Active when TALK TO PREVIEW is active

CR Guest Microphone Logic and Studio Guest Microphone Logic

INPUTS	FUNCTION	PIN	NOTES
GPI-1	ON Command	11	Turn channel ON
GPI-2	OFF Command	12	Turn channel OFF
GPI-3	TALK Command (to CR)	13	Mutes channel and routes source to CR Preview
GPI-4	MUTE Command	14	Mutes channel
GPI-5	TALK Command (to Source)	15	Activates TALK button on channel

OUTPUTS	FUNCTION	PIN	NOTES
GPO-1	ON Lamp	1	Active when ON_State unless TALK or MUTE are active
GPO-2	OFF Lamp	2	Active when OFF_State
GPO-3	TALK Lamp (to CR)	3	Active when TALK to CR is active
GPO-4	MUTE Lamp	4	Active when MUTE is active
GPO-5	TALK Lamp (to Source)	5	Active when TALK is active

Line Input Logic

INPUTS	FUNCTION	PIN	NOTES
GPI-1	ON Command	11	Turn channel ON
GPI-2	OFF Command	12	Turn channel OFF
GPI-3	Preview Command	13	Turns Preview on
GPI-4	Reset Command	14	Turns channel OFF without sending STOP pulse
GPI-5	Ready Command	15	Illuminates OFF lamp indicating READY, requires GPIO Ready to be enabled

OUTPUTS	FUNCTION	PIN	NOTES
GPO-1	ON Lamp	1	Active when ON_State
GPO-2	OFF Lamp	2	Active when OFF_State (and READY)
GPO-3	Preview Lamp	3	Active when Preview is active
GPO-4	Start Pulse	4	100 msec pulse when channel changes to ON_State.
GPO-5	Stop Pulse	5	100 msec pulse when channel changes to OFF_State

Codec Logic

INPUTS	FUNCTION	PIN	NOTES
GPI-1	ON Command	11	Turn channel ON
GPI-2	OFF Command	12	Turn channel OFF
GPI-3	TALK Command (to CR)	13	Mutes channel and routes source to CR Preview
GPI-4	MUTE Command	14	Mutes channel
GPI-5	TALK command (to Source)	15	Activates TALK button on channel

OUTPUTS	FUNCTION	PIN	NOTES
GPO-1	ON Lamp	1	Active when ON_State unless TALK or MUTE are active
GPO-2	OFF Lamp	2	Active when OFF_State
GPO-3	TALK Lamp (to CR)	3	Active when TALK to CR is active
GPO-4	MUTE Lamp	4	Active when MUTE is active
GPO-5	TALK Lamp (to Source)	5	Active when TALK is active

Telephone Hybrid Logic

INPUTS	FUNCTION	PIN	NOTES
GPI-1	ON Command	11	Turn channel ON
GPI-2	OFF Command	12	Turn channel OFF
GPI-3	Preview Command	13	Turns Preview on
GPI-4	Reset Command	14	Turns channel OFF without sending STOP pulse
GPI-5	Ready Command	15	Illuminates OFF lamp indicating READY, requires GPIO Ready to be enabled

OUTPUTS	FUNCTION	PIN	NOTES
GPO-1	ON Lamp	1	Active when ON_State
GPO-2	OFF Lamp	2	Active when OFF_State
GPO-3	Preview Lamp	3	Active when Preview is active
GPO-4	Start Pulse	4	100 msec pulse when channel ON_State or Preview is active, requires Hybrid answer mode set accordingly
GPO-5	Stop Pulse	5	100 msec pulse when channel changes to OFF_State

Computer Playback Device Logic

INPUTS	FUNCTION	PIN	NOTES
GPI-1	ON Command	11	Turn channel ON
GPI-2	OFF Command	12	Turn channel OFF
GPI-3	Preview Command	13	Turns Preview on
GPI-4	not used	14	
GPI-5	Ready Command	15	Illuminates OFF lamp indicating READY, requires GPIO Ready to be enabled

OUTPUTS	FUNCTION	PIN	NOTES
GPO-1	Next Pulse	1	100 msec pulse when ON button is pressed and in ON_State
GPO-2	OFF Lamp	2	Active when OFF_State (and READY)
GPO-3	Preview Lamp	3	Active when Preview is active
GPO-4	Start Pulse	4	100 msec pulse when channel changes to ON_State.
GPO-5	Stop Pulse	5	100 msec pulse when channel changes to OFF_State

The Show Profile defines a logic channel for Control Room monitor and Studio monitor. Assigning a number value to the appropriate field will assign the following functions to the channel. Assigning a GPIO port to the same number will then configure the port with these functions.

This is a common way to create an ON AIR light tally.

Control Room Monitor Logic

INPUTS	FUNCTION	PIN	NOTES
GPI-1	MUTE CR command	11	Mutes CR monitor
GPI-2	DIM CR Command	12	Dimming of CR monitor
GPI-3	Enable EXT PREVIEW Command	13	EXT PREVIEW source routed to Preview
GPI-4	TALK TO EXT Command	14	TALK to EXTERNAL activated
GPI-5	not used	15	

OUTPUTS	FUNCTION	PIN	NOTES
GPO-1	CR ONAIR lamp	1	Active when CR monitor is muted
GPO-2	DIM CR lamp	2	Active when CR monitor is DIMMED
GPO-3	Preview Lamp	3	Active when Preview is active
GPO-4	TALK TO EXT Lamp	4	Active when TALK TO EXT is active
GPO-5	TALK to CR Lamp	5	Active if any source is talking to CR

Studio Monitor Logic

INPUTS	FUNCTION	PIN	NOTES
GPI-1	MUTE studio (monitor 2) command	11	Mutes studio monitor
GPI-2	DIM studio (monitor 2) Command	12	Dimming of studio monitor
GPI-3	Remote Up Timer trigger	13	Resets timer and starts
GPI-4	Remote Down Timer start	14	Resets timer and starts
GPI-5	not used	15	

OUTPUTS	FUNCTION	PIN	NOTES
GPO-1	Studio (monitor 2) ONAIR Lamp	1	Active when studio monitor is muted
GPO-2	DIM Studio (monitor 2) Lamp	2	Active when studio monitor is DIMMED
GPO-3	Up Timer start Lamp	3	100 msec pulse when up timer starts
GPO-4	Down Timer start Lamp	4	100 msec pulse when down timer starts
GPO-5	TALK TO STUDIO (monitor 2) Lamp	5	Active when TALK TO STUDIO is active

Appendix C: Show Profile Details

From the configuration screen you have the option to add new shows, select a show for edit, or delete shows. One exception to deleting is the Default Show Profile can not be removed (deleted).

On editing a Show Profile, there are the following options

Channel #

Auxillary send & return

Monitor section

Record mode

Group start

Many of the options available through the channel link are covered in the Source profile section. The setting in the Show Profile supersedes the Source Profile or allows maintaining the Source Profile setting.

Channel

Source ID

Define the Source Profile to load to the channel

Feed to source mode

Override the Source Profile setting with the options available or retain Source Profile settings

Auto-start timer

Override the Source Profile setting or retain Source Profile settings

Signal mode

Override the Source Profile setting or retain Source Profile settings

Signal mode locked

Override the Source Profile setting or retain Source Profile settings

Fader trim gain

Override the Source Profile setting or retain Source Profile settings

Fader trim lock

Override the Source Profile setting or retain Source Profile settings

Panorama position

Override the Source Profile setting or retain Source Profile settings

Phase

Override the Source Profile setting or retain Source Profile settings

The entire EQ section of options

Override the Source Profile setting or retain Source Profile settings

Assign to PGM1

Enable the channel to the Program 1 mix (On) or remove the channel from the Program 1 mix (Off). “Retain existing setting” will not change any state.

Assign to PGM2

Enable the channel to the Program 2 mix (On) or remove the channel from the Program 2 mix (Off). “Retain existing setting” will not change any state.

Assign to PGM3 and the remainder Assign options up to AUX D follow the same logic.

AUX pre/post fader

Enable the channel to an AUX mix either PRE fader or POST fader. The non committal option is to make no change, “Retain existing setting”

AUX SEND gain

Declare the gain stage which the channel will pass through entering into the mix. A value of OFF is not allowing any audio through. A value of 0.0 is unity gain, no gain adjustment.

Noise gate, Compressor, and De-esser settings (Dynamic processing)

Override the Source Profile setting or retain Source Profile settings

Post-processing trim gain

Override the Source Profile setting or retain Source Profile settings

Backward feed dim gain

Override the Source Profile setting or retain Source Profile settings

Channel On/Off status

A setting of On will force the channel into the ON_State when the source loads

Control lock map

A check mark indicates that the particular button or fader will be LOCKED, meaning that the user will not be able to use the function associated with the button (or fader).

Group start

Define the status of the channel as the Master of a group of faders, a Slave to a group of faders, or Independent. A group of channels will follow the ON or OFF state of the Master channel

Motor. Fader Position

Define the fader position once the show is loaded. For example, if you want the fader at the -0- (unity gain) position, enter in 0 for Use. Off is the bottom position.

Individual headphone section

Current source

Defines the audio source of the unique headphone feed (backfeed) for any microphone source, excluding the Operator Microphone.

Source for Preset 1 & 2

Define the function of two preset buttons available on an accessory module

Headphone master gain

Sets the level of the unique headphone feed (backfeed)

Talkback start volume low limit

Record Mode

Define if the channel is to be engaged into the mix, removed from a mix, or not changed if surface enters into Record mode.

Define if the ON/OFF buttons have control of the channels state.

Auxillary send & return

There are 4 Auxillary (AUX) sends and two returns. From a channel you can engage a channel to be sent to one of the 4 AUX sends. This show configuration defines the master settings of the AUX mixes upon a Show Profile change. The AUX send properties may also be changed in real time from the surface.

AUX SEND (A-D) master gain

Define the gain of the mix. Use = 0.0 is unity gain, no change in gain of the total mix of the send. Negative values are attenuation of the total mix, while positive values are gain added to the total mix. Off value will not allow any audio to pass – OFF.

AUX SEND (A-D) On/Off status

Define if the AUX mix is enabled or not.

AUX RETURN (A-B) gain

Define the gain of the return input. Use = 0.0 is unity gain, no change in gain of the return. Negative values are attenuation of the return, while positive values are gain added. Off value will not allow any audio to pass – OFF.

AUX RETURN (A-B) On/Off status

Define if the return is enabled or not.

AUX RETURN (A-B) signal mode

Define the 2 channel image of the return.

AUX RETURN (A-B) panorama position

Negative 24 is completely to the left, 0 is centered, +24 is completely to the right channel.

AUX RETURN (A-B) assign to PGM

Define which mix the return audio will insert into after the above properties.

AUX RETURN (A-B) source ID

Define the Source Profile that is to be used for the return.

Monitor section

Timer mode:

Manual – no automatic count up timer triggers

Auto-reset – enable count up timer reset on new channel

Auto-add – a new channel will not reset a current count up sequence.

Show Tenths on:

Show tenths of a second on timers.

Preview interlock mode

Disable – require that you select and unselect sources to the preview mix

Enabled – pressing a new channel to Preview will remove the previous channel.

Preview speaker master gain

Define the default preview level

Source ID for EXTERNAL preview

Define the source that will be routed into the preview mix upon the enabling of the External Preview trigger.

Talkback gain

Define the gain of the talkback to CR.

Feed to Source Sum Gain

Define to gain stage that mono-summed return feeds pass through.

Sources for External 1 & 2

Define which Source Profile is assigned to the External monitor buttons.

Control Room Monitor Options

Source

Define the source that will be routed to the CR monitors when the show loads.

CR monitor master gain

Define the default gain position on show load.

Signal mode

Define the two channel image of the CR monitors

CR monitor dim gain

Define the attenuation of the DIM function. A value of 0 will have no impact. OFF will turn off the channel completely under DIM.

CR monitor muted state gain

In most environments, the CR monitor will mute with a microphone ON. If your needs are different, you can use this option to change the muting performance.

GPIO channel for CR monitor

Define a channel which can be assigned to a GPIO port and have that port follow the CR monitor logic as defined in Appendix-B

Control Room Headphone Options

Source

Define the source that will be routed to the CR headphone when the show loads.

CR headphone master gain

Define the default gain position on show load.

Signal mode

Define the two channel image of the CR headphone.

CR headphone independent

Define if the operator headphones will follow what is selected to the monitors (linked mode) or independent control.

Preview-in-headphone mode

Define if Preview mix will be inserted into the Headphone mix and in full stereo or split mode.

CR headphone EQ settings

Define if the CR headphone will be processed through EQ and what are the properties of the 3 band EQ.

Studio Monitor Options

Source

Define the source that will be routed to the CR headphone when the show loads.

Source IF for External

Define the external source if Ext is selected.

Studio monitor master gain

Default gain value for the monitors

Studio monitor dim gain

Default dim level for the DIM functions on the monitors

Studio monitor muted state gain

In most cases muting is desired, if the application require different performance, use this option to define that difference in performance.

GPIO channel for studio monitor

Define a channel which can be assigned to a GPIO port and have that port follow the logic function of the Studio monitor as defined in Appendix-B

Record mode

Enable the Record mode function with the Show Profile.

Assign a channel which can be assigned to a GPIO port allowing the port to provide triggers on the entrance and exit of the Record mode.

GPO-1 = Latched on when in Record mode

GPO-2 = Latched on when not in Record mode

GPO-3 = not used

GPO-4 = Pulse when Record mode started

GPO-5 = Pulse when Record mode ended

Group start

Per channel, within the Show Profile you can define a channel to be master and others to be slave. This allows a single ON or OFF button of the master channel to be pressed and the slave channels will follow the state.

Boiling pitch, feathers

An angry mob lights the night

Copper thieves must die!

Appendix D: StudioEngine

The StudioEngine is the work horse that does all the digital mixing. After initial setup, typically the Studio Engine remains in the rack doing the computational functions.

The Appendix will provide a brief summary of the HTML pages available on this device.

Fader Channels

This page provides a snap shot of the state of each fader in the Studio Engine. Providing information on the LW Channel loaded, the name of the source, a status state (OK or Err), and an audio level indication:

No audio:	< -52 dBfs : Color = blank
Audio present:	> -52 dBfs : Color = green
High:	> -12 dBfs : Color = yellow
Clipping:	> -3 dBfs : Color = red

This is a snap shot and will not update unless you reload the page.

Return and monitor output

Similar to the Fader Channels page. This page shows

Auxiliary returns,

Control Room monitor and headphone channels,

Studio Monitor channel,

Source Preview (Cue) channel,

External Preview channel.

Program and Monitor Output

The StudioEngine takes audio from the network and mixes the channels together based on the control information received by the surface. The surface controls many of these mixes and each one can be enabled into the network as a new source which you route to another location in the network. For example, Program 1 is the final mix from the surface, the Program 1 option should be enabled and given a unique channel number. This “source” can be routed to an Analog output of a node which will provide an analog output of the mix created with the surface. The following is a list of these options.

- Program 1 - 4: Your console’s main Program bus outputs.
- Program 4 Record: Special output for recording devices which feeds the contents the Program 4 bus post-fader and pre-On/Off switch.
- Aux Send 1 - 4: Your console’s stereo Auxiliary Send buses.
- CR Monitor Direct: This is the output of the monitor selector before the operator’s volume controls and mutes; useful for feeding a Producer’s position, or any other monitoring station with independent headphone volume controls.
- CR Monitor: Contains whatever audio source your operator has selected to feed the Control Room monitors.

- CR Headphones: Contains whatever audio source your operator has selected to feed the Control Room headphones.
- Preview: The feed to the Preview (Cue) speaker.
- Talk to CR: This channel feeds audio to the Control Room talkback channel whenever the Talk button is pressed on any Studio Accessory panel.
- Guest Headphones: Contains whatever audio source has been selected to feed Monitor 2.
- Studio Monitor: Contains whatever audio source your operator has selected to feed Monitor 2, post the volume knob for monitor 2.
- Talent Headphones: Contains whatever audio source has been selected to feed Monitor 2 and receives IFB if the talk to Monitor 2 button is pressed.
- Talkback: When any console Talk key is pressed, CR Mic audio is routed, pre-fader, to this channel.

Integrated Omnia Processing - StudioEngine contains special audio processing presets by Omnia that you can apply to either:

- Control Room Headphone channel.
- Program 1 in all monitoring paths

This is provided so that you can simulate the “air sound” provided by an on-air processor, so that your jocks can have real-time Program monitoring that sounds like “air” without having to set up a separate processor chain.

V-Mixer and V-Mode

The StudioEngine does the computations for the surface mixing functions and also includes some auxiliary mixing functions. A V-mix is composed of 5 inputs with one sub mix out. Each input is considered a Fader and there are 5 Fader outs. A Fader out is an optional output that follows the stages of the VMIX input. Each V-mix has an ON/OFF stage, a GAIN stage and TIME variables. The TIME variables are used to define the “Fader-up” or “Fader-down” time as a fader enters into ON state or OFF state. There are 16 V-mixes available in the Studio Engine. V-Mixers are utility mixers that can be used for any additional needs that a facility might have from static mixing to dynamic mixing. Pathfinder offers different options to control V-mixes either from a user interface, external triggers, or through automation.

V-Mode provides an auxiliary function to take an internal source from the Engine or an External source and manipulate it per the Audio mode options and create an audio stream to your special needs.

Audio Mode is the magic box that transforms your stereo stream into something else. You may choose from any of the following options:

- Pass Stereo: Passes the Left and Right channels as is, without modification.
- Upmix from L: In Surround mixing mode, takes Left channel and creates 5.1 upmix. In Stereo mixing mode, creates mono sum from Left channel.
- Upmix from R: In Surround mixing mode, takes Right channel and creates 5.1 upmix. In Stereo mixing mode, creates mono sum from Right channel.
- Upmix from L+R: In Surround mixing mode, sums Left and Right channels and creates 5.1 upmix. In Stereo mixing mode, creates mono sum from both channels.
- Upmix from stereo: In Surround mixing mode, creates 5.1 upmix from stereo input. In Stereo mixing mode, passes Left and Right channels without modification.
- Downmix from 5.1: In Surround mixing mode, takes 5.1 Surround source and creates stereo downmix. In Stereo mixing mode, mutes audio (do not use).
- Pass all: In Surround mixing mode, passes 5.1 Surround input without modification. In Stereo mixing mode, passes

Left and Right channels without modification.

- Split L -6: Stereo mixing mode only. Sums Left and Right channels, reduces gain of the sum by 6dB and places mix on Left output channel.
- Split R -6: Stereo mixing mode only. Sums Left and Right channels, reduces gain of the sum by 6dB and places mix on Right output channel.
- Split L -3: Stereo mixing mode only. Sums Left and Right channels, reduces gain of the sum by 3dB and places mix on Left output channel.
- Split R -3: Stereo mixing mode only. Sums Left and Right channels, reduces gain of the sum by 3dB and places mix on Right output channel.
- Combine L,L: Left channel of first VMIX input is sent to Left output channel, and Left channel of second VMIX input is sent to Right output channel.
- Combine L,R: Left channel of first VMIX input is sent to Left output channel, and Right channel of second VMIX input is sent to Right output channel.
- Combine R,L: Right channel of first VMIX input is sent to Left output channel, and Left channel of second VMIX input is sent to Right output channel.
- Combine R,R: Right channel of first VMIX input is sent to Left output channel, and Right channel of second VMIX input is sent to Right output channel.
- Combine LR, 5.1: Used in Surround mode to construct an eight-channel (5.1+Stereo) combined output. Takes Left and Right from first VMIX input and combines it with the 5.1 audio from the second VMIX input.
- Combine 5.1, LR: Same as above, but takes 5.1 audio from the first VMIX input and combines it with Left and Right from second VMIX input to create 5.1+Stereo combined output.

Stream statistics

This page provides information about all active audio streams in and out of the StudioEngine. This is useful primarily for diagnostics and contains no settings that can be modified. Axia tech support may ask you to report information reported on this page. In general, errors and defects should be minimal. DSP underruns should also be very low except for IP-Audio sources under certain circumstances. Contact tech support if you are experiencing problems and need help in this area.

Options

The properties in this page are in place to support legacy customers with older surfaces.

Network

- Host Name: Enter a unique name for this particular StudioEngine. This name is displayed on various information and diagnostic screens.
- Network Address: This field contains the IP address you've assigned for this StudioEngine.
- Netmask: The submask for your IP-Audio Network.
- Gateway: If your Axia network has a network gateway to allow outside access, enter its IP address here.
- Connected Console: Displays the IP address of the Element console that is connected to your Studio Engine.

System

The System Status section of the page provides information about the physical state of the StudioEngine.

- System Status and Network Status inform you as to the health of the StudioEngine and its connection to the Axia network. Both of these indicators must be OK for normal operation.
- CPU Usage indicates how hard the Engine is working. This value should never exceed 95%.
- Link Speed tells you the type of link with which the Engine is connected to the local network switch. This should always be 1 Gbs.
- Network Usage In and Out show how much bandwidth is being consumed on the Engine-to-switch link.
- Temperature monitors the temperature of the StudioEngine CPU. If the temperature exceeds operating limits, a warning will appear on the Engine's front-panel display.
- Fan 1 and Fan 2 Speed: Monitors the fan speeds of the Engine's two internal cooling fans.

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

The Restart section lets you remotely Restart the StudioEngine, performing a complete reboot. Of course, if the StudioEngine is rebooted, it will stop delivering audio until it has reloaded its operating system and the StudioEngine software. This will take only 1-2 minutes but it will seem like forever!

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

By default, access is open.

The Remote Access Password fields allow you to change the user password for HTTP access to the StudioEngine. By default, the user name is user and there is no password.

The Hardware and Firmware section is where you upload and apply new operating software for the StudioEngine.

- Use the File field and the Browse and Upload buttons to upload new software to the Engine.
- The Commit to Bank 0 button. Pressing this button takes the software in Bank 1 and loads it into Bank 0, which opens Bank 1 for new software.
- You can run Engine software from either Bank 0 or Bank 1. Simply click on the radio button next to the software bank you wish to run from, and click Apply Selected Bank. The Engine will reboot after changing banks, so plan this operation for off-hours or switch program output to another studio to avoid program interruptions.

Diagnostics

The use of this page will be by the request of the Engineers at Axia to look at internal states of the running Engine.

Appendix E: Warranty

Telos Alliance 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 or on behalf of TLS Corp. and its affiliated companies, collectively doing business as The Telos Alliance (hereinafter “Telos”).

With the exception of software-only items, the Products are warranted to be free from defects in material and workmanship for a period of five (5) years from the date of receipt of such Product by the end-user (such date of receipt the “Receipt Date”). Software-only items are warranted to be free from defects in material and workmanship for a period of 90 days from the Receipt Date. Telos will repair or replace (in its discretion) defective Products returned to Telos within the warranty period, subject to the provisions and limitations set forth herein.

This warranty will be void if the Product: (i) has been subjected, directly or indirectly, to Acts of God, including (without limitation) lightning strikes or resultant power surges; (ii) has been improperly installed or misused, including (without limitation) the failure to use telephone and power line surge protection devices; (iii) has been damaged by accident or neglect. As with all sensitive electronic equipment, to help prevent damage and or loss of data, we strongly recommend the use of an uninterruptible power supply (UPS) with all of our Products. Telos products are to be used with registered protective interface devices which satisfy regulatory requirements in their country of use.

This Warranty is void if the associated equipment was purchased or otherwise obtained through sales channels not authorized by Telos.

EXCEPT FOR THE ABOVE-STATED EXPRESS WARRANTY, TELOS MAKES NO WARRANTIES, EXPRESS OR IMPLIED (INCLUDING IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE).

In no event will Telos, its directors, officers, employees, agents, owners, consultants or advisors (its “Affiliates”), or authorized dealers or their respective Affiliates, 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, the Product must be registered via Telos’ website (found at: <http://telosalliance.com/legal/warranty>) at time of receipt by end-user and notice of a warranty claim must be received by Telos within the above stated warranty period and warranty coverage must be authorized by Telos. Contact may be made via email: support@telosalliance.com or via telephone: (+1) 216-241-7225. If Telos authorizes the performance of warranty service, the defective Product must be delivered to: Telos, 1241 Superior Avenue, Cleveland, Ohio 44114 or other company repair center as may be specified by Telos at the time of claim.

Shipping Costs and Warranty Service:

If the date the customer's notice of warranty claim is received by Telos (such date the "Warranty Claim Notice Date") is within the first 90 days following the Receipt Date, Telos will pay the costs of shipping such warranted Product to and from the end user's location, and the cost of repair or replacement of such warranted Product.

If the Warranty Claim Notice Date occurs after the first 90 days following the Receipt Date and before the end of the second (2nd) year, the customer will pay the freight to return the warranted Product to Telos. Telos will then, at its sole discretion, repair or replace the warranted Product and return it to the end user at Telos' expense.

If the Warranty Claim Notice Date occurs between the end of the second (2nd) year following the Receipt Date and the completion of the fifth (5th) year, the customer will pay the costs of shipping such warranted Product to and from the end user's location. Telos will then, in its sole discretion, repair or replace the warranted Product at Telos' expense. Telos also reserves the right, if it is not economically justifiable to repair the warranted Product, to offer a replacement product of comparable performance and condition direct to the customer at a discounted price, accepting the failed warranted Product as a trade-in.

The end user will in all cases be responsible for all duties and taxes associated with the shipment, return and servicing of the warranted Product.

No distributor, dealer, or reseller of Telos products is authorized under any circumstances to extend, expand or otherwise modify in any way the warranty provided by Telos, and any attempt to do so is null and void and shall not be effective as against Telos or its Affiliates.

Out of warranty units returned to the factory for repair may be subject to a \$500 evaluation fee, which fee must be prepaid prior to shipping the unit to Telos. If no repairs are required, the \$500 fee will be retained by Telos as an evaluation charge. If repairs are required, the \$500 fee will be applied to the total cost of the repair.

