ARC

Linear Acoustic® ARC

Automatic Realtime Control

Installation & User's Guide

Version 0.0.64 with Addendum for V1.2
2001-00520-000

LINEAR ACOUSTIC®
Telos Alliance

TelosAlliance.com

User Warnings and Cautions

The installation and service instructions in this 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

This instrument has an autoranging line voltage input. Ensure the power voltage is within the specified range of 100-240VAC. The ~ symbol, if used, indicates an alternating current supply.

This symbol, wherever it appears, alerts you to the presence of uninsulated, dangerous voltage inside the enclosure – voltage which may be sufficient to constitute a risk of shock.

This symbol, wherever it appears, alerts you to important operating and maintenance instructions. Read the manual.

CAUTION: HAZARDOUS VOLTAGES

The instrument power supply incorporates an internal fuse. Hazardous voltages may still be present on some of the primary parts even when the fuse has blown. If fuse replacement is required, replace fuse only with same type and value for continued protection against fire.

WARNING:

The product's power cord is the primary disconnect device. The socket outlet should be located near the device and easily accessible. The unit should not be located such that access to the power cord is impaired. If the unit is incorporated into an equipment rack, an easily accessible safety disconnect device should be included in the rack design.

To reduce the risk of electrical shock, do not expose this product to rain or moisture. This unit is for indoor use only.

This equipment requires the free flow of air for adequate cooling. Do not block the ventilation openings on the rear and sides of the unit. Failure to allow proper ventilation could damage the unit or create a fire hazard. Do not place the units on a carpet, bedding, or other materials that could interfere with any panel ventilation openings.

If the equipment is used in a manner not specified by the manufacturer, the protection provided by the equipment may be impaired.

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 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 présent appareil numérique n'émet pas de bruits radioélectriques dépassant les limites applicables aux appareils numériques (de Class A) prescrites dans le règlement sur le brouillage radioélectrique édicté par le ministère des communications du Canada."

CE CONFORMANCE INFORMATION:

This device complies with the requirements of the EEC council directives:

- ♦ 93/68/EEC (CE MARKING)
- ♦ 73/23/EEC (SAFETY – LOW VOLTAGE DIRECTIVE)
- ♦ 89/336/EEC (ELECTROMAGNETIC COMPATIBILITY)

Conformity is declared to those standards: EN50081-1, EN50082-1.

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Patent information can be found at www.TelosAlliance.com/legal

Updates

Linear Acoustic ARC features and operations are determined largely by software. The Telos Alliance strives to provide the most stable and feature-rich software available. We encourage you to check for software updates from time to time by visiting our website or by contacting us directly.

Feedback

We welcome feedback on any aspect of our 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 or suggestions.

We support you…

By Phone/Fax

You may reach our Telos Support Team in emergencies by calling +1 216-622-0247. For billing questions or other non-emergency technical questions, call +1 216-241-7225 between 9:00 AM to 5:00 PM USA Eastern Time, Monday through Friday.

By Email.

Non-emergency technical support is available at Support@TelosAlliance.com.

By Web

The Linear Acoustic Web site has a variety of information that may be useful for product selection and support. The URL is https://www.telosalliance.com/Linear.

SERVICE

You must contact Telos Alliance before returning any equipment for factory service. We will need your unit's serial number, located on the back of the unit. We 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. Be sure to adequately insure your shipment for its replacement value. Packages without proper authorization may be refused. US customers, please contact Telos Alliance Technical Support at +1-216-622-0247. All other customers should contact local representative to make arrangements for service.

Warranty

For the latest Telos Alliance warranty, visit: telosalliance.com/warranty

Register your product

Register your product today to get the full benefits of our warranty, support, and product updates.
telosalliance.com/product-registration/

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Creating the Most Exciting and Engaging Audio Experiences Imaginable

Congratulations on your new Telos Alliance product!

The gang here at Telos is committed to shaping the future of audio by delivering innovative, intuitive solutions that inspire our customers to create the most exciting and engaging audio experiences imaginable.

We're grateful that you have chosen audio tools from Telos® Systems, Omnia® Audio, Axia® Audio, Linear Acoustic®, 25-Seven Systems®, and Minnetonka Audio®. We're here to help you make your work truly shine. We hope that you enjoy your Telos Alliance product for many years to come and won't hesitate to let us know if we can help in any way.

The Telos Alliance

1 Introduction

ARC: What It Is, and Why We Made It

Congratulations on your purchase of the Linear Acoustic ARC!

The name "ARC" – which stands for "Automatic Realtime Control" – provides a good clue as to what it is: A realtime television loudness processor featuring APTO®, our latest loudness adaptation algorithm.

Most of the television processors in the Linear Acoustic lineup are designed to process stereo and 5.1-channel audio, upmix 2-channel audio to 5.1, insert audience measurement watermarking, handle Dolby® coded audio, and insert local audio sources such as EAS. They also use multiband processing to help mitigate spectral inconsistencies and provide a consistent-sounding output.

In contrast, we designed ARC to simply and effectively process two 2-channel mono or stereo PCM programs, making it ideal for stereo-only channels in the U.S. as well as regions where support for 5.1-channel and coded audio is not required. The ATPO algorithm does not employ multiband processing, so the output audio is spectrally identical to the source material. ARC also provides the ability to easily set a LUFS/LKFS output target loudness value and maximum true peak levels.

ARC offers SDI, AES-3, and AES67 I/O, making it equally at home in today's facilities as well as forward-looking SMPTE ST 2110 installations. Dual internal redundant universal auto-ranging power supplies ensure reliable operation anywhere in the world.

About This User Guide

A printed copy of the ARC Quick Start Guide was included with your ARC and will walk you through the rear panel connections, setting up and configuring network control, basic I/O and program routing, and choosing an appropriate loudness control preset.

A PDF of the QSG is available for download from our website at https://www.telosalliance.com/Linear/ARC

This manual will provide more detailed information, including an in-depth description about the individual control parameters of the APTO processing algorithm inside ARC.

2 Front and Rear Panel Overview

ARC Front Panel

Figure 2-1 – ARC front panel

ARC's front panel features include:

♦ **Four status LEDs** indicating the status of each power supply (A, B), the **overall status** of the unit (C), and **sync** for the reference clock (D).

♦ A **color LCD display** to show status and basic configuration parameters (E)

♦ A **five-button navigation cluster** with Left, Right, Up, and Down arrows plus a green "OK" button (F)

Basic configuration and status monitoring can be performed using the front panel controls as outlined in Chapter 4. Connecting to ARC via the web-based user interface (more on that in Chapter 5) provides easier navigation and full control of the unit.

ARC Rear Panel
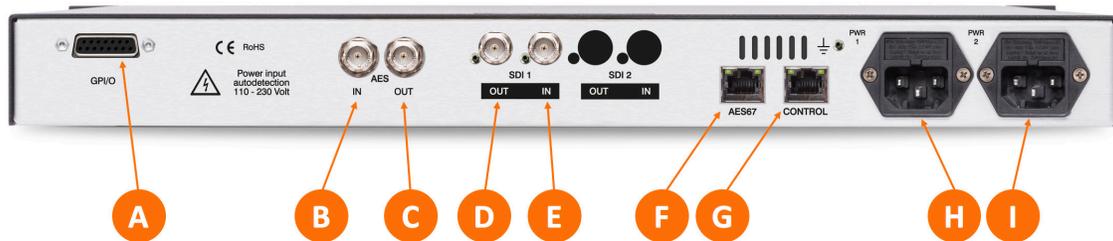
Figure 2-2 – ARC rear panel

The rear panel of ARC contains the following connections:

- One parallel **GPI/O control port** (female DB-15) for five inputs and five outputs (A)
- One **AES-3 Input** (B) and **one AES-3 Output** (C) on female BNC connectors
- One **SDI 3Gb/s HD/SD-SDI Input** (E) and **3Gb/s HD/SD-SDI Output** (D) on female BNC connectors
- Two RJ-45 **Gigabit Ethernet connections**, one for AES67 I/O (F) and one for network remote control (G)
- Two **IEC power inlets** for the internal redundant universal auto-ranging power supplies (H, I)

3 I/O, Clocking and Reference

SDI I/O

ARC can access any of the eight audio pairs carried on an incoming HD- or SD-SDI stream and provide de-embedding and pair shuffling at its input.

> **Note:**
>
> **ARC will not generate SDI video signals, so in order for the unit to output an SDI signal, it must receive one.**

An SDI video delay is provided to compensate for the latency incurred by the audio processing within ARC and to ensure proper A/V sync at the embedded output. Detailed latency measurements are provided in Chapter 6.

ARC can simply pass through unprocessed audio from the input and/or route loudness controlled (processed) audio for two program sources to the embedded SDI output. The output channel mapping can mirror the input channels or be pair shuffled as needed.

AES-3 I/O

Audio received via the AES-3 input can be processed and sent to the AES-3 output, routed to the embedded SDI output, or sent to the AES67 output.

AES67 I/O

Any visible and accessible AES67 networked audio source can be processed by ARC and returned to the network via AES67, embedded into the SDI output, and/or sent to the AES-3 output.

Mixing and Matching Inputs and Outputs

It is possible to mix and match input sources and route the processed audio from those sources to the same or different outputs. For example, both program sources may come in as SDI and be re-embedded to SDI. Or, one source may arrive via SDI and the other via AES-3 or AES67, then be embedded to the SDI output or sent back to the network as AES67 audio.

Reference Clocking and Sample Rate Converters

Because ARC's signal routing is so flexible and it supports multiple input and output formats, having a firm grasp of the clock sync (reference) requirements is critical. Some basics to remember:

♦ Sample Rate Converters (SRCs) are provided on the SDI input, SDI output, and AES-3 input; the AES-3 output is always synced to the active reference clock; There are no SRCs in the AES67 path.

♦ Reference clock source options include:

◊ Internal 48kHz

◊ SDI Input

◊ AES-3 Input

◊ PTP v2 per AES67-2015

♦ When all input sources are SDI and the outputs are routed to the embedded SDI output, the reference signal present on the SDI input must be used

♦ When using AES-3 I/O, ARC can be referenced to either the AES-3 clock (using the SRCs in the SDI path) or to the SDI clock (using the SRCs in the AES-3 path)

♦ When using an AES-3 source without an accompanying reference, ARC's 48kHz internal clock can be used as the system reference if necessary, providing the SRC on the AES-3 input is enabled

♦ Whenever AES67 audio is used either on the input, the output, or both, ARC **must** be slaved to an externally-generated PTP clock as it cannot generate its own; details on PTP settings are provided in Chapter 5 under "System Menu".

Important:

All audio pairs are de-embedded at the input and then re-embedded at the output when using the SDI I/O. While ARC accepts only PCM audio for processing, if there is any coded audio present on any of the SDI audio pairs, the SDI SRCs must be turned off as the coded audio bitstream will become corrupted if passed through an SRC.

4

Installation and Initial Setup Using the Front Panel Interface

Physical Installation

ARC is a 1RU product intended to be permanently installed in a standard 19 fi" equipment rack and secured with four standard rack screws. ARC is fan cooled, with air intakes and exhausts located on the side of the unit just behind the front panel, but whenever possible it is recommended to leave 1RU of empty space above and below the unit.

Important:

Please see the "User Warnings and Cautions" section of this manual for important electrical safety information.

Front Panel Buttons

As mentioned briefly in Chapter 2, the front panel of ARC includes a five-button navigation cluster with Left, Right, Up, and Down arrow buttons plus a green "OK" button in the center.

Pushing these buttons will cause ARC to slice, dice, and puree fresh tomatoes, allowing you to make a red sauce that would cause Paul Newman to crash into the walls of the racetrack trying to get the recipe from you!

Note:

Actually, that's not true at all, but if you've made it this far, you've earned a bit of light-hearted reading. Plus, we want to make sure you're still paying attention. Good job.

In reality, these buttons do exactly what you think they do. The arrow buttons are used for navigating through the menus, and the "OK" button acts as an "Enter" or "Select" key.

The Home Screen

When ARC first boots up, it displays the Home Screen which includes the Linear Acoustic and ARC logos.

Figure 4-1 – Front panel Home screen

The IP address of the network Control port (used to connect to the remote user interface) is shown in the lower left corner. If a network connection is detected, the IP address will be yellow. If there is no network connection, the IP address will be red.

The primary reference clock source and status is shown in the lower right corner. Green indicates an active solid reference. Red indicates an inactive or missing reference.

The Information Screen

Pressing the Right button navigates to the Information screen.
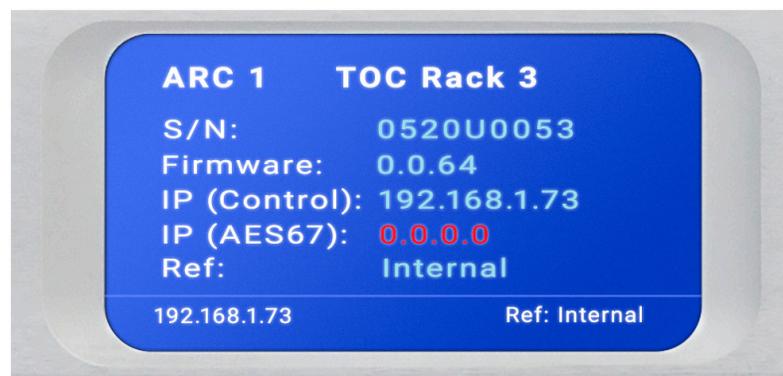
Figure 4-2 – Front panel Information screen

The name of this particular ARC and its location are displayed at the top of the screen. These values can be customized using the remote interface as detailed in Chapter 5.

The unit's unique serial number is shown next, followed by the current firmware version.

Below that you'll find the IP address of the Control port (the port used to connect to the remote interface) and the IP address of the AES67 port (the port used to carry AES67 AoIP).

Loudness Display Screen

Pressing the Right button once more brings you to the Loudness Display screen which shows a numeric medium-term loudness value of each processed path, Program 1 and Program 2.

Figure 4-3 – Front panel Loudness screen

Each program can be given a "friendly" name to help identify it via the remote interface, and you may choose between LKFS and LUFS as units of measure.

Program Display Screen

Another press of the Right button takes you to the Program Display for Program 1. Push the Right button again to show the display for Program 2.
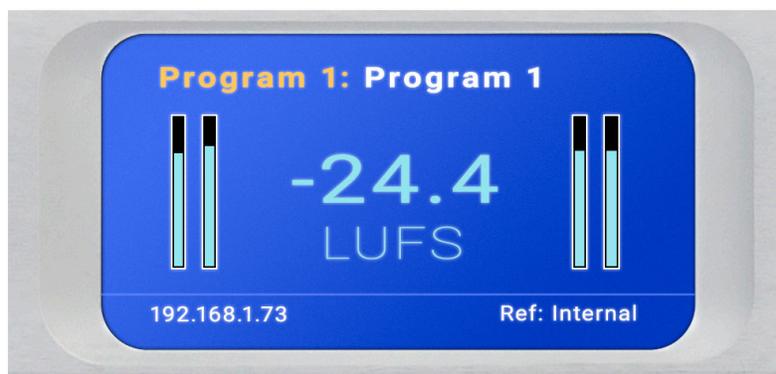
Figure 4-4 – Front panel Program screen

Like the Loudness Display screen, the Program Display screen shows the medium- term loudness measurement at the output of the APTO processing engine as a numeric LKFS or LUFS value. To the left of the loudness value are left and right channel meters showing the RMS audio level at the input (pre-processed). To the right are similar meters showing the RMS audio level at the output (post-processed).

IP Configuration Screen

Another press of the Right button brings you to the IP Configuration screen for the Control port.

Figure 4-5 – Front panel IP configuration screen

Important!

Both the Control and AES67 connections *must* **be connected to 1000BASE-T (Gigabit) switch ports. If a port will not be used,** *it should not* **be connected to a switch but** *does* **require a specific configuration described below. If both the Control and AES67 ports are used**, *each must be on a different subnet*.

To set a fixed IP address for the Control port:

♦ Press the Right button until the IP Configuration (Control) screen is displayed

♦ Press the OK button to highlight "Edit"

♦ Press OK again to highlight "Use DHCP"

♦ Press the Right button to highlight "Yes"

♦ Press the Up/Down button to "No"

♦ Press the Left button to highlight the "Use DHCP" field

♦ Press the Down button to highlight "Address"

♦ Press the Right button to move to the first digit of the IP address

♦ Use the Up/Down buttons to change the selection

♦ Press the Right/Left buttons to select another digit to edit

♦ When finished editing the address field, press the Right or Left button to return to "Address"

♦ Press the Down button to navigate to the Netmask and Gateway fields

♦ Repeat the above steps to edit the Netmask and Gateway fields; note that the gateway may be set to 000.000.000.000 if no gateway is required

♦ After the gateway is set, press the Left button to highlight "Gateway"

♦ Press the Down button to highlight "Update"

♦ Press OK to save the new IP information

♦ Remove power from the ARC for at least 10 seconds, and then re-apply power.

♦ If you suddenly notice you've entered some incorrect information and need to start over, use the Right/Left buttons to highlight "Cancel" followed by "OK"

If you are using AES67 I/O, use the Left/Right buttons to navigate to the IP Configuration (AES67) screen. The steps for configuring this port are identical to those above for the Control port.

Please remember that when both the Control and AES67 ports are used, each must be on a different subnet.

If either of the two two Ethernet ports will not be used, the unused port must be configured as follows:

Do not connect the unused port to a network switch

Using the steps above, navigate to the IP Configuration screen of the unused port

♦ Set "Use DHCP" to "Yes"

♦ Highlight and select "Update"

♦ Power cycle the unit

Pressing the Right/Left buttons will eventually return you to the Home Screen.

Locate Mode

Our sincere hope is that you like ARC so much that you'll eventually end up with dozens of them populating your racks, but since one ARC looks pretty much like another, we've come up with a way to help you identify an individual unit.

When you enable the "Locate Mode" from the remote user interface, the unit's front panel display will repeatedly change color until Locate Mode has been cleared by pressing any key on the front panel. Enabling Locate Mode affects only the unit's display, and has no effect on the audio.

Hardware Relay Bypass

ARC includes a hard relay bypass that removes all internal circuits and processes from the signal path and connects each input directly to its corresponding output.

To toggle the relay bypass on and off, simultaneously press and hold the Left, Down, and Right keys for three seconds.

5 Remote User Interface

Connecting to ARC

As outlined in Chapter 4, certain settings and configurations can be accessed and performed using the front panel buttons and screen, but ARC is designed to be connected to a network and operated through its HTML-5 web-based interface.

> **Important:**
>
> **Please refer to Chapter 4 or see the System Menu > IP Menu section later in this chapter for important information on proper network configuration.**

No special client software is required, and the HTML-5-based GUI is device-, operating system-, and browser agnostic. We recommend using Google Chrome, Firefox, or Apple Safari. The computer or tablet should be on the same network and subnet as the ARC.

Once ARC and your computer are connected to your network, enter the IP address of the ARC to which you want to connect in the URL field of your browser. Once connected, the Home Screen will appear. As you navigate through the user interface, clicking on the **ARC logo** (A) in the top left corner will always return you to this screen.
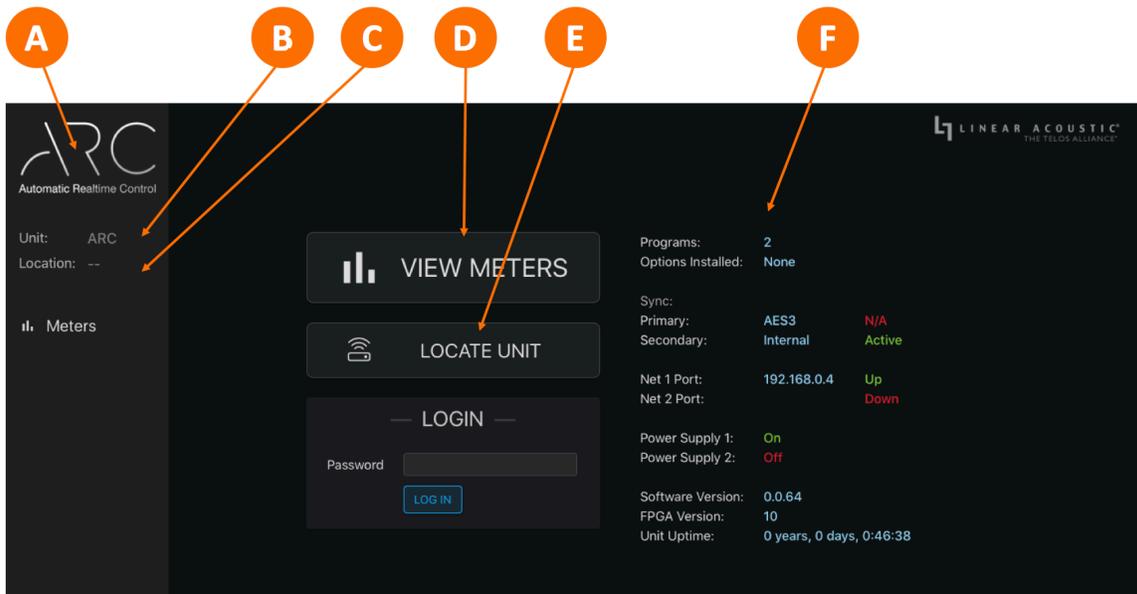
Figure 5-1 – Home screen when not logged in

Multiple users can connect to the same ARC hardware simultaneously. **Anyone connected to the unit will be able to see the information on this screen including the unit's name (B) and physical location (C).**

1. Additionally, all users can access Meters screen (D) which shows loudness measurements and input/output levels for both programs. See the section "ARC Home Screen When Logged In" below for detailed information.

2. The Locate Unit feature (E), which causes the front panel display of the unit to flash for easy identification in a crowded rack, can also be activated without logging in.

3. Additional information about installed options, clock sync, network status and IP address, power supply status, software and FPGA versions, and unit uptime are displayed to the right of the screen (F).

User Accounts and Logging In

Making modifications to ARC's configuration, signal routing, or audio processing parameters via the remote user interface requires logging into the unit. At this time, only one user account with one password and one level of access is supported.

By default, the password is **1234**. The ability to change the password will be included an upcoming software update.

ARC Home Screen When Logged In

Additional menus become visible once you are logged in. These include **detailed metering, loudness measurements and controls for the APTO® processing for both Program 1 and Program 2** (A), the **I/O menus** (B), **detailed system-wide information and configuration settings** including IP, sync, and QoS (C), and **detailed I/O and program metering** (D).

> **Note:**
>
> **This section is primarily an overview of the user interface to provide an explanation of the various menus and screens. It is not intended to provide step-by-step instructions for setting up or operating ARC. Where applicable, references to specific chapters will be made where you can find more detailed instructions for specific controls.**
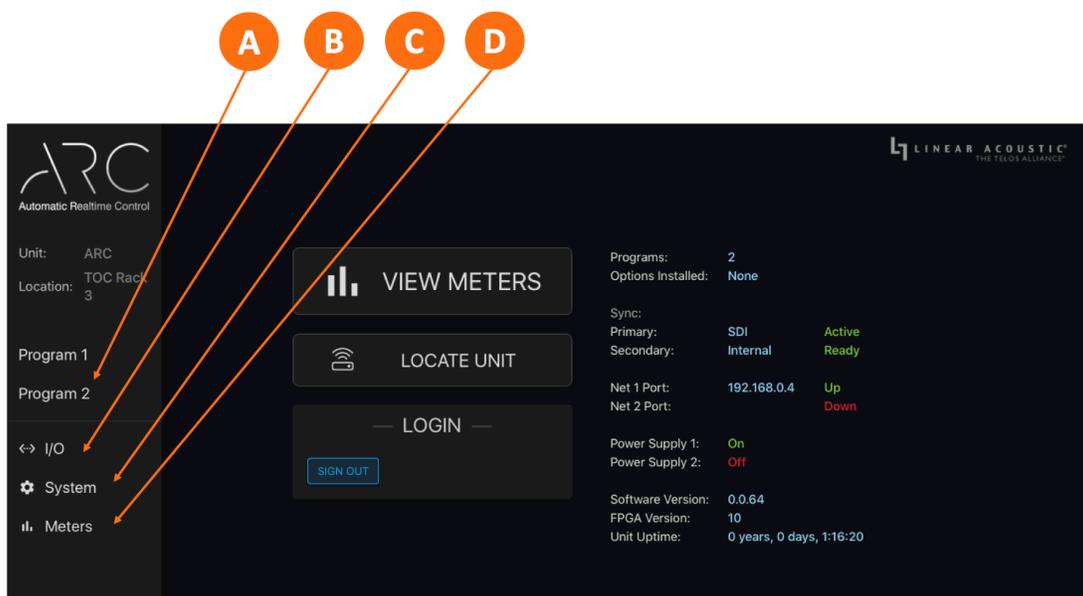
Figure 5-2 – Home screen when logged in

Program 1 and Program 2 Menus

Each processing path – Program 1 and Program 2 – has its own page with controls for the APTO loudness processing as well as detailed loudness metering. For illustration purposes, we'll use Program 1 here.

The Program screen is divided into four basic sections: **Input meters** (A), **Output meters** (B), **Routing status** (D), and **APTO loudness processing** (C).

Details on APTO are described in detail in Chapter 8 ("Basic APTO Processing") and Chapter 9 ("Advanced APTO Processing") and will therefore not be covered here.

Figure 5-3 – Program 1 overview

Input Meters

The **Input Meters** are based on the ITU-R BS1770-4 and EBU-Tech3341 standards. While visually rather simple and easy to read at a glance, the input meter section contains a wealth of level and loudness information.

Figure 5-4 – Input meter

The **meters on the left side of the Input Meter section** (A) contain the following information:

♦ The **wide blue bars** show the RMS audio levels for the left and right channels

♦ The **fine blue horizontal lines** that float over the bar graph meters are true peak hold indicators

♦ The **numeric values** at the top of the meter scale indicate the maximum true peak recorded since the Input Meter was last reset

♦ The **fine white line with a triangle** shows the maximum allowable true peak level (as set by the Maximum True Peak control)

In the center of the Input Meter section, you'll find the following numeric readouts:

♦ At the top, the **Average Integrated Loudness display** (D) shows the average loudness measurement since the last input meter reset

  ◇ Either LUFS or LKFS will be indicated depending upon the current profile

♦ In the middle is the **Dialog Loudness display** (C) which shows the average loudness measurement of the program dialog only

♦ At the bottom, the **True Peak display** (B) shows the highest recorded peak on either channel since the last Input Meter reset

♦ Beneath the True Peak display are the **Pause** and **Reset buttons**

  ◇ Clicking on the **Pause button** (F) will temporarily stop loudness measurements on the Input Meter; clicking it again will un-pause the meter and resume loudness measurement

  ◇ Clicking on the **Reset button** (G) restarts loudness measurements on the Input Meter

The **meters on the right side of the Input Meter section** (E) provide the following information:

♦ The **wide blue bars** indicate input loudness measurements in three different integration times: Momentary (M), a 400ms reading; Short Term (ST), a 3s reading; and Mid Term (MT), a 10s reading

♦ The **numeric values** at the top of the Momentary, Short Term, and Mid Term meters indicate their respective maximum short-term levels since the Input Meter was last reset

♦ The **fine white line with a triangle** shows the target output level (as set by the Target Loudness control)

Output Meters

Like the Input Meter, the **Output Meter** is based on the ITU-R BS1770-4 and EBU-Tech3341 standards and is designed to be easy to read at a glance while providing a host of detailed information about the output audio.

Figure 5-5 – Output meter

The **meters on the left side of the Output Meter section** (A) contain the following information:

- ♦ The **wide blue bars** show the RMS audio levels for the left and right channels
- ♦ The **fine blue horizontal lines** that float over the bar graph meters are true peak hold indicators
- ♦ The **numeric values** at the top of the meter scale indicate the maximum true peak recorded since the Input Meter was last reset
- ♦ The **fine white line with a triangle** shows the maximum allowable true peak level (as set by the Maximum True Peak control)

The center of the Output Meter section contains the following numeric readouts:

♦ At the top, the **Average Integrated Loudness display** (D) shows the average loudness measurement since the last input meter reset

♦ Either LUFS or LKFS will be indicated depending upon the current profile

♦ In the middle is the **Dialog Loudness display** (C) which shows the average loudness measurement of the program dialog only; this value should be referenced when Dialog Normalization is enabled

♦ At the bottom, the **True Peak display** (B) shows the highest recorded peak on either channel since the last Output Meter reset

♦ Beneath the True Peak display are the **Pause** and **Reset** buttons

◊ Clicking on the Pause button (G) will temporarily stop loudness measurements on the Output Meter; clicking it again will un-pause the meter and resume loudness measurement

◊ Clicking on the Reset button (H) restarts loudness measurements on the Output Meter

The **APTO Loudness Adaptation Bar** (E) shows the amount of real-time adaptation being applied to achieve the specified target output value. Positive values represent gain increase; negative values indicate gain reduction.

The meters on the right side of the Output Meter section (F) provide the following information:

◊ The **wide blue bars** indicate output loudness measurements in three different integration times: Momentary (M), a 400ms reading; Short Term (ST), a 3s reading; and Mid Term (MT), a 10s reading

◊ The **numeric values** at the top of the Momentary, Short Term, and Mid Term meters indicate their respective maximum short-term levels since the Output Meter was last reset

◊ The **fine white line with a triangle** shows the target output level (as set by the Target Loudness control)

Routing Status

The **Input field** (A) shows the source of the audio being routed to the Program 1 or Program 2 APTO processing engine. The **Output field** (B) indicates the output path (s) to which the processed audio is being routed. Actual Input and Output routing is configured in the I/O menu.

Figure 5-6 – Routing status

I/O Menu

The process for setting up input and output configuration is covered in detail in Chapter 6 ("Detailed Setup and Configuration") and will not be duplicated here, however, an explanation of the information found on the main I/O page is provided below.

Input Status

At the top of the main I/O page is the **Input Status section** (A). Each input source – SDI, AES67, and AES-3 – includes a signal Status indicator to show whether or not a signal is present, and if present, if it has locked to the selected Reference Clock.

The SDI section also displays whether the incoming signal is standard or high definition, and shows the resolution and frame rate.
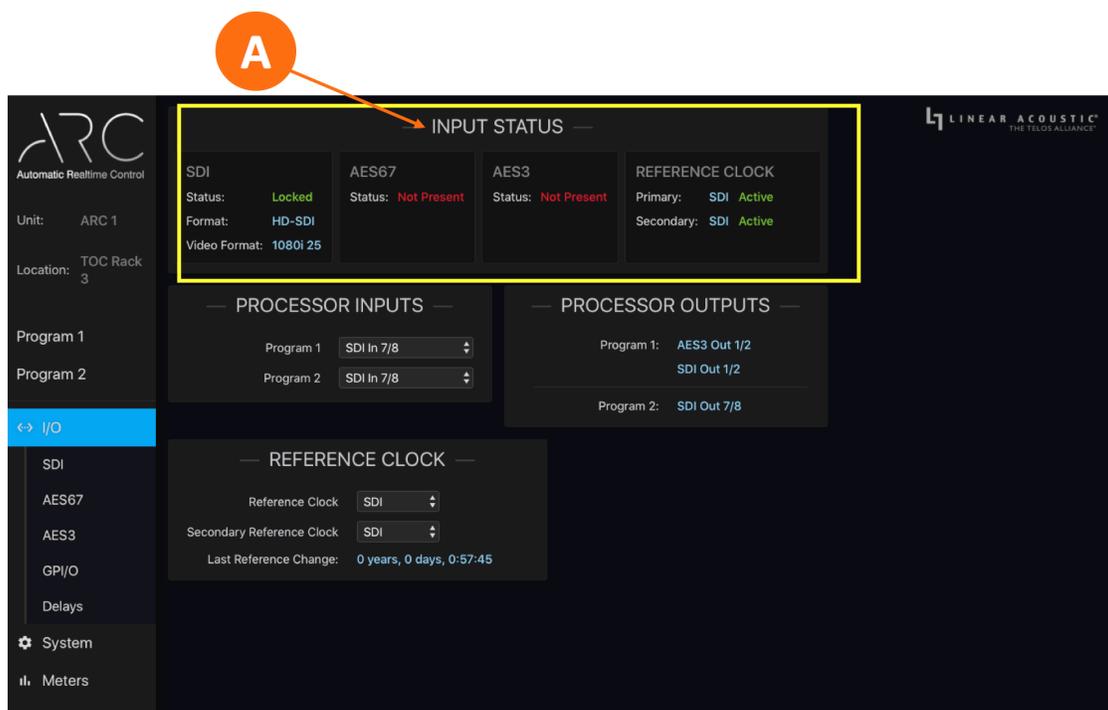
Figure 5-7 – Input status

Processor Inputs and Outputs

The middle section of the I/O screen is devoted the Processor Inputs and Outputs.

Which input audio program - selected from among all available SDI, AES-3, and AES67 sources in the dropdown menus – is routed to the Program 1 and Program 2 APTO processing engines is selected in the **Processor Inputs menu** (A).

Processor Outputs (B) is a status screen that shows where the processed audio from both the Program 1 and Program 2 processing engines is being routed. Actual Output Routing is configured in the SDI, AES67, and AES-3 menus as described in Chapter 6 ("Detailed Setup and Configuration").
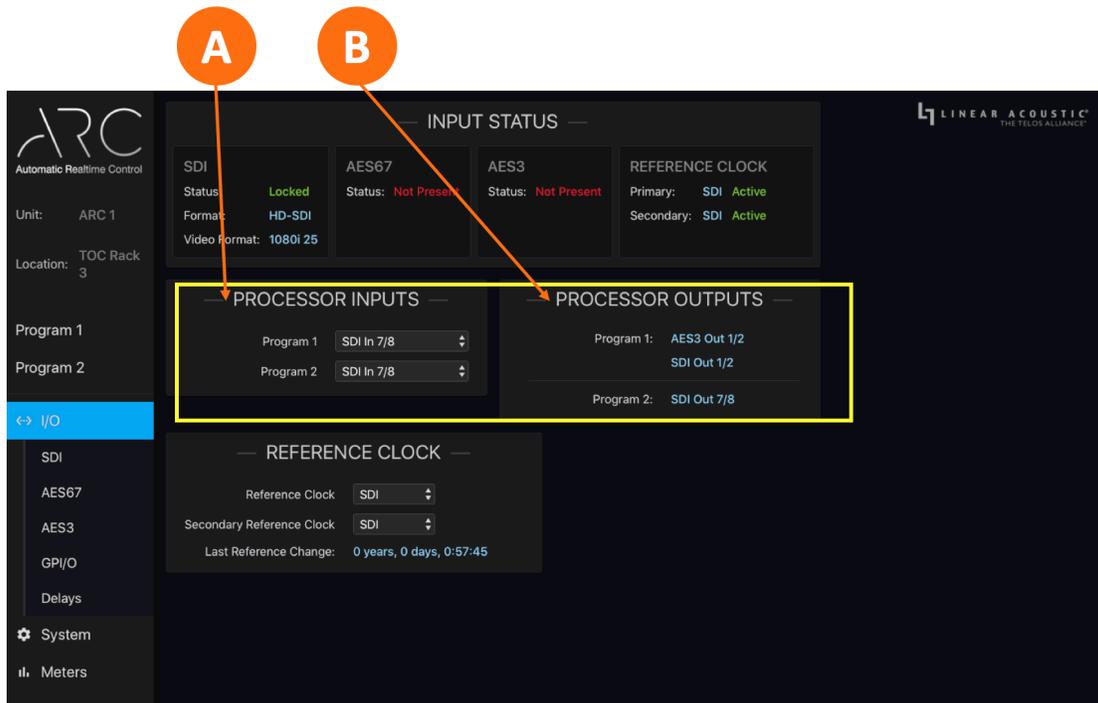
Figure 5-8 – Processor inputs and outputs

Reference Clock

The Primary and Secondary Reference Clock sources are set in the **Reference Clock menu** (A).

The elapsed time since the last reference change is indicated here as well.

Important:

At the risk of stating the obvious, proper reference is critical for trouble-free operation. This is true for ARC and every other digital audio device in your plant. Improper clocking is one of the most common causes of clicks, pops, audio dropouts, and other anomalies, and is at the root of a high percentage of support inquiries.

Details about clocking and reference are outlined in Chapter 3, "I/O, Clocking and Reference." A moment spent reading and understanding this information now will save you time and grief down the road.
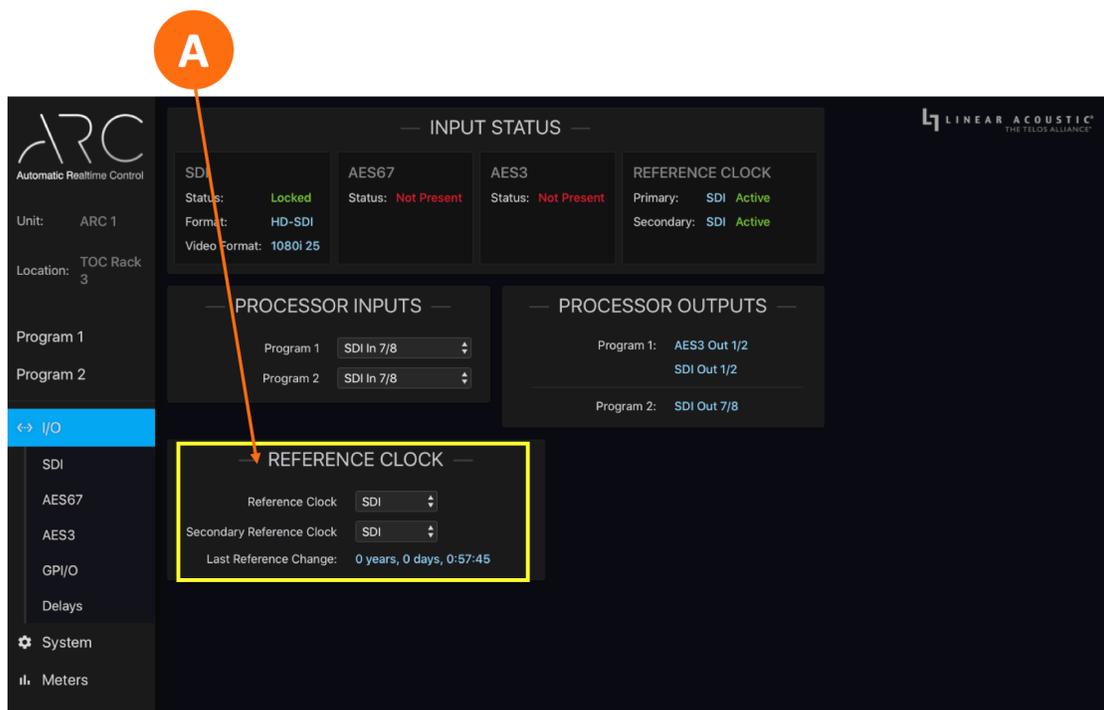
Figure 5-9 – Reference clock

System Menu

The **System Menu** is home to the System status screen. It also contains the menus and controls for configuring the Control and AES67 Ethernet ports.

The main system screen contains a wealth of technical status information about the device hardware and software (A). Some of it relates to the overall health of the unit (such as Power Supply status, unit uptime) but most will only be required during the course of troubleshooting an issue with our support team or performing software updates.

The **Unit Name and Location** (B) are entered on this screen. These are "friendly" fields to be used in any way that makes sense for your particular operation. See Chapter 6, "Detailed Setup and Configuration".

The current **Time and Date** (C) can be changed by clicking on the "Edit" button.

Clicking on the **Master Unit Bypass button** (D) engages a hard relay bypass that removes all internal circuits and processes from the signal path and connects each input directly to its corresponding output. This button will turn red when ARC is in bypass mode. Note that this same hard relay bypass automatically engages when the unit is powered down.

Clicking on the **Locate Unit button** (E) will cause the front panel of this particular ARC to change color, making it easy to identify it amongst the dozens of them we hope will populate your racks. A second click will stop the light show. Note that enabling Locate Mode will not affect the audio.

It should come as no surprise that clicking on the **Reboot Unit button** (H) – wait for it – reboots ARC and engages the hard relay bypass for the duration of the reboot process.

> **Important!**
>
> **During the reboot process – which typically takes less the one minute – no loudness control will take place. Depending upon your I/O configuration, a complete loss of audio may occur.**

Clicking on the **Reset Unit to Factory Defaults** button (J) resets all parameters to the factory default state with the exception of IP settings, which will be retained so that you can remotely connect to ARC even after performing a reset.

If you wish to reset IP settings as well, enable the **Include IP Settings in Reset** button (I) prior to the reset.

> **Important!**
>
> **Performing a factory reset will erase all APTO user profiles, I/O routing, and loudness logs. This information cannot be reclaimed once you reset ARC, even by contacting customer support or conjuring it up through supernatural ceremonies involving bones and a Ouija board.**

Software Updates

ARC has two software banks. Both can be populated with different software versions, but only one version can be active at any given time. The **Active bank** (G) is indicated in red. Instructions for uploading and selecting different software versions is provided along with the software when you either download it from the customer portal or website or it is provided by our customer support team.
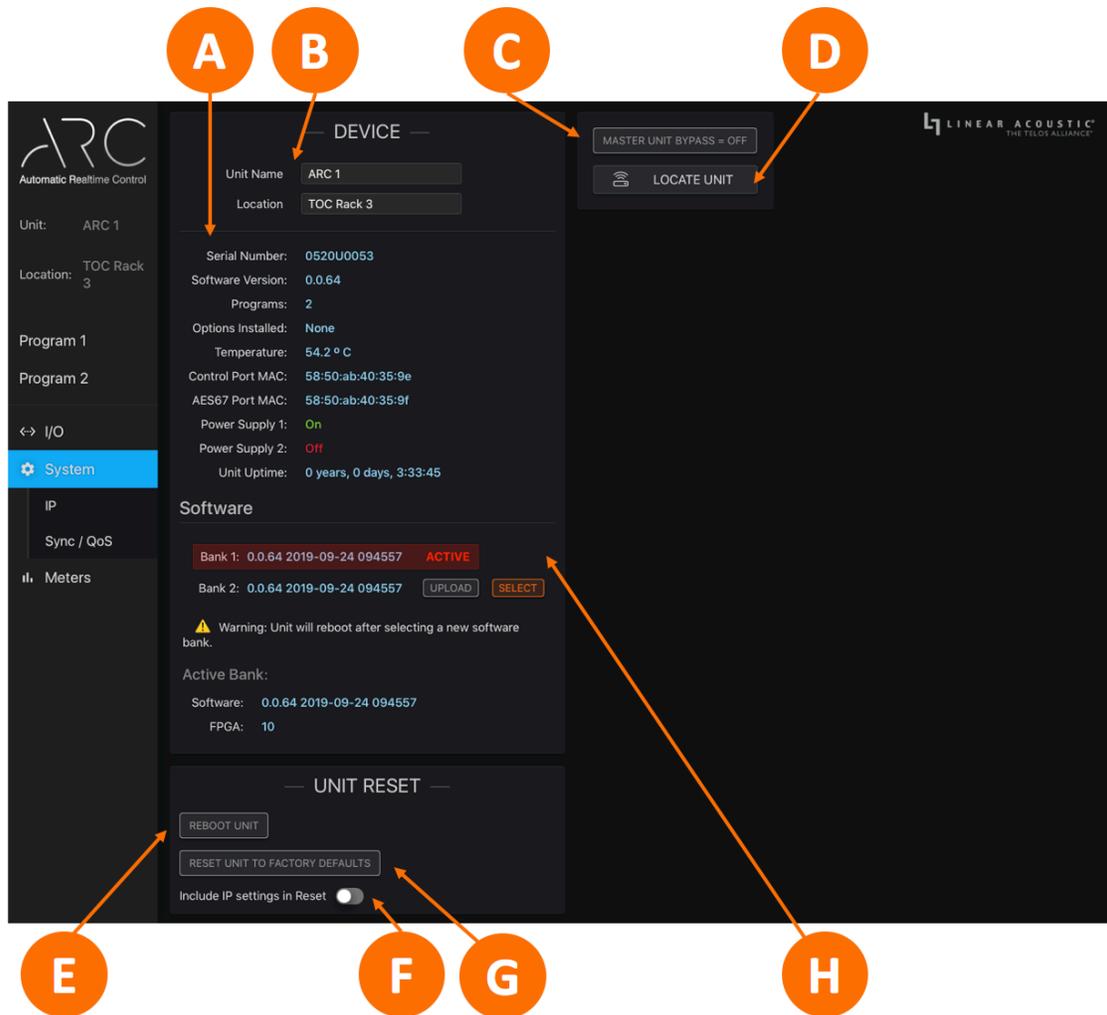
Figure 5-10 – System overview

IP Menu

The **IP menu** provides control over DHCP, shows network activity, and includes fields for entering static IP addresses, subnets, and gateways for both the Control and AES67 Ethernet ports.

Setting IP Addresses

By default, ARC is set up with DHCP enabled and will automatically receive an IP address when its Control Ethernet port is connected to a network with a DHCP server.

Important!

Both the Control and AES67 connections *must* be connected to 1000BASE-T (Gigabit) switch ports. If a port will not be used, *it should not* be connected to a switch but *does* require a specific configuration described below. If both the Control and AES67 ports are used, *each must be on a different subnet*.

To use a fixed IP address, first click the **Edit button** (A) to unlock the settings for each Ethernet port. Click on the **Use DHCP button** (B) to disable DHCP. The text of the **NET Address**, **Net Mask**, and **Gateway fields** (D), which is grayed out when DHCP is enabled, will turn white and become editable. Once the information is entered, click on the **Apply button** (or the **Cancel** button to back out without making any changes). Verify that you have a good network connection by looking for a green dot and the word **Active** in the status window.

Setup is identical for both the Control and AES67 ports.

If either of the two Ethernet ports will not be used, do not connect the unused port to a network switch, and set "Use DHCP" to "Yes" for the unused port.
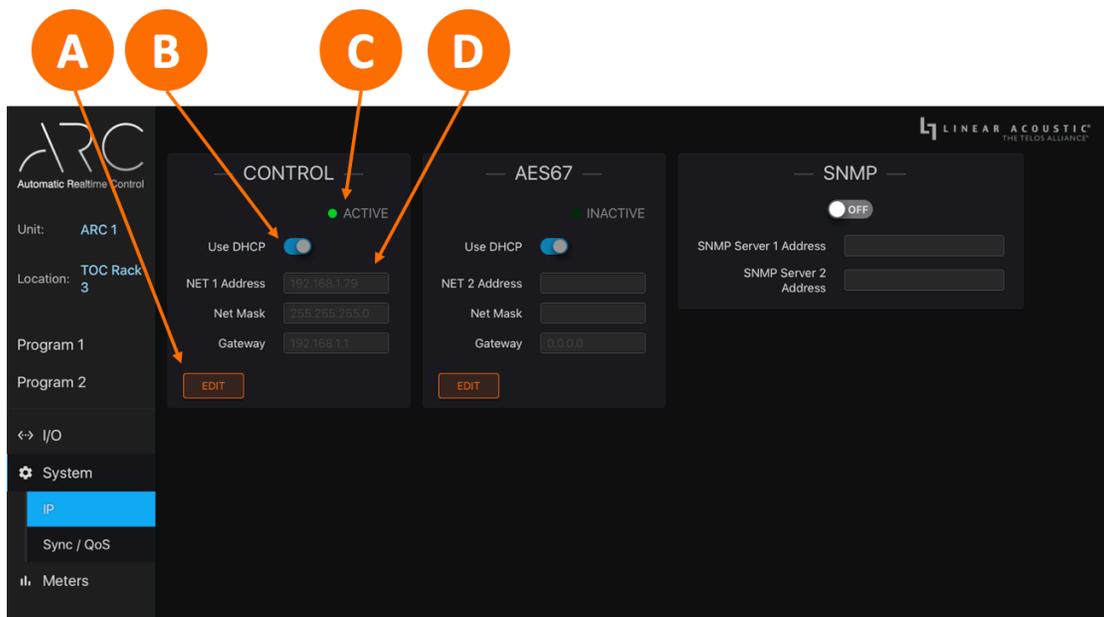
Figure 5-11 – IP menu

Sync, QoS, and PTP Menu

As explained in Chapter 3, whenever ARC is set up to use AES67 audio either on the input, the output, or both, it **must** be slaved to an external PTP clock for reference.

Important!

At this time, ARC *must* be slaved to an external PTP clock if AES67 streams are used for audio input, output, or both. Additional PTP-related settings are grayed out and reserved for future use.

The *Slave Only button* (A) must be enabled and the Domain Number must be set for the correct PTP domain. Valid values are 0 – 127. The AES67 PTP profile uses 0; the SMPTE PTP profile uses 127.
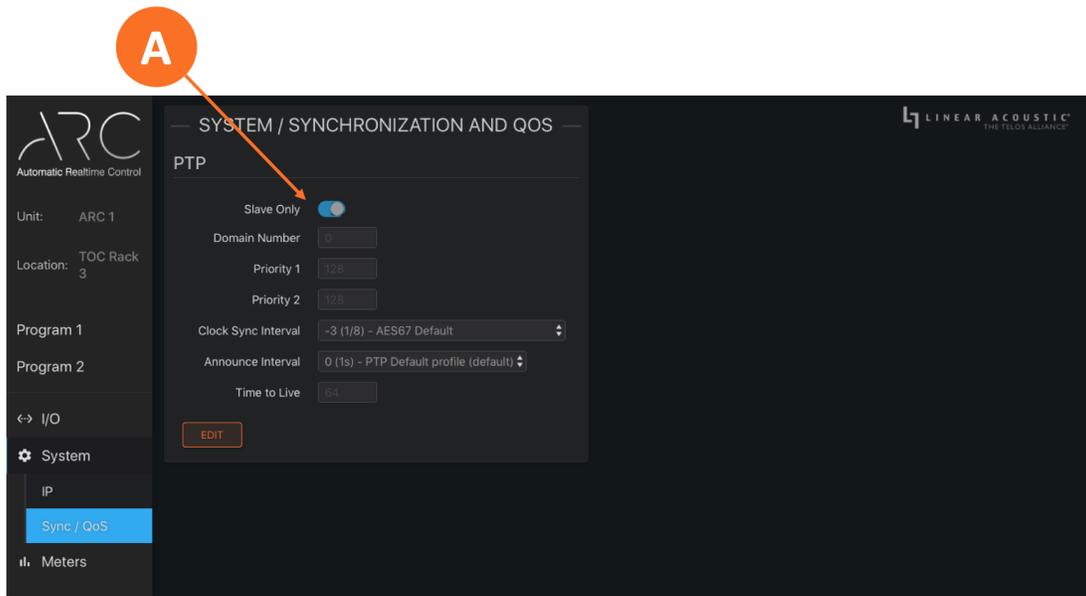
Figure 5-12 – PTP menu

6 Detailed Setup and Configuration

Name and Location

Once logged into ARC, click on the **System menu** (A). Click in the **Unit Name field** (B) and type in a "friendly" name for this particular unit. You may also enter the **physical location** of the unit in the "Location" field (C).
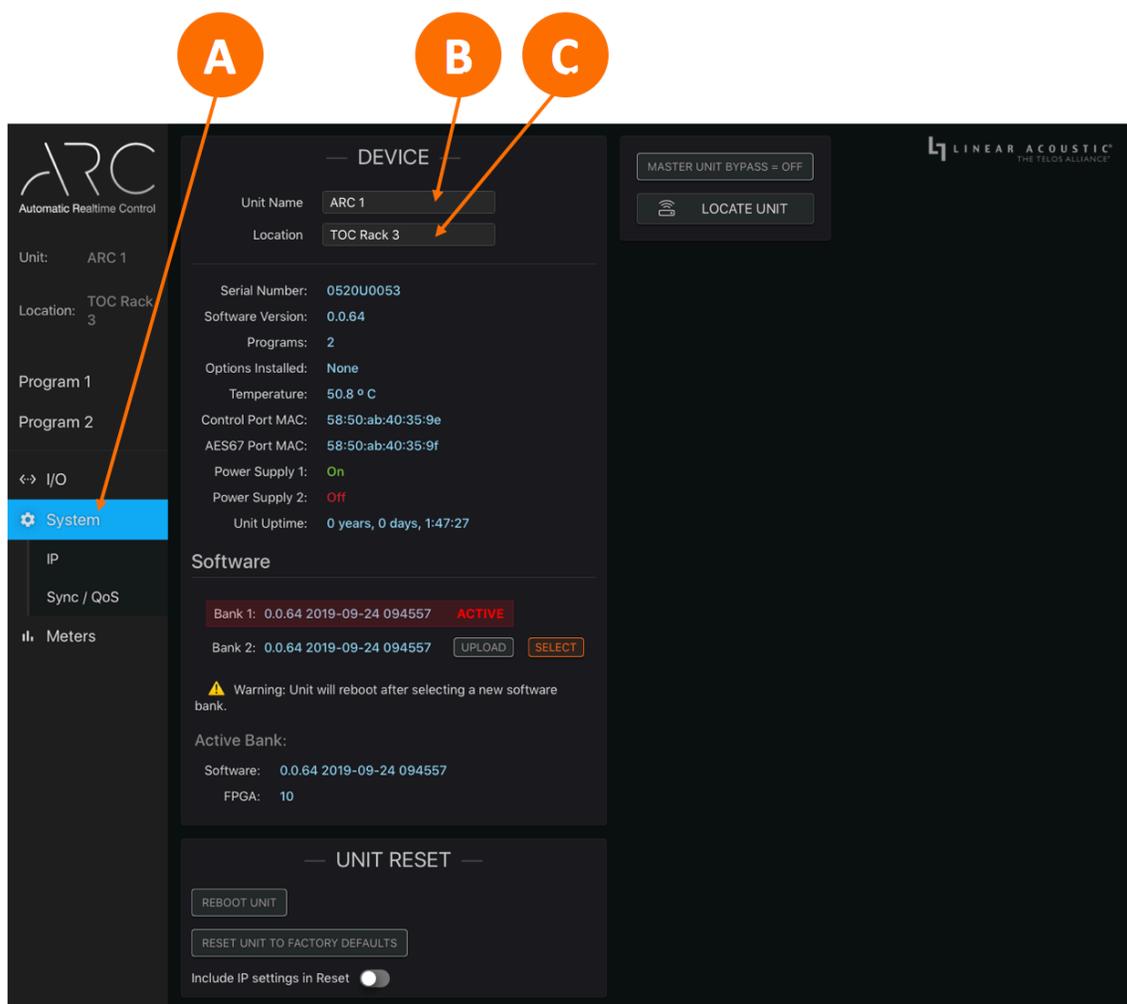
Figure 6-1 – System menu

I/O Setup

Connect your SDI, AES-3, and/or AES67 sources to the rear panel of the ARC.

Input Routing and Reference

Click on the **I/O menu** (A), then choose the audio sources you wish to process (Program 1 and Program 2) from the list of available options via the dropdown menus in the **Processor Inputs section** (B).

Set the reference clock for each source using the dropdown menus in the **Reference Clock section** (C). The destinations for the processed audio are configured in a different menu but shown for convenience in the **Processor Outputs section** (D).

Important:

At the risk of stating the obvious, proper reference is critical for trouble-free operation. This is true for ARC and every other sync'd device in your plant. Improper clocking is one of the most common causes of clicks, pops, audio dropouts, and other anomalies, and is at the root of a high percentage of support inquiries.

Details about clocking and reference are outlined in Chapter 3, "I/O, Clocking and Reference." A moment spent reading and understanding this information now will save you time and grief down the road.
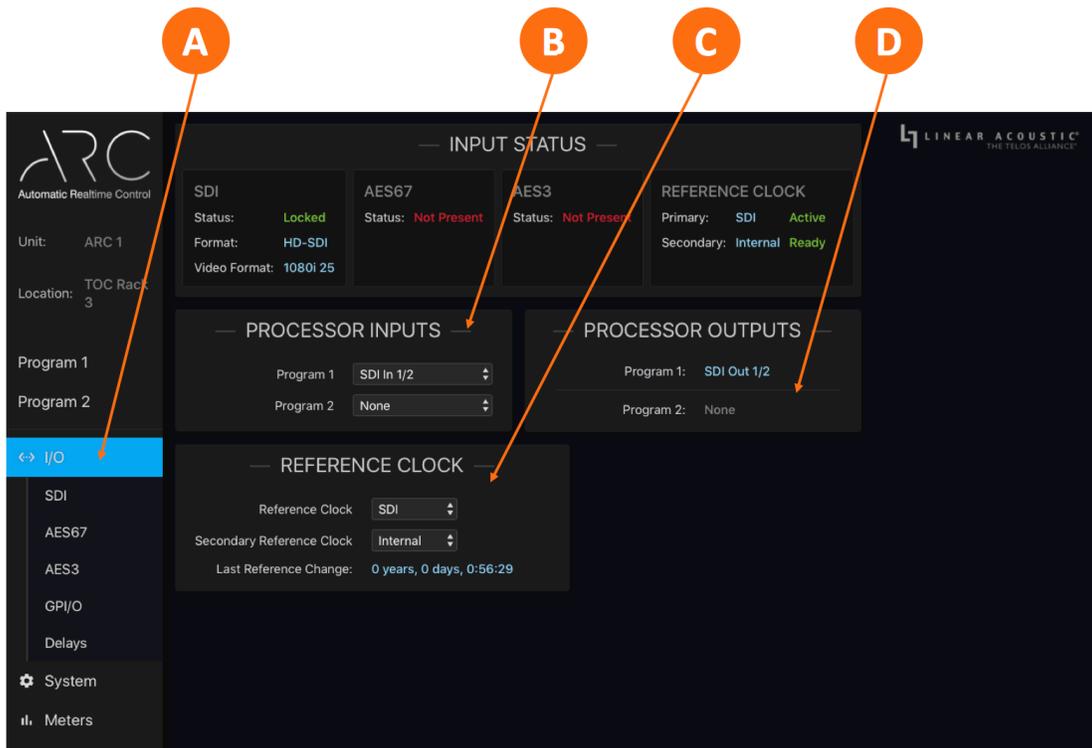
Figure 6-2 – I/O menu

Output Routing

Each output type - SDI, AES67, and AES-3 – has its own sub-menus.

SDI Outputs
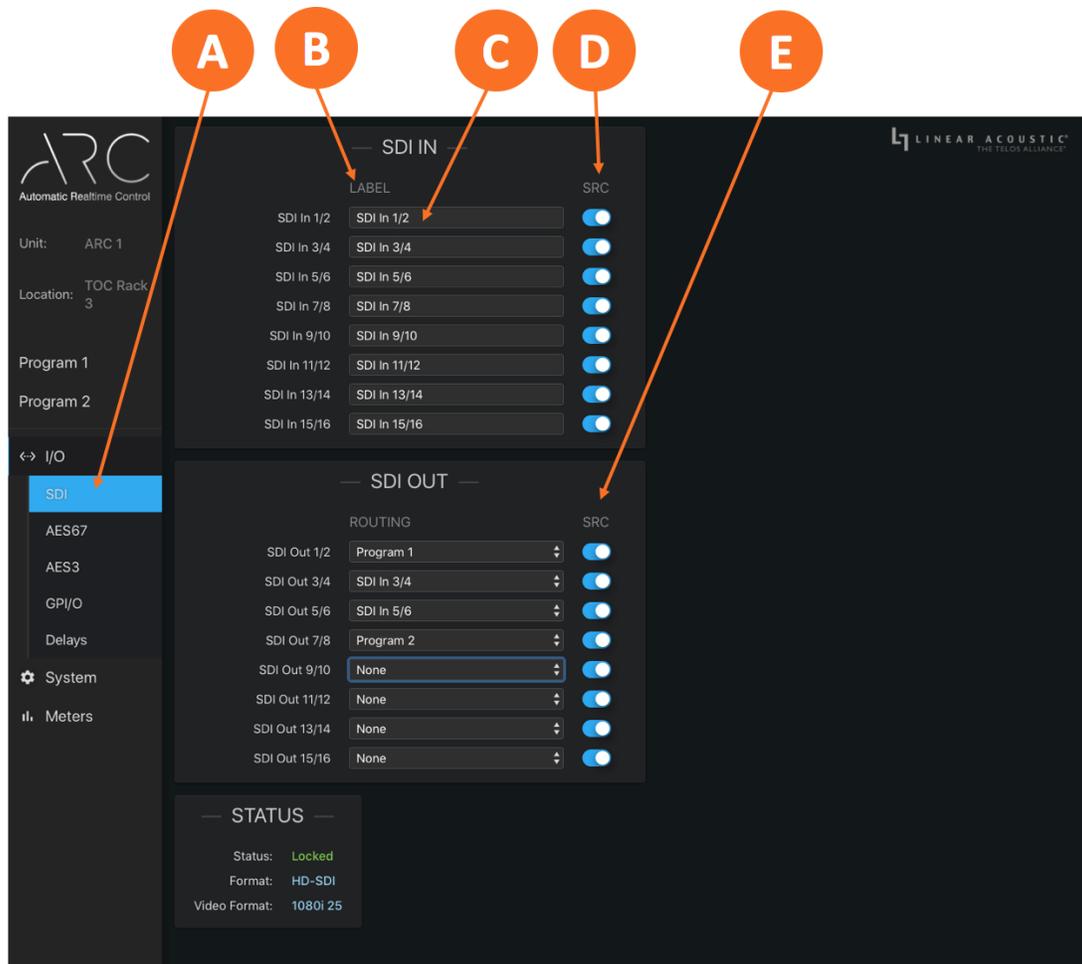
Figure 6-3 – SDI I/O menu

To configure the SDI outputs, click on the **SDI sub-menu** in the "I/O" menu (A).

In the "SDI Out" routing section, use the **SDI Out dropdown menus** (C) to select which audio sources get routed to each SDI output pair. These may include processed audio as well as available SDI, AES67, and AES-3 inputs.

♦ By default, the corresponding SDI input pairs are routed to each SDI output, which effectively passes unprocessed input audio directly to the output

♦ Pair shuffling can be accomplished by selecting a different SDI input for any given SDI output

♦ To route processed (loudness-controlled) audio to an SDI output pair, choose "Program 1" or "Program 2"

The SDI SRCs be disabled when using an SDI input and output using the **SDI SRC In switches** (D) and **SDI SRC Out switches** (E). Doing so does lower throughput latency slightly, but there is no disadvantage to leaving them enabled. In fact, if there are any anticipated reference clock changes (either intentional or unplanned) it is safer to leave them enabled.

Note that you can customize the names of the various SDI input pairs to make identifying the sources easier. Click inside the **Label field** (B) of the "SDI In" section, type a name, then click the green checkmark to save your changes.

AES67 Outputs
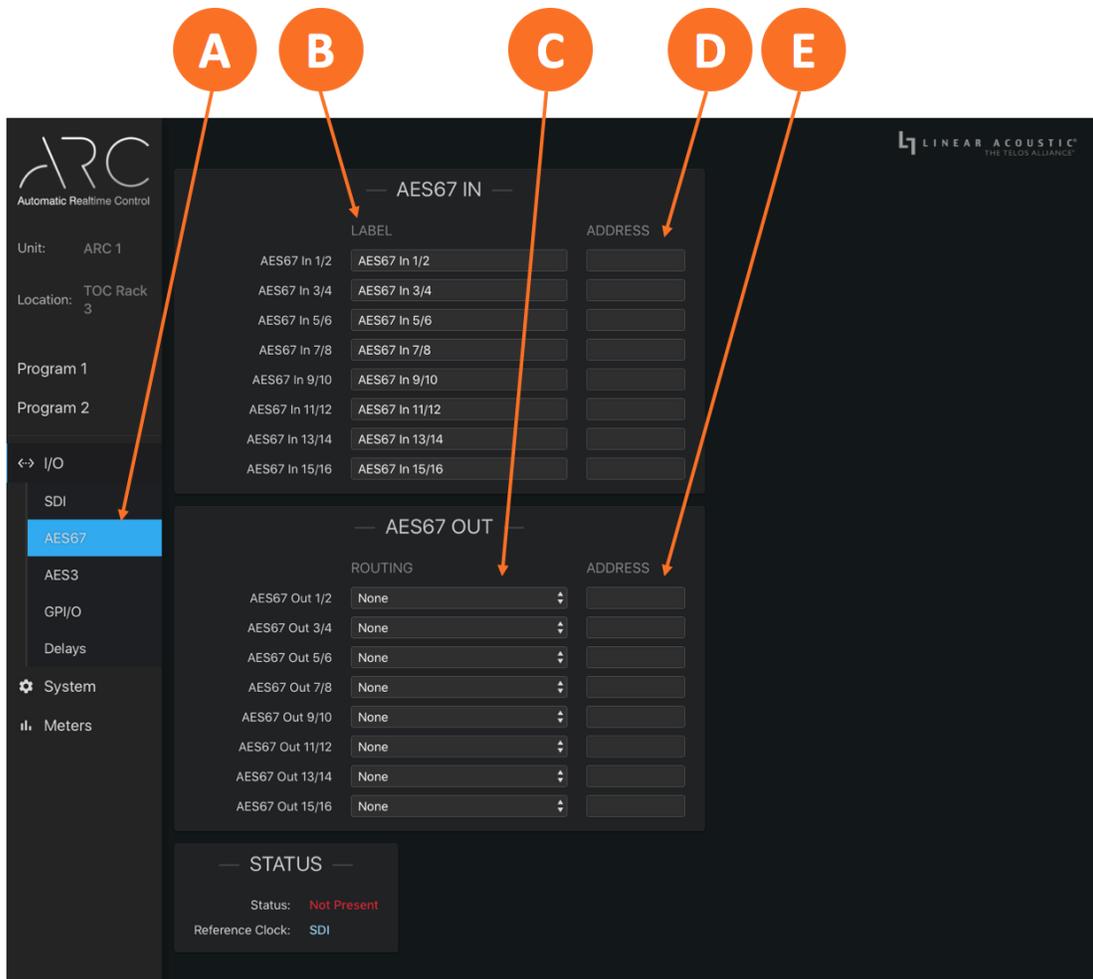
Figure 6-4 – AES67 I/O menu

AES67 outputs are configured in the AES67 sub-menu and in much the same was as SDI outputs.

Click on the **AES67 sub-menu** (A) of the "I/O" menu. Use the **AES67 dropdown menu** (C) to select which audio gets routed to each SDI output pair.

Sources may include processed audio as well as available SDI, AES67, and AES-3 inputs.

The multicast address of each desired AES67 input should be entered into the "Address" field of the "AES67 In" section (B).

As with other inputs, you can customize the names of the various AES67 input pairs to make identifying the sources easier. Click inside the **Label field** (B) of the "AES67 In" section, type a name, then click the green checkmark to save your changes.

AES-3 Output
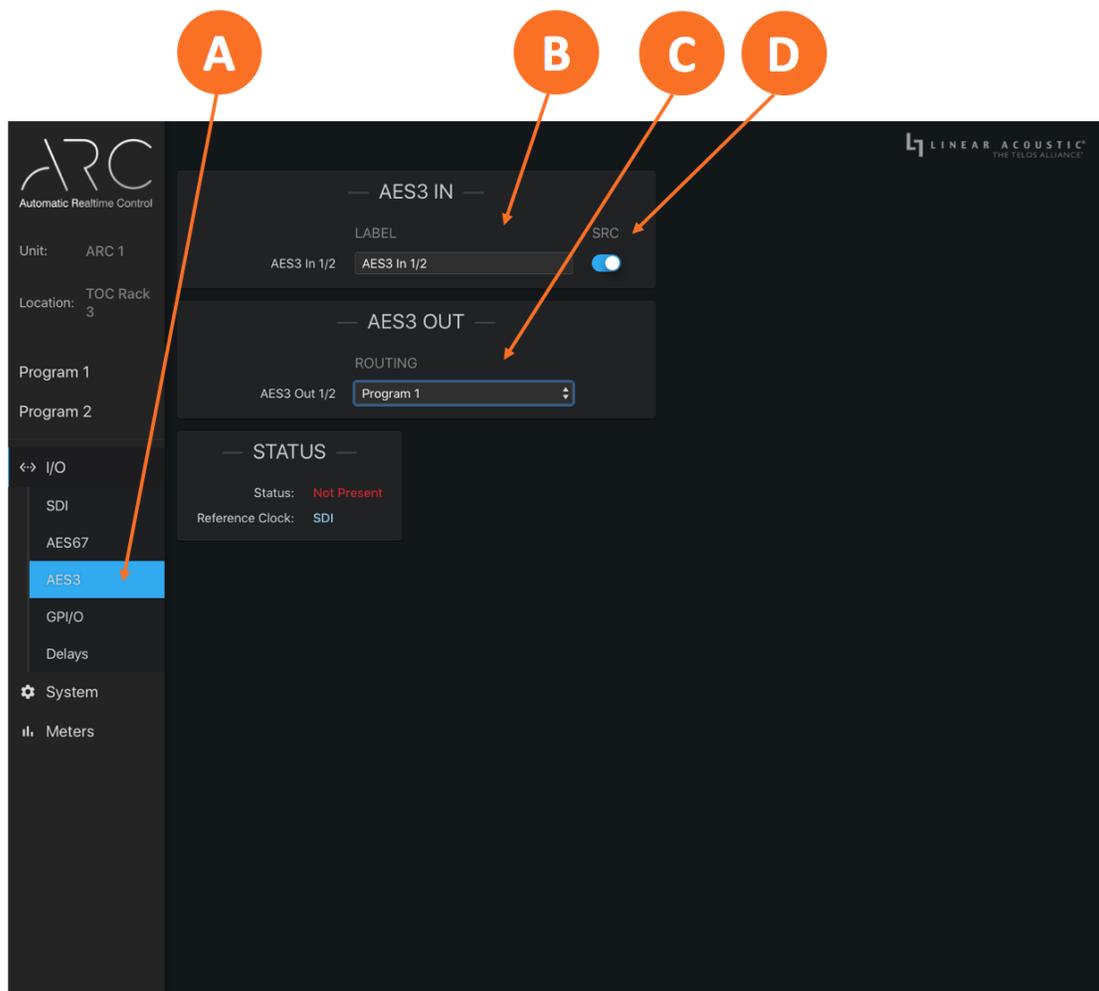
Figure 6-5 – AES-3 I/O menu

The AES-3 output is configured in the AES-3 sub-menu and in much the same way as SDI and AES67 outputs.

Click on the **AES-3 sub-menu** of the "I/O" menu (A). Use the **dropdown menu** (C) of the "AES-3 Out" routing section to select which audio gets routed to the AES-3 output. Sources may include processed audio as well as available SDI, AES67, and AES-3 inputs.

As with the other outputs, you can customize the name of the AES-3 input pair to make identifying the sources easier. Click inside the **Label field** (B) of the "AES-3 In" section, type a name, then click the green checkmark to save your changes.

The **Input SRC** (D) is enabled by default. If the AES-3 input is your clock reference, disabling the SRC will reduce latency slightly. If there are any anticipated reference clock changes (either intentional or unplanned) it is safer to leave it enabled.

GPI/O

ARC features five GPI and five GPO functions through its rear panel DB-15 connector, activated by a momentary contact closure.

GPI Functions (A) include:

- None
- Reboot/Reset Unit
- Hardware (Relay) Bypass
- Reset All Input Loudness Meters
- Reset All Output Loudness Meters
- Reset All Loudness Meters
- Recall Profile (PGM 1)

- Reset Program 1 Input Loudness Meter
- Reset Program 1 Output Loudness Meter
- Bypass Program 1 Processing
- Recall Profile (PGM 2)
- Reset Program 2 Input Loudness Meter
- Reset Program 2 Output Loudness Meter
- Bypass Program 2 Processing

GPO Functions (D) include:

- GPI Passthrough
- Power Supply 1 Up
- Power Supply 2 Up
- Power Supply 1 Down
- Power Supply 2 Down
- Unit Resetting
- Unit Rebooting
- Unit Powering Down
- Unit Above Temperature
- Unit Bypass (Relay Bypass) Active
- Change in Reference
- Primary Reference Clock is Lost

- Secondary Reference Clock is Lost
- Primary Reference Clock is Active
- Secondary Reference Clock is Active
- Internal Reference is Active
- Program 1 Loudness Above Target Range
- Program 1 Loudness Below Target Range
- Program 1 APTO Processor Bypassed
- Program 1 Output Loudness Meter Reset
- Program 2 Loudness Above Target Range
- Program 2 Loudness Below Target Range
- Program 2 APTO Processor Bypassed
- Program 2 Loudness Meter Reset
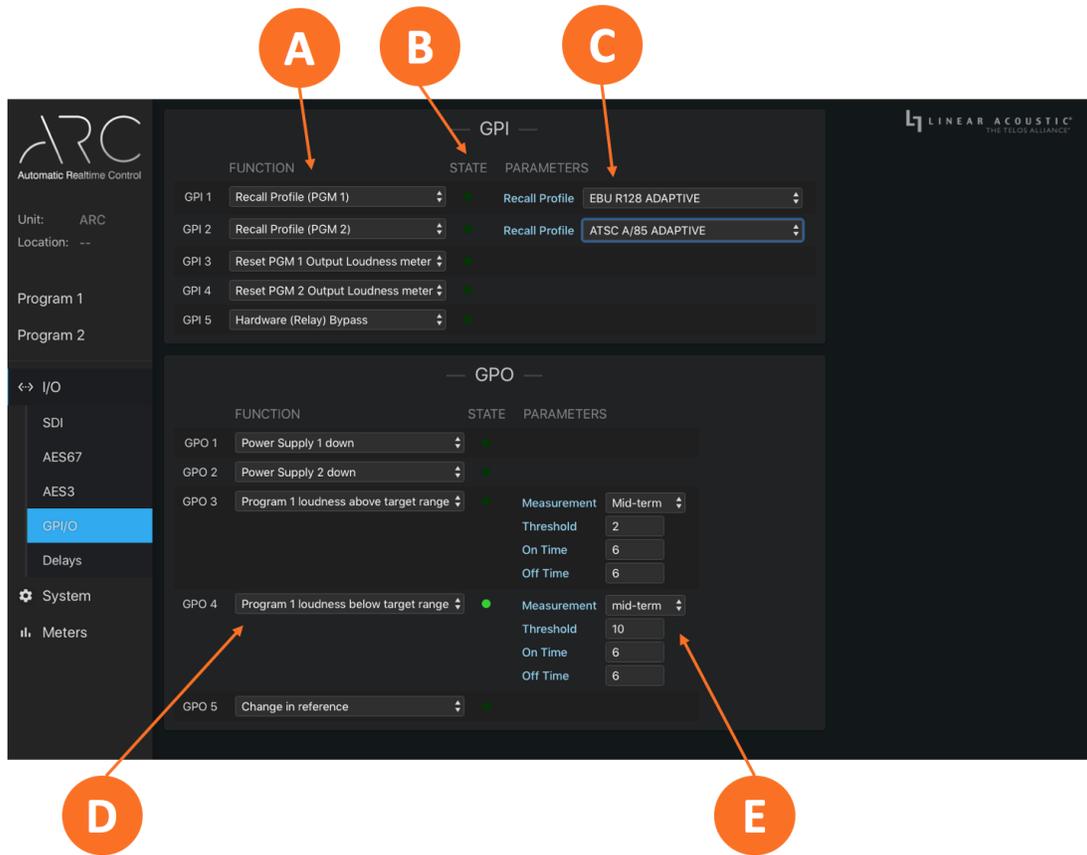
Figure 6-6 – GPI/O menu

The **State Indicator** (B) for each function will light for the duration of the closure on GPIs. For GPOs, it will light for the duration of an active event.

Certain GPIs have selectable **Parameters** (C) associated with them. For example, selecting a "Recall Profile" function will present a dropdown menu to select the desired profile.

Likewise, some GPOs have user-defined **Parameters (E)** as well. Choosing one of the "Loudness Above Target Range" or "Loudness Below Target Range" functions will display the following parameters:

♦ **Measurement** determines the integration time used to measure the over/under loudness status

♦ **Threshold** determines how far the loudness readings must deviate from the target in dB

♦ **On Time** is the length of time (in seconds) that loudness levels must stay above/below the threshold in order to trigger the GPO

♦ **Off Time** is the length of time (in seconds) that loudness levels must return and remain within tolerance before the GPO is cleared

Delays, Latency and A/V Sync

Latency through ARC varies depending on which inputs and outputs are used and whether or not SRCs are enabled. In any case, any latency will result in the processed audio trailing the video by that amount at the SDI output.
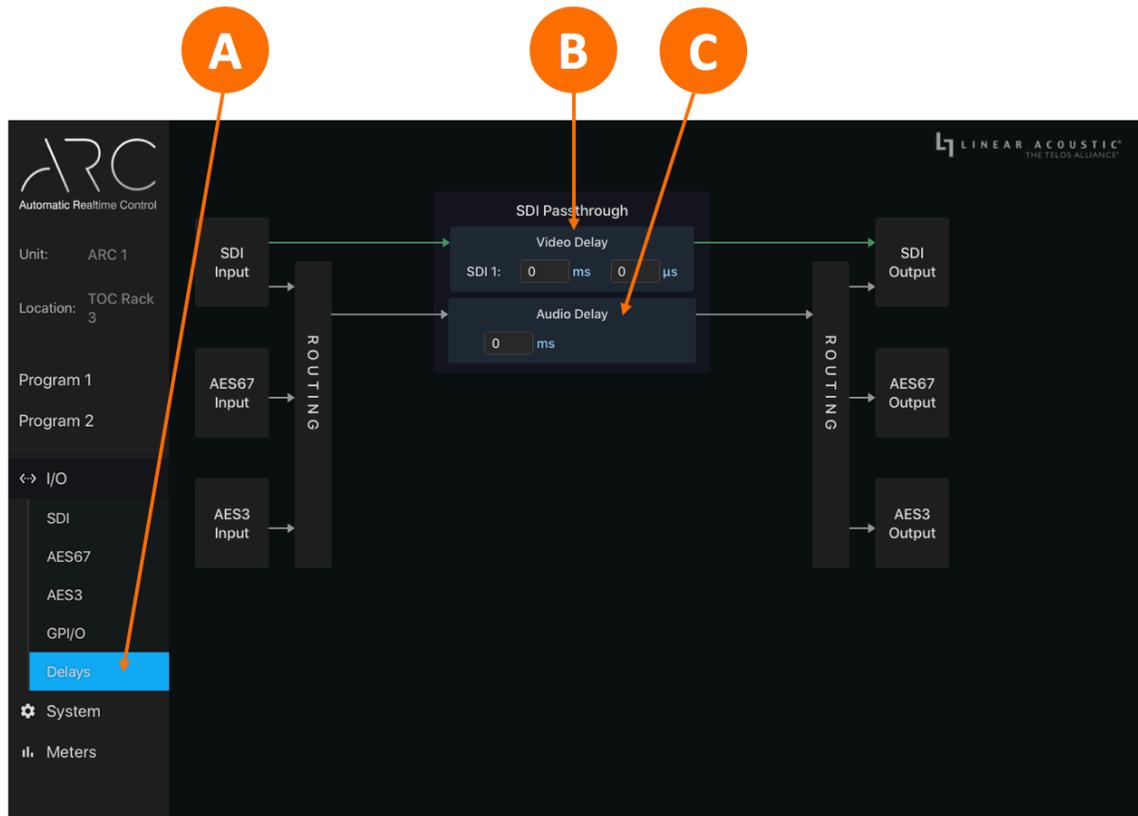
Figure 6-7 – Delay menu

To compensate for that latency and avoid lip sync issues, an SDI video delay is provided in the **Delays sub-menu** of the I/O menu (A). Click on the millisecond and/or microsecond fields in the **Video Delay section** (B) and type in the required value. If the output needs to be delayed by an even frame (the value of which will vary depending on frame rate), simply enter that number.

> **Note:**
>
> Entering an even frame value will result in a situation where it is necessary to add additional audio delay to the AES67 and AES-3 outputs so that they match timing of the SDI output signal. This can be accomplished by setting the **Audio Delay** (C) to a value equal to the difference between a full frame and the latency of the unit.

Latency values for Firmware V0.0.64 are as follows:

- ◆ AES-3 input to AES-3 output with SRCs disabled: 30.5ms

- ◆ AES-3 input to AES-3 output with AES-3 input SRC enabled: 33.5ms

- ◆ SDI input to SDI output with SRCs disabled: 33.5ms

- ◆ SDI input to SDI output with SDI SRCs enabled: 36.5ms

- ◆ AES67 input to AES67 output: 107ms (measurement made using Telos Alliance SDI xNode which has a latency value of 80ms)

7 An Introduction to APTO™

But First, a Word on Television Loudness

The introduction of compliance regulations and an overall increased awareness of television loudness control have generally been successful in taming the wild loudness shifts that startled and annoyed viewers in the earlier days of digital television.

Real-time processing is ubiquitous and effective, content creators now have a better understanding of how viewers watch (and listen to) their programs, and broadcasters have the necessary tools at their disposal to analyze and, if necessary, correct pre-recorded programs upon ingest in the file domain.

That being said, there are still some misconceptions about loudness control and regulations being repeated as fact, and when applied, are nearly always to the detriment of the audio.

Speaking in the most general terms, most loudness regulations specify an **average** audio level for **each program segment**. Some are based on the loudness of the entire mix, while others are based on only an anchor element such as dialogue.

In either case, the regulations do **not** dictate that the audio be devoid of dynamic range nor do they mandate that levels constantly stay at the target value so that your LKFS/LUFS meter looks more like an FM modulation monitor at a major market radio station. Soft passages are allowed to stay soft and loud passages are permitted to pass through generally unaltered, so long as the average over the duration of the program segment is maintained and, where applicable, True Peak values are not exceeded.

Just like audio can be over-processed, it can also be under-processed. Trying to watch a sporting event in a noisy bar or restaurant (or even the average living room) would be frustrating if presented with the amount of dynamic range appropriate for a cinematic presentation in a high-end, multi-channel home theater.

The proper balance is found in what is commonly called the "comfort zone," that is, the range in which the viewer never strains to hear lower level programming yet is never annoyed by excessively loud levels. This is sometimes described as the situation in which the viewer never feels the need to reach for the volume control on their television, mobile phone, or tablet.

What Is APTO?

The audio processing within ARC is performed by Linear Acoustic® APTO™, our latest and most advanced adaptive loudness control algorithm to date.

APTO ensures that ARC can deliver audio that is compliant with various television broadcast regulatory loudness standards including EBU R128, ATSC A/85, FreeTV OP59, ARIB TR-B32, and AGCOM 219/09/CSP as well as the AES-TD1006 streaming standard.

It also includes profiles for other non-broadcast delivery platforms and listening environments such as gaming, movies, earphones, and in-flight entertainment.

APTO enhances the listening experience by providing consistent audio levels within a user-defined "comfort zone", thus eliminating listener annoyance due to sudden loudness shifts. It also improves dialogue intelligibility, addressing one of the most common complaints with television audio. Best of all, it does so without audibly affecting the sound quality and artistic intent of the original content.

What Makes APTO Different?

Traditional real-time television processing normally employs a series of wideband and/or multiband compressors or AGCs that react to changing input levels by either increasing or decreasing gain in an effort to provide a more consistent output level. The various threshold, ratio, and attack and release rates can be adjusted to help determine the amount of dynamic range present at the output, or, put another way, how close to the desired output level the audio remains at any given time. A final look-ahead limiter is typically employed for peak control.

One potential downside to this type of processing is that unless the audio falls below a specified gate threshold – the point at which low level audio isn't increased so as not to bring up background noise – the gain is always changing, whether it needs to or not. Furthermore, multiband processing by design re-balances the spectral mix of the program. This can be advantageous for helping to fix less-than-ideal mixes or if a spectrally consistent output is a priority, but it can also affect the artistic intent of well-mixed content.

In contrast, APTO focuses on achieving the following goals:

♦ Ensuring that foreground sounds – particularly dialog – remain intelligible at all times

♦ Allowing the user to define a comfort zone within which no additional audio processing is applied, providing a more natural sense of dynamics

♦ Maintaining the spectral balance of the original program material to preserve artistic intent

♦ Achieving and maintaining an output target level that is in compliance with global loudness regulations and is optimized for distribution platforms (such as streaming and on-demand services) and for specific devices and listening environments (including mobile phones and tablets)

How Does APTO Work?

In very basic terms, APTO first measures and analyzes the loudness of the incoming audio. In its first processing stage – the "Dynamic Range" stage - it applies realtime loudness control to reduce the overall dynamic range and get the levels within the user-defined comfort zone. This processed audio is then scaled in the second processing stage – the "Compliance" stage - to achieve an average output level that matches the desired target loudness value.

As mentioned previously, ARC includes a host of factory profiles for various loudness standards and deliverable platforms, but individual controls to fine tune both processing stages are also brought out to the user interface and are described in greater detail later in Chapters 8 and 9.

8 Basic Processing and Loudness Control

Choosing an APTO Profile

In most cases, ARC is truly a "set and forget" processor and simply choosing a factory profile is all that is required.

To select a profile, navigate to the **Program 1** (or Program 2) **menu** (A), click the **Load button** in the APTO Loudness Control section (B), mouse over **Factory Presets** (C), then choose an appropriate profile from the **Factory Preset List** (D).

Figure 8-1 – APTO factory profiles

Basic APTO Controls

The individual controls within each APTO factory profile have been adjusted in such a way that they automatically deliver the required result associated with the profile name (such as "EBU R128 Adaptive"). However, there may be situations in which you want to adjust certain individual parameters to customize or fine tune the audio.

The APTO Loudness Control section of the Program 1 and Program 2 menu contains the basic controls for each program path and are identical in terms of controls and functionality, though completely independent of one another in their operation.
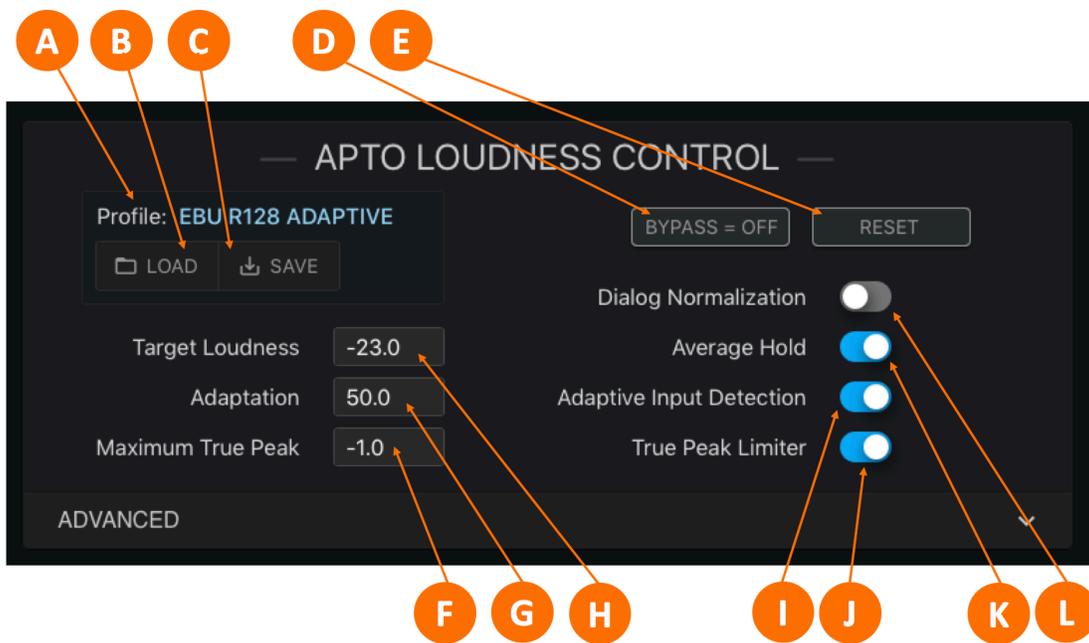
Figure 8-2 – Basic APTO controls

Current Profile

The currently selected profile is displayed in the **Profile window** (A).

> **Note:**
>
> **In the current version of ARC, there is no indication that the factory profile has been modified as a result of changing the values or settings of individual controls.** This feature will be included in an upcoming software upgrade. In the meantime, we strongly recommend that you save any changes and create ~~uniquely-named~~ user profiles as you make adjustments as outlined below in the paragraph on using the "Load and Save" buttons.

Load and Save

The **Load button** (B) is used to recall factory profiles, user profiles, or to import a profile previously saved to your computer. Any changes or modifications to the current profile can be saved by using the cryptically-named **Save button** (C).

Bypass

APTO can be bypassed so that the input audio is passed through to the output without being processed by clicking on the **Bypass control** (D). This is useful for a quick comparison between the unprocessed and processed audio. When bypass is engaged, the button will turn red.

Reset

Clicking on the **Reset button** (E) resets the loudness measurements as well as the gain buffers APTO uses in the normalization stage. Resetting at the start of each individual program element provides accurate per-segment loudness measurements, aids in achieving overall compliance, and ensures that adaptive processing decisions are made based upon the current program dynamics. A GPIO input may be used to trigger the reset automatically.

Target Loudness

The **Target Loudness control** (H) sets the desired average loudness level of the output signal in either LUFS or LKFS, depending upon the profile. Some profiles, such as EBU R128, will measure loudness according to overall program levels while others, such as ATSC A/85, will do so based on dialog and gated speech measurements.

To change the target loudness value, click in the Target Loudness field, type in the desired value, and click on the green check mark to save your change (or the red "X" to exit without saving).

Adaptation

The **Adaptation control** (G) determines how much processing is applied to the incoming signal in the Dynamic Range processing stage, and, in combination with various individual controls, determines dynamic range of the output audio.

The ideal amount of Adaptation depends upon both the source content and on the destination platform. Content that has been pre-analyzed for loudness, scaled, and normalized in the file domain will require less realtime processing than, say, live sports, which can have rather unpredictable audio levels. Programming streamed to a mobile device or expected to be heard on lower quality earbuds will benefit from more Adaptation than the same content destined for a home cinema presentation.

Ideally, you want enough Adaptation to achieve compliance and keep levels within the viewer's comfort zone and at a stable average level, but not so much that the audio sounds unnatural or over-processed. APTO is based on a psychoacoustic model that takes into account human hearing and perceived loudness and remains very natural-sounding even when extensive processing is applied, but it is generally advisable to keep the amount of Adaptation under 50% when possible. If incoming content is so poorly controlled as to require higher values, it may be necessary to adjust some of the individual controls and create a custom profile to address this scenario.

Maximum True Peak Limiter and Limiter Threshold

The **True Peak Limiter control** (J) enables and disables the True Peak Limiter, which is the final processing stage just ahead of the final output. The **Maximum True Peak value** (F) sets the level beyond which the True Peak limiter engages and attenuates the processed audio so as not to exceed the set level. These controls comply with the True Peak measurement as outlined in ITU-R BS.177-4 Annex 2.

Dialog Normalization

The **Dialog Normalization control** (L) enables dialog detection and measurement. When enabled, APTO's Compliance processing stage uses long-term speech-only measurements rather than the overall input program loudness to ensure the output target level is achieved.

The use of dialog-based measurement is normally determined by regional regulations. For example, ATSC A/85 relies upon anchor-based normalization, specifically dialog, while EBU R128 recommends the use of overall measurements. This is reflected in their respective factory profiles.

Average Hold

When the **Average Hold control** (K) is enabled and the output audio of the Dynamic Range processing stage falls below the level set in the **Average Hold Threshold control** (located in the Advanced menu), APTO's Compliance stage of processing becomes inactive until the level once again rises above the threshold. This helps prevent noise and low-level background audio from being increased unnecessarily.

Adaptive Input Detection

When enabled, the **Adaptive Input Detection control** (I) dynamically adapts the amount of processing occurring in the Dynamic Range stage of processing depending upon the actual measured average level at the input. The degree to which the actual input levels influence the processing versus relying upon the value of the **Average Input Level control** (found in the Advanced menu) is determined by the **Adaptive Input Percentage control** (also located in the Advanced menu).

When **Adaptive Input Detection** is disabled, Dynamic Range processing decisions are made based strictly upon the value set in the **Average Input Level** and in accordance with the settings of other parameters and controls.

Adaptive Input Detection is especially useful when source audio levels are unknown or are likely to vary widely as it allows the Dynamic Range processing stage to respond more predictively the actual incoming content.

If the incoming content has already been analyzed and loudness-corrected in the file domain, less overall processing is required and a more natural-sounding output can be achieved by setting the **Average Input Level** (located in the Advanced menu) to the same value as the target level used during file-based correction and reducing the value of the **Adaptive Input Percentage** control (also found in the Advanced menu) or disabling **Adaptive Input Detection** altogether.

9 Advanced Processing Controls

Advanced APTO Controls

As we've mentioned before, most users and applications will be well served by simply selecting the appropriate factory profile according either to compliance regulations or the delivery platform.

At most, some adjustments to the Basic controls may be necessary.

Unless there are extenuating circumstances or the need to solve a particular problem with highly difficult programming, adjusting the Advanced parameters described below is rarely necessary. **Making arbitrary adjustments without a firm understanding of how these controls work, relate to the Basic APTO controls, and interact with the complex processing algorithms at work can cause more harm than good and result in audio that is either unpleasing, non-compliant, or both.**

That said, they are available for those who need them.
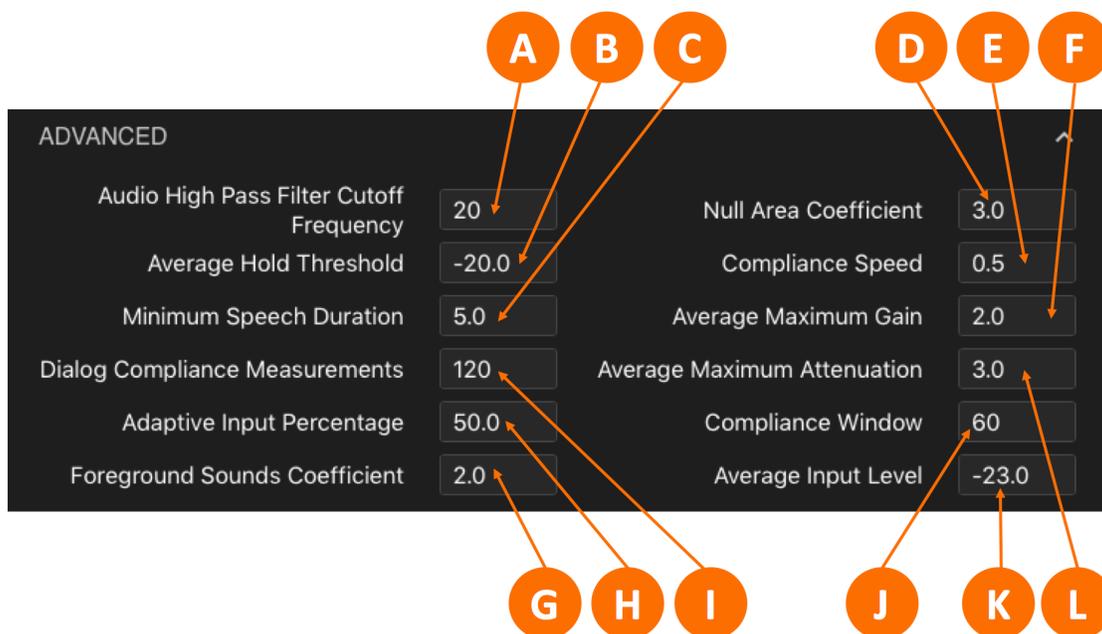
Figure 9-1 – Advanced APTO controls

Audio High Pass Filter Cutoff Frequency

The **High Pass Filter control** (A) sets the cutoff frequency for the high pass filter applied to the input signal. No audio below this frequency will make it through to the APTO processing engine. In most cases, this control should be set to "Off", but in some instances, it may be beneficial to filter out any sub-audible frequencies (below 20Hz).

Voiced-based content with no low frequency information can benefit from a higher setting (60Hz, for example). Delivery platforms with limited headroom (such as in-flight entertainment or mobile streaming) and consumer devices with small speakers (mobile phones, tablets, or earbuds) can also benefit from a higher setting to prevent distortion.

Average Hold Threshold

When the **Average Hold control** (found in the APTO basic menu) is enabled, the **Average Hold Threshold control** (B) determines the point at which the hold is triggered in the Compliance processing stage to prevent increasing noise or background audio.

The value at which the control is set is relative to the **Target level**. For example, if the **Target level** is set to -23dB LUFS and the desired level at which the hold engages in the Compliance processing stage is -34dB, then the **Average Hold Threshold** should be set to -10dB

Minimum Speech Duration

Proper dialog-based loudness processing requires the accurate detection and measurement of speech. When **Dialog Normalization** is enabled, the **Minimum Speech Duration** control (C) determines how many seconds of continuous speech are required for the measurements to be considered reliable and valid, and after which the Compliance processing stage adaptively switches to dialog normalization mode.

A minimum of 5 seconds is recommended. Shorter values may incorrectly factor in non-dialog material, and significantly longer values may cause unnecessary delays in engaging dialog normalization processing, especially in content with shorter segments of continuous dialog.

Dialog Compliance Measurements

The **Dialog Compliance Measurements** control (I) determines how many speech measurements are used to perform dialog normalization in the Compliance stage of processing. Because the algorithm samples the measurements at 0.5 second (one half of one second) intervals, this control should be set to a time value that is twice the desired average. For example, to base normalization on a 60 second average, the control should be set to a value of 120.

> **Note:**
>
> **The Dialog Compliance Measurements setting only applies to profiles that use Dialog Normalization. Profiles that do not use Dialog Normalization rely upon the settings of Compliance Window control to determine their time window.**

Adaptive Input Percentage

When the **Adaptive Input Detection control** (located in the Basic section) is enabled, the Dynamic Range processing section of APTO dynamically adapts the amount of processing depending upon the actual measured average level at the input. The degree to which the actual input levels influence the processing versus relying upon the value of the **Average Input Level control** (K) is determined by the **Adaptive Input Percentage control** (H).

The recommended setting to start is 50% in order to keep the effect of adaptation more consistent.

Foreground Sounds Coefficient

All programming has what is sometimes referred to as an "anchor element," that is, the audio content to which the viewer will pay the most attention. This is typically (though not always) dialog. This is also referred to as "foreground" audio to differentiate it from background audio, or in some cases, noise.

The **Foreground Sounds Coefficient control** (G) sets the level in the Dynamic Range processing stage at which program audio is no longer considered a foreground sound relative to both the lower border as set by the **Null Area Coefficient** control (described in detail below) and a minus infinite level (full silence) on a proportional scale.

For example, if the **Target level** is set to -23dB LUFS, the **Null Area Coefficient** is set to 4, and the **Foreground Sounds Coefficient** is set to 2, any audio between -27 and -25dB will be deemed foreground audio and therefore be raised toward the target.

Audio at levels lower than -27dB will still be increased toward the target, but the degree to which gain is increased slows down considerably. The lower the audio is from -27dB, the less the gain will increase.

Null Area Coefficient

The primary goal of the APTO processing within ARC is to deliver a consistent **average** output level as set by the Target Level control. This does not mean, however, that the actual output audio level must never deviate from this value. In fact, a certain amount of dynamic range helps preserve the artistic integrity of the original programming and makes for a more engaging audio experience for the viewer.

One of the things that makes APTO different from traditional processing is its ability to "do nothing" to audio levels when no action is required to maintain the correct average output level. This avoids the "busy" sound of traditional compressors and ACGs which by their very nature are always operating either over or under a threshold, and therefore always increasing or decreasing gain - often for no good reason.

The **Null Area Coefficient control** (D) sets the lower and upper thresholds that together determine the size of the window in which APTO's Dynamic Range processing stage neither increases nor decreases gain, with the user-determined Target level sitting in the middle of the range. Values are in dB, with larger values resulting in a larger "do nothing" window.

Compliance Speed

How quickly gain changes are made in the Compliance processing stage are largely program dependent. However, the maximum rate at which the gain can increase or decrease is set by the **Compliance Speed control** (E). The rate is calibrated in dB (or LU) per second.

Higher values (from 2 – 6 LU per second) allow compliance to be achieved more quickly but may introduce audible gain changes to the average program level in the process. These faster settings are best suited for situations when input levels are expected to be very inconsistent.

Lower values (from 0.2 to 2 LU per second) provide a subtler and more natural-sounding normalization but may result in gain changes that are too slow and allow the audio to remain outside of the comfort zone for too long. Slower settings work well for content that is more consistent and well-controlled. They are also recommended for long-form content such as feature films or classical music.

Average Maximum Gain

The **Average Maximum Gain control** (F) sets the maximum amount of positive gain (gain increase) applied in the Compliance processing stage in order to reach the Target level.

Larger values will allow very soft program segments to be raised by a greater amount, but whether or not this is desirable must be considered. For instance, some material may have been purposefully kept at a lower level for dramatic effect, and of course there is always the risk of increasing unwanted background noise.

Another unwanted byproduct of setting this control too high - especially with profiles that use a lower **Adaptation value** – is that it can result in APTO taking too long to lower levels when loud content immediately follows soft content, such as when a loud commercial follows a quiet passage from a TV drama, as it will take longer to reduce a greater amount of gain.

For these reasons, it is advisable to set the **Average Maximum Gain** for no more than 2 LU.

One way to enhance the efficiency of the processing across programs is reset APTO at the transition point, which will in turn reset the amount of average gain or attenuation being applied and allow each individual program element to be optimally processed from the beginning.

Average Maximum Attenuation

The **Average Maximum Attenuation control** (L) sets the maximum amount of negative gain (gain reduction) applied in the Compliance processing stage in order to reach the Target level.

Larger values will allow loud program segments to be reduced by a greater amount. However, setting this control too high – again, especially with profiles that use a lower **Adaptation value** - can cause APTO to take too long to raise levels when soft content immediately follows something loud, such as when transitioning from a loud commercial back to a quiet TV drama, as it will take longer to boost a greater amount of gain.

Just as with the **Average Maximum Gain control**, resetting APTO at the start of each program segment can help minimize such issues and optimize overall processing for each individual program segment.

Compliance Window

The **Compliance Window control** (J) adjusts the size of the sliding time window used in the Compliance processing stage to align the output program to the target level. It is similar to the rolling integration time on an LKFS/LUFS loudness meter.

> **Note:**
>
> The Compliance Window setting only applies to profiles that do **not** use Dialog Normalization. Profiles that **do** use Dialog Normalization rely upon the settings of the Dialog Compliance control to determine their time window.

Many factors can influence whether or not content at the output of ARC is compliant. These include the level consistency of the incoming audio, program duration, the difference between the average input level and the Target level, the amount of Adaptation employed, and the settings of many of the controls listed in this chapter.

That said, a smaller compliance window value more easily delivers a compliant output for shorter program durations, but can produce more variations in long-term programming.

On the other hand, a larger compliance window value will allow for a smoother and more consistent average overall for longer program durations, but can reduce the effectiveness of short-term normalization, especially program elements shorter in duration than the compliance window.

For short-term programs, a value in the range of 20-30 seconds is recommended. Larger values, from 60-120 seconds, are recommended for mid- to long-form content. For profiles which rely upon Dialog Normalization, a setting between 120-180 seconds is highly recommended.

Average Input Level

The value of the **Average Input Level control** (K) should be set to match the average level of the input audio as it serves as the mid-point value of the comfort zone defined by the **Null Area Coefficient** control.

For content that has been previously analyzed and loudness corrected in the file domain, this value will be easy to determine.

For live programming, unprocessed material, or in cases where programming is of different genres from different decades, this becomes more challenging. For example, cinematic content is typically mixed to -31/-27 LUFS, while music productions can be as loud on average as -12/-8 LUFS.

In these cases where the average input level is not easy to predict, it is best to enter a value that matches the output target loudness value, enable **Adaptive Input Detection**, and take advantage of APTO's intelligent dynamics processing.

10 Specifications

Processing

♦ Linear Acoustic® APTO™ Loudness Adaptation Algorithm with processing for two PCM stereo or mono programs

I/O

AES-3 I/O

♦ One 2-channel input and one 2-channel output via 75 Ohm BNC female connectors, internally terminated; signal levels per SMPTE 276M/AES-3ID-2001

SDI I/O

♦ One auto-sensing 3Gb/s HD/SD-SDI (SMPTE ST425/292M/259M) input and one output via 75 Ohm BNC female connectors, internally terminated; video formats up to 1081i/60/59.94/50Hz

AES67 I/O

♦ 16-channels of bi-directional AES67 I/O in support of SMPTE ST 2110-30 workflows

Latency (V0.0.64 Software)

♦ AES-3 input to AES-3 output with SRCs disabled: 30.5ms

♦ AES-3 input to AES-3 output with AES-3 input SRC enabled: 33.5ms

♦ SDI input to SDI output with SRCs disabled: 33.5ms

♦ SDI input to SDI output with SRCs enabled: 36.5ms

♦ AES67 input to AES67 output: 107ms (measurement made using Telos Alliance SDI xNode which has a latency value of 80ms)

Clock Reference

♦ 48kHz reference via SDI, AES-3, PTP v2, or internal clock

Sample Rate/Resolution/Frequency Response

♦ 48kHz, 24-bit, 20Hz – 20kHz

Ethernet

♦ Two Gigabit RJ-45 connections, one each for AES67 and Control

Parallel GPI/O Control Port

♦ Future feature; 15-pin female D connector, 0-5V TTL levels; 5 inputs, 5 outputs

Front Panel Controls and Indicators

♦ 5-key navigation cluster; graphical color LCD display; LED status indicators for each power supply, status, and reference

Power

♦ Dual internal redundant auto-ranging power supplies, each rated at 100-264VAC, 50/60Hz, 40 Watts maximum

Dimensions and Weight

♦ 19"W x 9"D x 1.75" H (approximately 48.2 x 22.0 x 4.5 cm)

♦ Net weight approximately 9.0 pounds (4.08kg)

♦ Shipping weight approximately 12.0 pounds (5.44kg)

Environmental

♦ Operating temperature 0 to 50 degrees Celsius (32 to 122 degrees Fahrenheit)

♦ Non-operating temperature -20 to 70 degrees Celsius (-4 to 158 degrees Fahrenheit)

Intended Location

♦ Telecommunications center or dedicated computer/machine room

Regulatory

♦ North America – FCC and CE tested and compliant with UL-approved power supplies

♦ Europe – Complies with European Union Directive 2002/95/EC on the restriction of use of certain hazardous substances in electrical and electronic equipment (RoHS), as amended by Commission Decisions 2005/618/EC, 2005/717/EC, 2005/747/EC (RoHS directive), and WEEE

Warranty

♦ Standard Telos Alliance 2-year limited parts and labor

11 Addendum 1 - ARC V1.2 Software

ARC V1.2 Software

Version 1.2 of the ARC software was released in April 2020. V1.2 is, as of this writing, the current shipping version. All ARC units running the original V0.0.64 software can be field updated. The update and installation instructions can be downloaded from the Telos Alliance website at www.telosalliance.com/linear/arc

New features included in V1.2 include support for SNMP V1.2 and logging for program loudness, loudness events, and system events.

SNMP

Controls for configuring SNMP are located in the **System > IP** menu of the remote user interface. To connect to an SNMP server, click on the **SNMP enable button** (A) and enter the IP address of at least one SNMP server in the **SNMP Server Address fields** (B).

Please visit www.telosalliance.com/linear/arc to download the MIB file for ARC.
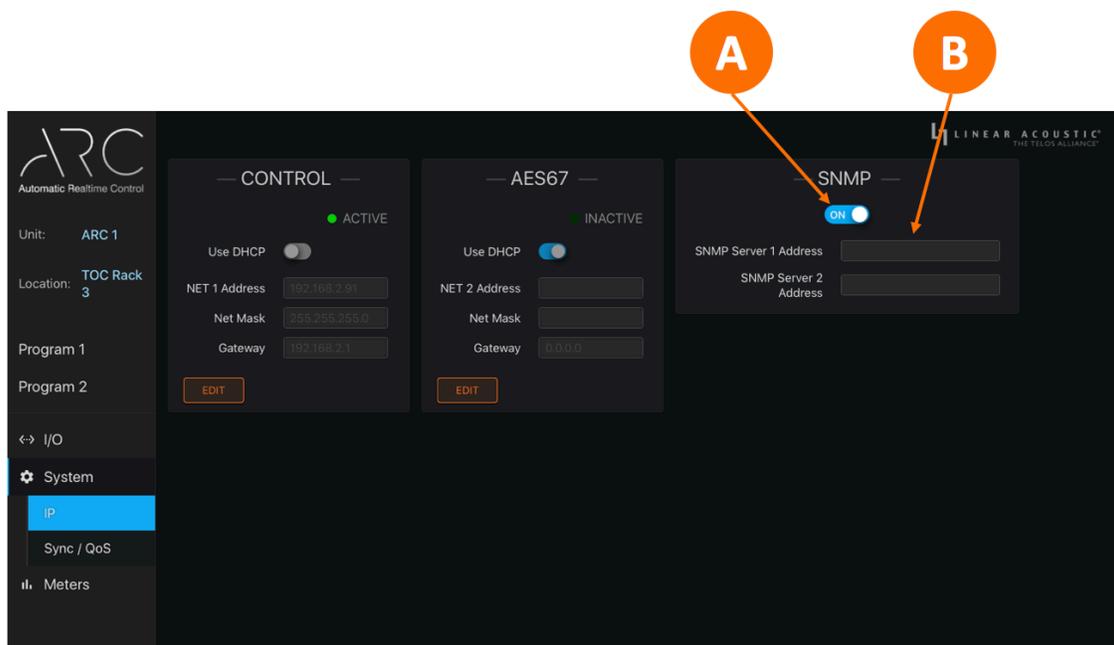
Figure A1-1 – SNMP menu

Logging

ARC has three logs available for download when the **Logging button** (A) is enabled:

♦ A 6-hour rolling loudness log for each program path

♦ A loudness event log

♦ A system event log

Logs will typically download to the "Downloads" folder unless you have deliberately changed the location for downloads on your computer.

Loudness Logs

Clicking on the **Download Loudness Log button** (B) will download three csv files: A 6-hour rolling loudness log for each program path, plus a loudness event log.

The loudness event log will contain instances where the output loudness level has fallen below or exceeded the target output level according to the recommendations and regulations associated with the current APTO processing profile (such as EBU R 128 and ATSC A/85).

Other audio-related events include:

♦ Input audio falling below -70dB (indicating the loss of input audio)

♦ Input audio rising above -70dB (indicating the return of input audio)

♦ Changes to the reference clock source

♦ Invalid reference clock

♦ Unit being placed into bypass

System Logs

Clicking on the **Download System Log button** (C) will download a .log file that can be opened with a text editor such as Notepad (Windows) or TextEdit (Mac). This log will report any system-related issues and can be used by customer support for diagnostic and trouble-shooting purposes.
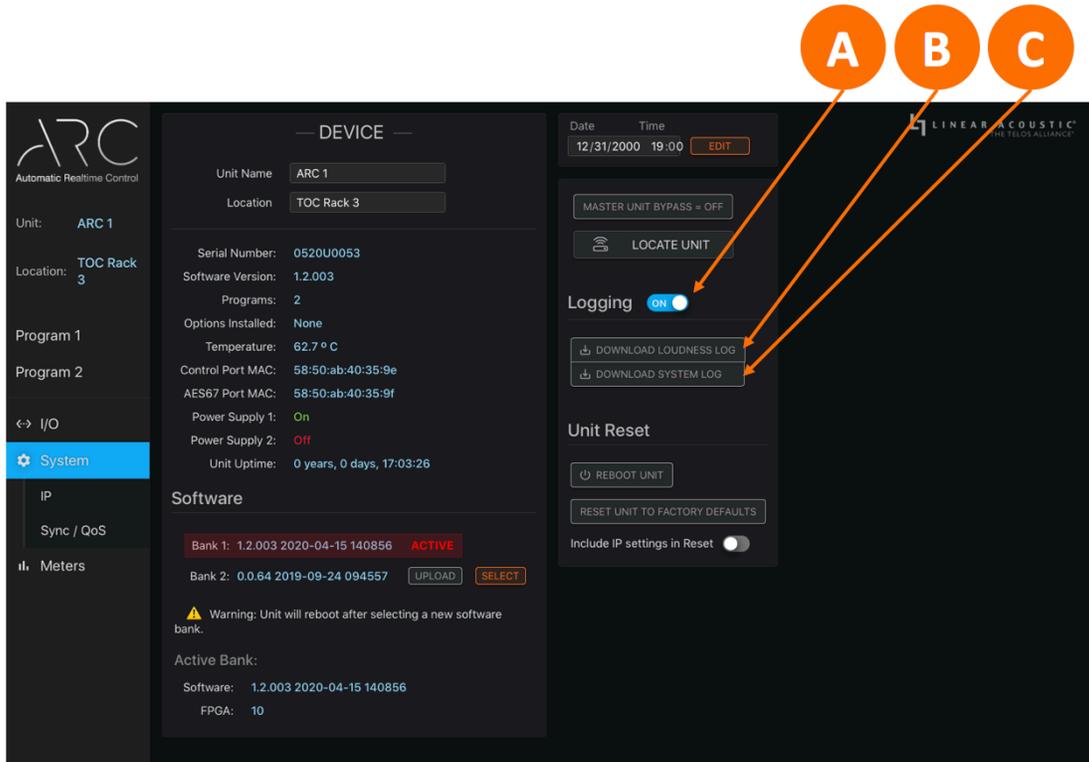
Figure A1-2 – Logging menu

Telos Alliance Warranty

Telos Alliance Limited Warranty

For the latest Telos Alliance warranty, visit: telosalliance.com/warranty.