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Application Examples

Selenio 6800™ DAPM6802+D
Dual Audio Processing Module

Edition A

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Publication Information

Trademarks

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DAPM6802+
Dual-Channel
Audio Processing Module

Installation and Operation Manual

Edition A
October 2012

Contents

Preface

Manual Information

Purpose

This manual details the features, installation, operation, maintenance, and specifications for the DAPM6802+ Dual-Channel Audio Processing Module.

Audience

This manual is written for engineers, technicians, and operators responsible for installation, setup, maintenance, and/or operation of the DAPM6802+ Dual-Channel Audio Processing Module.

Revision History

Table 1-1 Revision History of Manual

Edition	Date	Comments
A	October 2012	Initial release

Writing Conventions

This manual adheres to the following writing conventions.

Table P-2. Writing Conventions

Term or Convention	Description
Bold	Indicates dialog box, property sheet, field, button, check box, list box, combo box, menu, submenu, window, list, and selection names
Italics	Indicates email addresses, names of books and publications, and first instances of new terms and specialized words that need emphasis
CAPS	Indicates a specific key on the keyboard, such as ENTER, TAB, CTRL, ALT, DELETE
`Code`	Indicates variables or command-line entries, such as a DOS entry or something you type into a field.
>	Indicates the direction of navigation through a hierarchy of menus and windows.

Table P-2. Writing Conventions *(Continued)*

Term or Convention	Description
hyperlink	Indicates a jump to another location within the electronic document or elsewhere
Internet address	Indicates a jump to a Web site or URL
Note:	Indicates important information that helps to avoid and troubleshoot problems

Obtaining Documents

Product support documents can be viewed or downloaded from our website. Alternatively, contact your Customer Service representative to request a document.

Unpacking/Shipping Information

Unpacking a Product

This product was carefully inspected, tested, and calibrated before shipment to ensure years of stable and trouble-free service.

1. Check equipment for any visible damage that may have occurred during transit.
2. Confirm that you have received all items listed on the packing list.
3. Contact your dealer if any item on the packing list is missing.
4. Contact the carrier if any item is damaged.
5. Remove all packaging material from the product and its associated components before you install the unit.

   Keep at least one set of original packaging, in the event that you need to return a product for servicing.

Product Servicing

Except for firmware upgrades, DAPM6802+ modules are not designed for field servicing. All hardware upgrades, modifications, or repairs require you to return the modules to the Customer Service center.

Returning a Product

In the unlikely event that your product fails to operate properly, please contact Customer Service to obtain a Return Authorization (RA) number, and then send the unit back for servicing.

Keep at least one set of original packaging in the event that a product needs to be returned for service. If the original package is not available, you can supply your own packaging as long as it meets the following criteria:

- The packaging must be able to withstand the product's weight.
- The product must be held rigid within the packaging.
- There must be at least 2 in. (5 cm) of space between the product and the container.
- The corners of the product must be protected.

Ship products back to us for servicing prepaid and, if possible, in the original packaging material. If the product is still within the warranty period, we will return the product prepaid after servicing.

Safety

Carefully review all safety precautions to avoid injury and prevent damage to this product or any products connected to it. If this product is rack-mountable, it should be mounted in an appropriate rack using the rack-mounting positions and rear support guides provided. It is recommended that each frame be connected to a separate electrical circuit for protection against circuit overloading. If this product relies on forced air cooling, it is recommended that all obstructions to the air flow be removed prior to mounting the frame in the rack.

If this product has a provision for external earth grounding, it is recommended that the frame be grounded to earth via the protective earth ground on the rear panel.

IMPORTANT! Only qualified personnel should perform service procedures.

Safety Terms and Symbols in this Manual

WARNING

Statements identifying conditions or practices that may result in personal injury or loss of life. High voltage is present.

CAUTION

Statements identifying conditions or practices that can result in damage to the equipment or other property.

References

ANSI/SMPTE 259M-2006
SDTV Digital Signal/Data – Serial Digital Interface

SMPTE 292M-2006
1.5 Gb/s Signal/Data Serial Interface

SMPTE 291M-2006
Ancillary Data Packet and Space Formatting

SMPTE 346-M 2000
Time Division Multiplexing Video Signals and Generic Data over High-Definition Interface

SMPTE 352-M 2002
Video Payload Identification for Digital Interfaces

SMPTE 2020-1-2008
Format of Audio Metadata and Description of the Async Serial Bitstream Transport.

SMPTE 2020-2-2008
Vertical Ancillary Data Mapping of Audio Metadata - Method A

SMPTE 2020-3-2008
Vertical Ancillary Data Mapping of Audio Metadata - Method B

SMPTE RP 184-2004
Specification of Jitter in Bit-Serial Digital Systems

EIA/TIA-422-B 1994
Electrical Characteristics of Balanced Voltage Digital Interface Circuits

EN55103-1
EMC emission requirements applies to professional audio, video, audio-visual and entertainment lighting control apparatus

EN55103-2
EMC immunity requirements applies to professional audio, video, audio-visual and entertainment lighting control apparatus

ITU-R BT.601-5
Studio Encoding Parameters of Digital Television for Standard 4:3 and Wide-Screen 16:9 Aspect Ratios

ITU-R BT.709-5
Parameter Values for the HDTV Standards for Production and International Programme Exchange

47 Code of Federal Regulations
Part 15 FCC rules—Radio Frequency Devices

Restriction on Hazardous Substances (RoHS) Compliance

Directive 2002/95/EC—commonly known as the European Union (EU) Restriction on Hazardous Substances (RoHS)—sets limits on the use of certain substances found in electrical and electronic equipment. The intent of this legislation is to reduce the amount of hazardous chemicals that may leach out of landfill sites or otherwise contaminate the environment during end-of-life recycling. The Directive, which took effect on July 1, 2006, refers to the following hazardous substances:

- Lead (Pb)
- Mercury (Hg)
- Cadmium (Cd)
- Hexavalent Chromium (Cr-V1)
- Polybrominated Biphenyls (PBB)
- Polybrominated Diphenyl Ethers (PBDE)

According to this EU Directive, all products sold in the European Union will be fully RoHS-compliant and “lead-free.” (See our website for more information.) Spare parts supplied for the repair and upgrade of equipment sold before July 1, 2006 are exempt from the legislation. Equipment that complies with the EU directive will be marked with a RoHS-compliant emblem, as shown in **Figure 1-1**.

Figure 1-1 RoHS Compliance Emblem

Waste from Electrical and Electronic Equipment (WEEE) Compliance

The European Union (EU) Directive 2002/96/EC on Waste from Electrical and Electronic Equipment (WEEE) deals with the collection, treatment, recovery, and recycling of electrical and electronic waste products. The objective of the WEEE Directive is to assign the responsibility for the disposal of associated hazardous waste to either the producers or users of these products. As of August 13, 2005, the producers or users of these products were required to recycle electrical and electronic equipment at end of its useful life, and may not dispose of the equipment in landfills or by using other unapproved methods. (Some EU member states may have different deadlines.)

In accordance with this EU Directive, companies selling electric or electronic devices in the EU will affix labels indicating that such products must be properly recycled. (See our website for more information.) Contact your local sales representative for information on returning these products for recycling. Equipment that complies with the EU directive will be marked with a WEEE-compliant emblem, as shown in **Figure 1-2**.

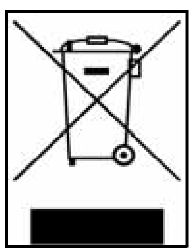


Figure 1-2 WEEE Compliance Emblem

1 Introduction

Overview

The DAPM6802+ is a Dual-Channel Audio Processing Module for managing loudness and surround sound audio streams. Featuring DTS® Neural Surround UpMix, DownMix and MultiMerge and Loudness Control, the DAPM6802+ combines unparalleled surround sound image placement and stability with loudness management processing to deliver a natural, open quality not found in traditional multiband compression technology. Flexible user-defined workflows and intelligent metadata handling ensure that the right processing is applied at the right time, allowing broadcasters to meet regulations while preserving the artistic integrity of the content. The DAPM6802+ can be configured to support a wide variety of dual-channel applications.

The DAPM6802+ differs from the APM6803+ in the following ways:

- Dolby® options and 3G-SDI options are not supported on the DAPM6802+.
- With a higher density, the DAPM6802+processes two SDI channels and four outputs (two per channel), instead of one in/out SDI channel, as found on the APM6803+.
- The DAPM6802+ module does not provide frame sync or AES in/out support.
- The DAPM6802+ includes two advanced audio processors, instead of four.

Main Features

- HD/SD-SDI input and output capability
- HD/SD-SDI compliant bypass relay
- Two fully independent SDI channels with multiple channels of embedded audio
- Virtual audio stream interface for simplified configuration and control
- DTS Neural Loudness Control on each output stream
- Dual Surround Sound processors for implementing DTS Neural Surround™ UpMix, DownMix and MultiMerge
- Surround Field Protection using DTS Neural Surround™ MultiMerge ensures smooth, consistent Surround Sound output, while input switches between stereo and surround sound sources
- Dynamic DTS Neural license allocation
- Automatic audio/video delay alignment for consistent lip synchronization
- Automation control for dynamic on-air changes to loudness control profile
- Q-SEE™-compliant thumbnails and alarms
- Custom Loudness Control and UpMix/DownMix/MultiMerge Presets
- Custom GPI and parameter scripting support

Monitoring and Operating

The DAPM6802+ can be operated locally (using card-edge controls); or operated and monitored remotely with:

- Web browser
- Control software applications such as CCS Navigator™
- CCS-compliant remote control panels such as NUCLEUS.
- Third-party SNMP-based control applications

Front Module

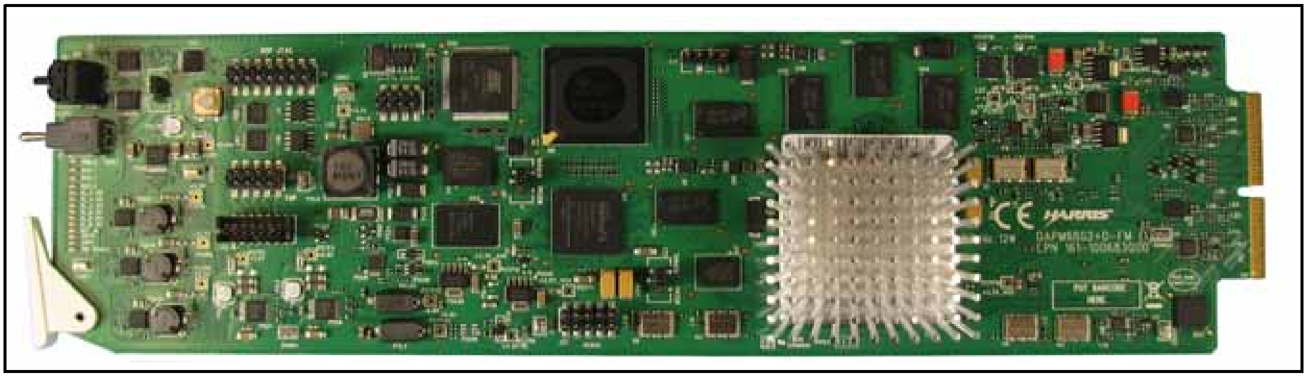


Figure 1-1 DAPM6802+ Front Module

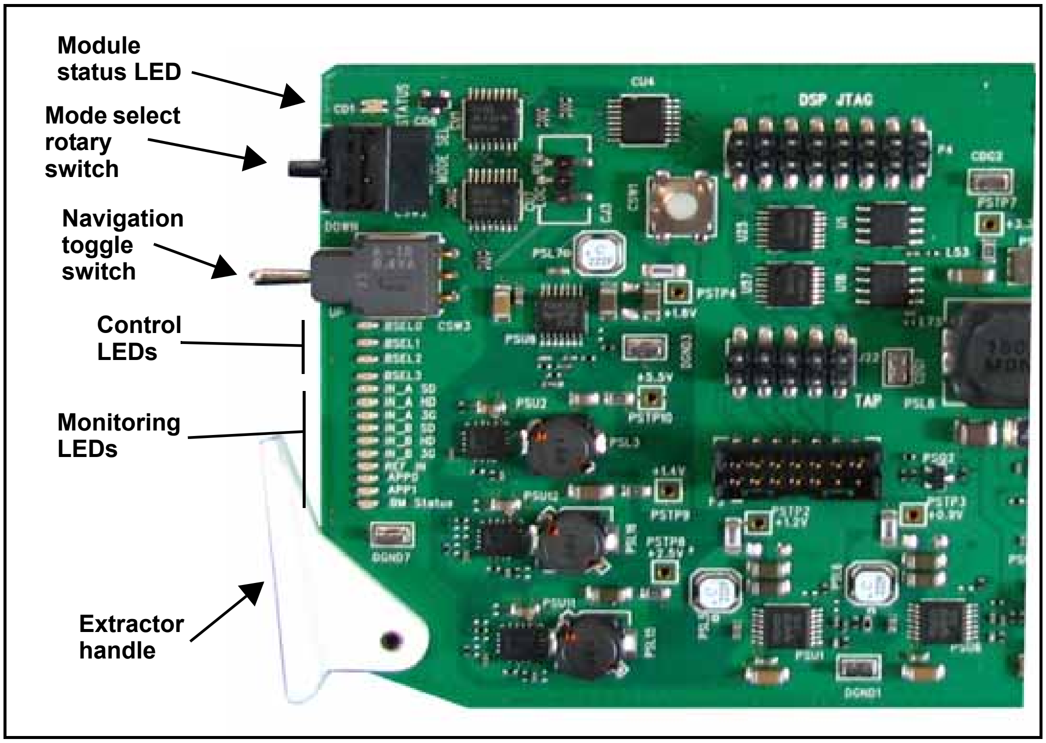


Figure 1-2 DAPM6802+ Card Edge

Table 1-1 Generic 6800**+** Module Features

Feature	Description
Module status LEDs	Various color and lighting combinations of these LEDs indicate the module state. See page 20 for more information.
Mode select rotary switch	This switch selects between various control and feedback parameters.
Navigation toggle switch	This switch navigates up and down through the available control parameters: ■ **Down**: Moves down through the parameters ■ **Up**: Moves up through the parameters
Control LEDs	Various lighting combinations of these Control LEDs (also referred to as “Bank Select LEDs”) indicate the currently selected bank. See page 19 for more information.
Monitoring LEDs	See page 19 for a description of these LEDs.
Local/Remote control jumper	■ **Local**: Locks out external control panels and allows card-edge control only; limits the functionality of remote software applications to monitoring. ■ **Remote**: Allows remote or local (card-edge) configuration, operation, and monitoring of the DAPM6802+ module.

Back Modules

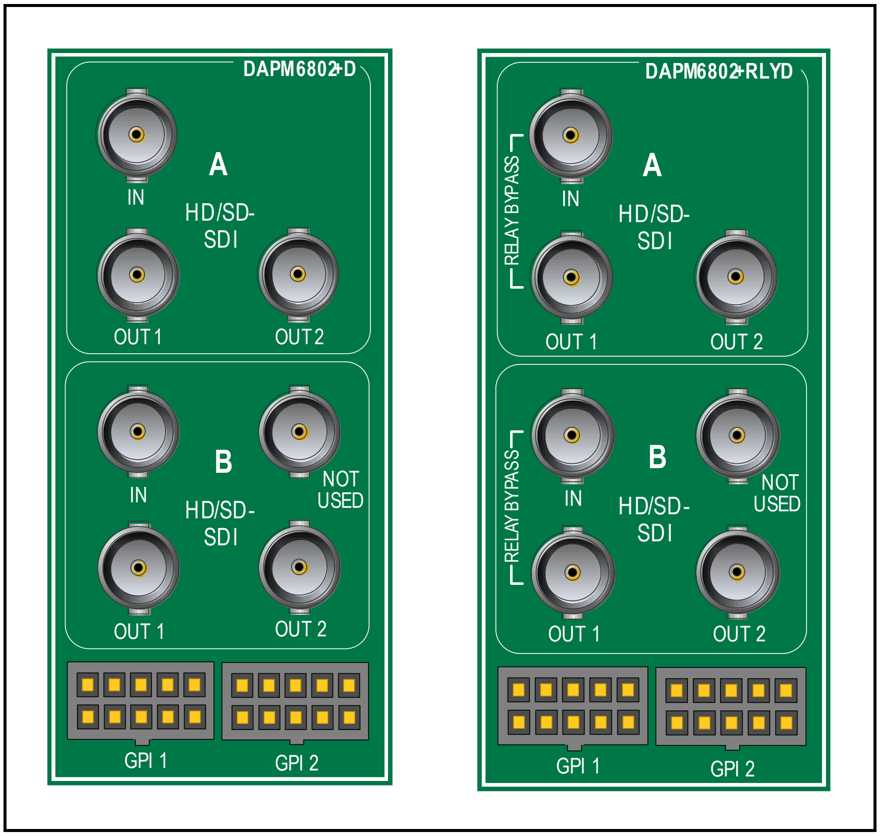


Figure 1-3 DAPM6802+D Standard and DAPM6802+RLYD Relay Back Modules

Relay Back Module

The DAPM6802+RLYD relay back module is designed to provide passthrough output in the event of a power failure. In both **Channel A** and **Channel B**, the SDI input is directly connected to **SDI Out 1**.

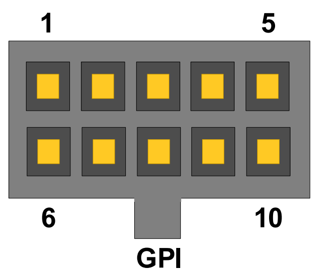
The relay back module functions as follows:

- When the front module is operating normally, the relay is closed and the signal is not bypassed.
- If the front module is pulled out of the frame or the frame loses power, the relay is opened and the signal is bypassed from **SDI In** to **SDI Out 1**.
- When the front module is inserted into the frame, the signal is bypassed until the card boots up. After the card successfully boots, the relay is closed again.

General Purpose Interface (GPI) Connectors

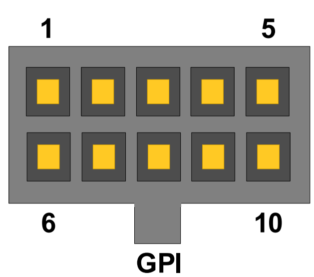
There are two GPI Input/Output (I/O) connectors on the DAPM6802+ back modules. The tables below describe the inputs and outputs as seen on the back module connectors. .

Table 1-2 GPI I/O 1



GPI I/O 1	Pin No.	Description
	1	GPI In 1
	2	GND
	3	GPI In 2
	4	GND
	5	GPI In 3
	6	GPI Out 2
	7	GND
	8	GPI Out 1
	9	GND
	10	GPI In 4

Table 1-3 GPI I/O 2



GPI I/O 2	Pin No.	Description
	1	GPI In 5
	2	GND
	3	GPI In 6
	4	GND
	5	GPI In 7
	6	GPI Out 4
	7	GND
	8	GPI Out 3
	9	GND
	10	GPI In 8

Functional Block

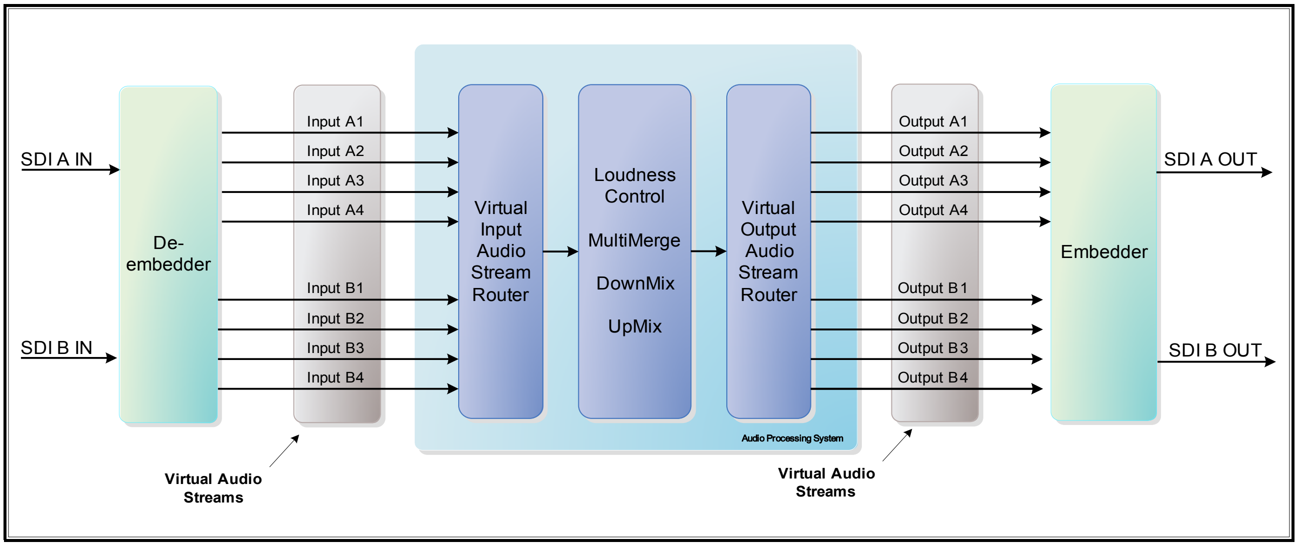


Figure 1-4 DAPM6802+ Functional Diagram

Signal Flow

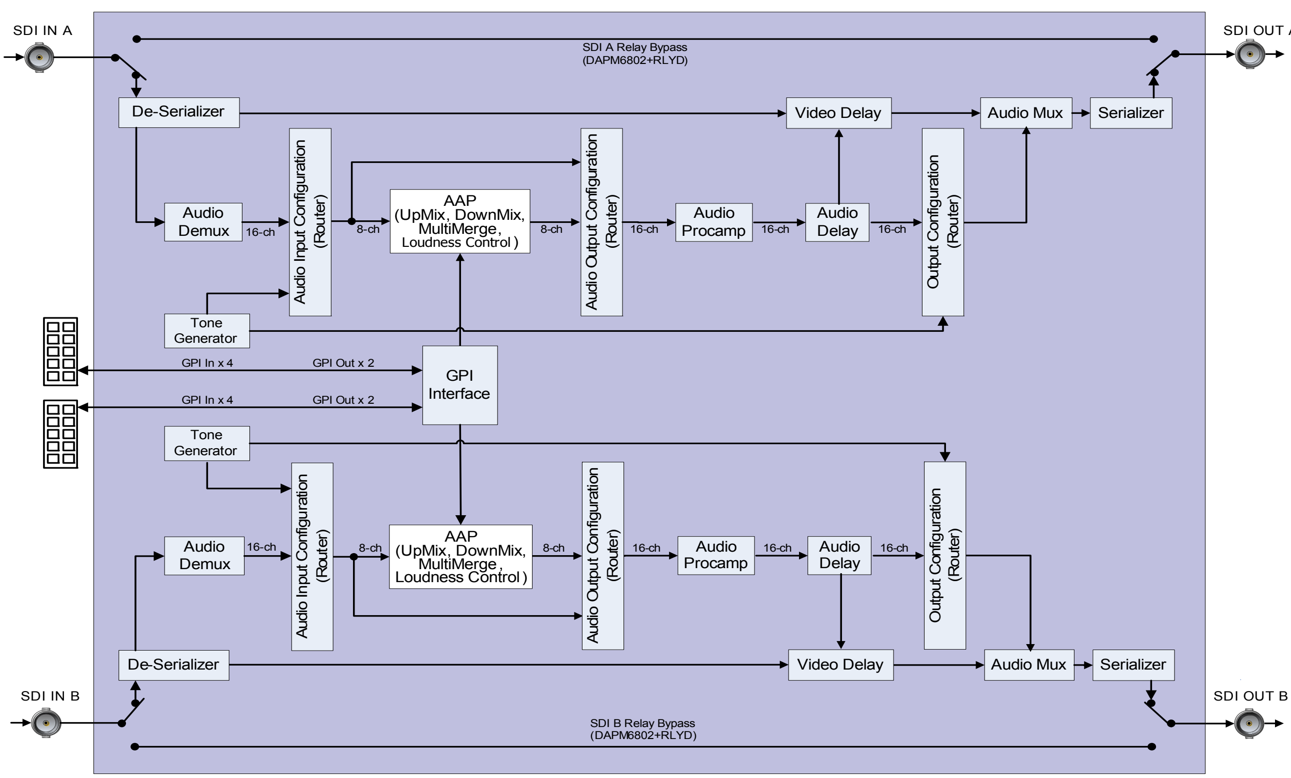


Figure 1-5 DAPM6802+ Signal Flow

Architecture

The DAPM6802+ includes an Audio Processing System consisting of one high-end Digital Signal Processor (DSP), as well as custom-built hardware, that can process multiple channels of audio (see ***Figure 1-4*** on page 6)**.** The Audio Processing System is a powerful and flexible signal processing engine. A virtual stream interface and flexible license management make it possible to configure the system architecture dynamically and tailor the processing according to individual needs. For details, see ***DTS Neural Audio Processing*** on page 23.

Virtual Stream Interface

The DAPM6802+ uses a virtual stream interface to simplify configuration, hiding the complexity of the underlying system and making it possible to create a high-level, processing-intensive signal flow with just a few mouse clicks. The interface to the Audio Processing System consists of four input and four output virtual audio streams. Signal routing is applied automatically by the DAPM6802+ to each sub channel within the stream as appropriate. Managing audio at the "stream" level makes configuration easier, quicker, and less error-prone, enabling the full processing power of the APM6803+ without bogging down the user with complicated routing and configuration menus.The DAPM6802+ also provide traditional individual channel routing management.

Resource Utilization

To take full advantage of the processing power of the DAPM6802+, you can modify the architecture as needed. If a particular configuration exceeds the capabilities of the Audio Processing System, the module tracks and reports key resource availability. For more details, see ***Licensing and Utilization Parameters*** on page 13.

Licensing

The Audio Processing System is highly configurable and can support a wide range of DTS Neural processing blocks. For DTS Neural functions, a flexible token system makes it possible to change the type of processing without purchasing additional licenses. All licenses are field- upgradable; you can start with a minimal setup and grow the system as requirements dictate. For more details, see page 13.

Real Time Loudness Control

DTS Neural Loudness Control uses patented, critical band analysis to implement a psychoacoustic model of the human ear, resulting in highly accurate perceptual loudness measurements. This technology makes it possible to implement wideband control and deliver natural, open-sounding audio with the original spectral integrity preserved. DTS Neural Loudness Control exceeds ITU-R BS.1770 requirements. The DAPM6802+ offers a choice between ITU-R BS.1770 loudness measurement and DTS Neural Loudness measurement.

Automation Control

Predefined Loudness Control settings are available for instant recall by automation, allowing users to create simple, effective automation profiles. Additional user-defined presets are available for maximum flexibility. The DAPM6802+ has a large number of GPI ports available for automation control.

2 Installation

Unpacking the Module

Before you install modules, perform the following:

- Check the equipment for any visible damage that may have occurred during transit.
- Confirm receipt of all items on the packing list. Contact your Customer Service representative if parts are missing or damaged.
- Remove the anti-static shipping pouch, if present, and all other packaging material.
- Retain the original packaging materials for possible re-use.
- See ***Unpacking/Shipping Information*** on page viii for information about returning a product for servicing.

Maximum 6800+ Frame Power Ratings

The power consumption for a DAPM6802+ module is 12 W.

Table 2-1 describes the maximum allowable power ratings for 6800**+** frames. Note the given maximums before installing any 6800**+** modules in your frame.

DAPM6802+ modules operate only in fan-cooled FR6802**+** and FR6822**+** frames, subject to the limitations shown in **Table 2-1**. These modules cannot be installed in FR6802+DM or 6800/7000 series frames.

Note: *To maintain proper temperatures, ensure that the front panel is closed at all times and that the fan module is fully operational.*

Table 2-1 Maximum Power Ratings for 6800**+** Frames

6800+ Frame Type	Max. Frame Power Dissipation	Number of Usable Slots	Max. Power Dissipation Per Slot
FR6802**+**XF (frame with AC power supply)	120 W	20	6 W
FR6802**+**XF48 (frame with DC power supply)	105 W	20	5.25 W
FR6802**+**QXF frame (with AC or DC power supply)	120W	20	6 W
FR6822**+** frame (with AC or DC power supply)*	120W	20	6 W
*Recommended frame			

See your 6800+ series *Frame Installation and Operation Manual* for information about installing and operating the frame and its components.

CAUTION
Before installing this product, read the *6800+ Series Safety Instructions and Standards Manual* shipped with every *6800+ Frame Installation and Operation Manual* or downloadable from our website. This safety manual contains important information about the safe installation and operation of 6800+ series products.

Installing 6800+ Modules

Installing and Removing DAPM6802+ Modules

DAPM6802+ modules require no specialized installation or removal procedures. However, when installing both front and rear modules, ensure that the back module is installed first before plugging in the front module.

When removing both the front and rear modules, ensure that the front module is unplugged from the frame first, before removing the rear module.

The DAPM6802+ package includes two mini-mate header cables for GPI connectors.

Jumper Setting

Note: *The DAPM6802+ main module has one jumper (***CJ3***), which sets the module for local or remote control. You need to configure modules for local or remote operation prior to power-up. To change the configuration, remove the module, reset the jumper, and then re-insert the module into the frame.*

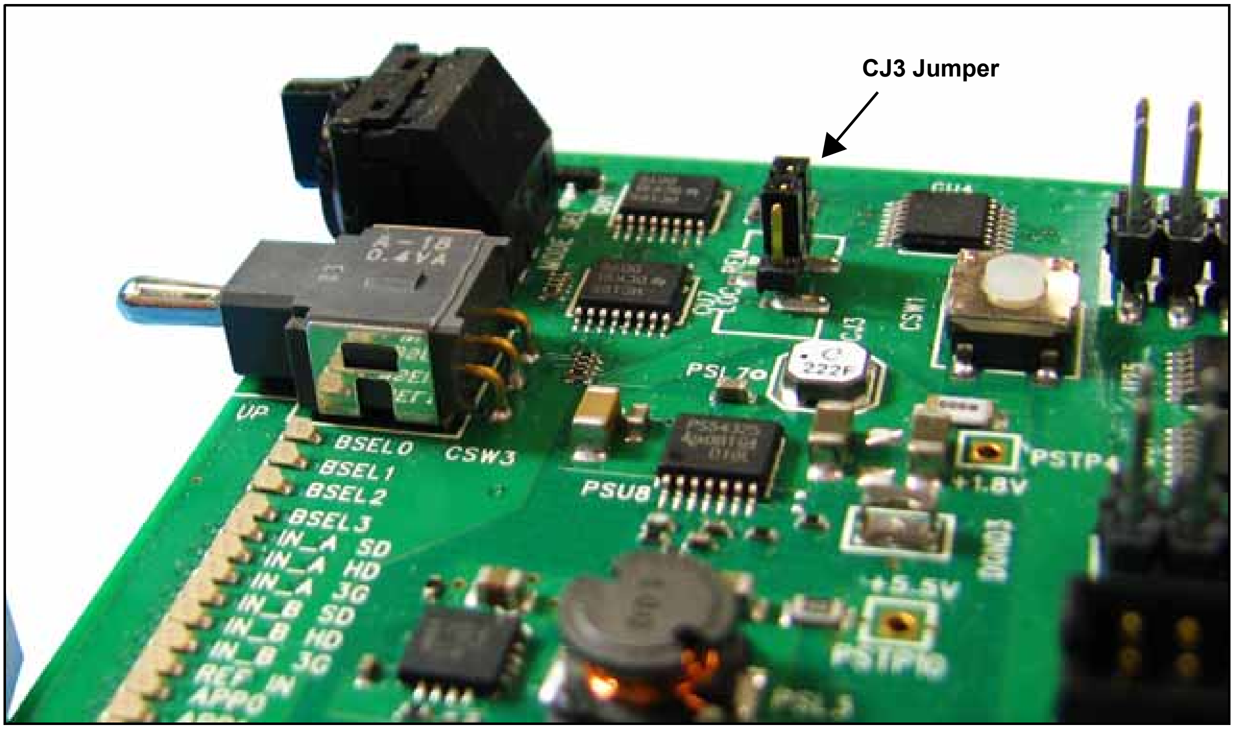


Figure 2-1 Jumper Location

1 Locate jumper **CJ3** on the module (behind the mode select rotary switch). **Figure 2-1** shows the location of the **CJ3** jumper.

2 Place the jumper on pins 1 and 2 to set the module for **Remote** control, or pins 2 and 3 to set the module for **Local** control (the white triangle near the jumper pins on the module indicates pin 1).

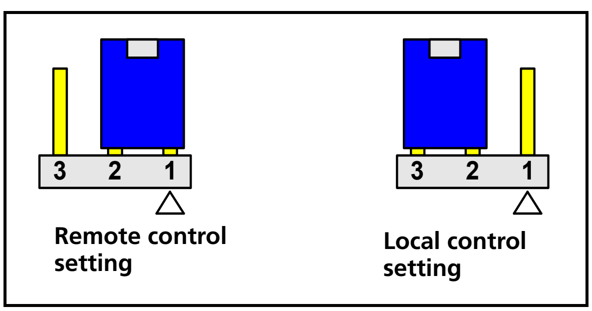


Figure 2-2 CJ3 Settings for Local and Remote Control

The **Local** setting locks out external control panels and allows card-edge control only; remote software applications can monitor, but not control the module. The **Remote** setting allows either remote or local (card-edge) configuration, operation, and monitoring of the DAPM6802+.

Upgrading Module Firmware

This module's firmware can be updated using CCS Navigator version 4.8 or later, or the HTTP software upgrade tool. In order to perform these upgrades, your frame must be equipped with a 6800+ETH module. See your frame manual for more information.

Note: *Ensure the 6800+ETH resource module is at version 4.8 or later. When upgrading, do not remove the DAPM6802+ module from the frame.*

DTS Neural Licensing

DTS Neural licenses enable a wide variety of DTS Neural audio processing options such as Loudness Control, UpMix, DownMix, and MultiMerge. These options make it possible to offer advanced audio processing for high-definition and surround sound programming using 5.1 and stereo sources (2.0).

DTS Neural Credits

The number of credits determine how many DTS Neural functions are available. The following tables provide the number of credits and utilization required for each DTS Neural function.

Table 2-2 DTS Neural Credits

Function	Credits Required	Utilization (%)
UpMix	3	38
DownMix	3	13
MultiMerge	4	51
Loudness Control 2.0	1	14
Loudness Control 5.1	3	25
Loudness Control 4 x 2.0	4	44
Loudness Control 5.1+2.0	4	34
Loudness Control 2.0 + UpMix	4	51
Downmix + Loudness Control 2.0	4	19
MultiMerge + Loudness Control 2.0 + Loudness Control 5.1	8	71
Loudness Control 2 x 1.0	1	20

You can have a maximum of 14 credits. For example, a combination of SDI_A of MultiMerge and SDI_B of Loudness Control 5.1 will require 7 credits.

Credits are flexible. You can use available credits for different functions at different times. For instance, if you have 10 credits, you could use a **Loudness Control 2.0 + UpMix** at one time and a **MultiMerge** at another time. Each 68OPT-DTS license option provides a single credit. In other words, to use the **MultiMerge** function, you would need to purchase four 68OPT-DTS licenses.

If the selected DTS Neural function exceeds the number of licensed credits or 100% utilization, one of the following messages may be displayed.

Table 2-3 AAP Error Messages

AAP Status Feedback	Description
Failed-No License	Adequate licenses are not available for processing.
Failed-DSP Full	The DSP (Digital Signal Processor) does not have enough resources.

Licensing and Utilization Parameters

Licensing and Utilization parameters can be viewed or set by accessing your module and frame through a web browser or through CCS Navigator.

- The **Licensing Parameters** show how many DTS Neural credits you have available. (**Parameters > General > Licensing**)
- The **Utilization Parameters** show how many of your licensed credits are used. (**Parameters > General > Utilization**)

Licensing Parameters

Table 2-4 Licensing Parameters

Parameter Name	Function	Options
Serial Number [RO]	Displays the Serial Number.	
License Key	Allows you to input your license key.	
DTS Credits Licensed [RO]	Shows number of DTS Neural credits licensed.	0 to 14

Utilization Parameters

Table 2-5 Utilization Parameters

Parameter Name	Function	Range
DTS Credits Licensed [RO]	Shows number of DTS Neural credits licensed.	0 to 14
DTS Credits Used [RO]	Shows number of DTS Neural credits used.	0 to 14

Adding a License Key

Note: *For assistance with a license key, or to purchase a license key, please contact your Sales representative.*

Through the HTTP Interface

1. Go to **Parameters** > **General** > **Licensing**
2. Enter your license in the **License Key** text box.

Through CCS Navigator

To enter a license key, your CCS software must be in Control mode.

1. Select the DAPM6802+ module in the **Navigation** pane, right click, and then select **Control** to open the module's **Control** window.
2. Select the **Parameters** tab.
3. In the tree view, select **Parameters > General > Licensing** and enter your license key in the **License Key** field.

.

3 Parameters, Alarms, and LEDs

Overview

The following sections are covered here:

Parameter Categories

The following table lists the main categories that Parameters are divided into and what you can control from those categories:

Table 3-1 Parameter Categories

Parameter Name	Options
General Parameters	■ Licensing and Utilization parameters. ■ GPI Input/Output/Custom Parameters ■ Parameter Control Script
SDI A-B Video Configuration	■ SDI ■ Video Delay
SDI A-B Audio Configuration	■ Input Configuration ■ Advanced Audio Processing ■ Output Configuration ■ Output Router ■ Test Tones
SDI A-B Audio Status	■ V-bit Status ■ Error Status ■ Control Packet Status
Alarms	Alarm Configuration

Operation Notes

- See **Chapter 5, *Custom Scripting*** for details on custom parameter scripts.
- For a complete list of parameters, refer the HTM Parameter list (available with the documentation).
- You can access DAPM6802+ parameters through CCS Navigator, an HTTP web browser, or a third-party SNMP-based control application (depending on your host frame's options).
- When you change a parameter, the effect is immediate. However, the module requires up to 30 seconds to save the latest change. After 30 seconds, the new settings are saved and will be restored if the module loses power and must be restarted.
- If you make changes to certain parameters, other related parameters may also be affected. For example, virtual stream selections can have an impact on routing.
- General Presets (loading and saving of presets) are not currently supported.

Changing Parameter Settings Using Card-Edge Controls

1. Rotate the hex switch (mode select rotary switch) to **0**.
2. Once the hex switch is set to "0," toggle the navigation switch up or down to select a bank.

   View the two control LEDs next to the navigation toggle switch to see which bank is currently selected. See **Table 3-2** on page 16 to view the various banks, hex switch positions, and corresponding parameter options and values.

Table 3-2 Selected Bank as Indicated by Control LEDs

Bank Number	LED 3	LED 2	LED 1	LED 0
0	Off	Off	Off	Off
1	Off	Off	Off	On
2	Off	Off	On	Off
3	Off	Off	On	On
4	Off	On	Off	Off
5	Off	On	Off	On
6	Off	On	On	Off
7	Off	On	On	On
8	On	Off	Off	Off
9	On	Off	Off	On
A (10)	On	Off	On	Off
B (11)	On	Off	On	On
C (12)	On	On	Off	Off
D (13)	On	On	Off	On
E (14)	On	On	On	Off
F (15)	On	On	On	On

3 Rotate the hex switch to the parameter number (1 to 9) or letter (A to F) of the option you want to set.
4 Toggle the navigation switch to select and set the value of the chosen parameter.
5 Do either of the following:
   - Rotate the hex switch to another parameter number/letter in the current bank, and then repeat step 4.
   - Rotate the hex switch to "0" again to select a different bank, and then repeat steps 3 and 4.

   Use an available 6800**+** software control option to aid in viewing, setting, and confirming the parameter value.

Note: *Refer to the HTM Parameter list (available with the documentation) for more details on changing parameter settings using card edge controls.*

Recalling Factory Default Parameter Settings

To return the DAPM6802+ module to its factory default settings, you can either reset each parameter individually or do a global recall following this procedure.

1 Rotate the hex switch to **0**.
2 Toggle the navigation switch to the bank number **0**.
3 Use the control LEDs to verify which bank you have selected, or use an available 6800**+** software control option (serial/local or Ethernet/remote) to aid in confirming your bank selection.
4 Rotate the hex switch to the global recall parameter **F**.
5 Toggle the navigation switch to **On**.

   Use an available 6800**+** software control option to aid in viewing, setting, and confirming the parameter value.

Note: *After doing a factory recall, wait for about 40 seconds before attempting any other operation.*

Changing Parameter Settings Using CCS Software

You can change the parameter settings, view read-only parameters, view alarms, and adjust alarm settings using CCS software.

Before using CCS Navigator to change your module's parameter settings, you must discover the module. Discovery is the process by which CCS Navigator finds, and then connects to your module.

Discovering Your Module Using CCS Software

To discover your module, your Navigator software must be in Build mode.

Note: *Wait for the DAPM6802+module to become available (the module should show as "Network Active") before you start accessing parameters through Navigator.*

1 If the Discovery window is not open, click **Tools > Discovery** in the main menu. A **Discovery** window opens, most likely in the bottom left corner of the screen.

2 Click **Options**, and then click **Add**.
3 Enter the IP address of the frame that contains your module, or the frame that contains a 6800+ETH module that provides access to your module.
4 Click **OK** to close the **Add Host** dialog box, and then **OK** again to close the **Discovery Options** dialog box.
5 Click **Start**. This triggers Navigator to run a discovery. When the discovery finishes, **Discovery Completed** is displayed in the **Discovery** pane.
6 Click **Save** to save the results of your discovery to the **Discovery** folder of the **Navigation** pane.
7 Switch to Control mode by selecting **Operational Mode > Control** from the main menu.
8 Double-click DAPM6802+ in the Navigation pane. The **Control** window opens displaying the module's controls.

You can now switch to Control mode by selecting **Operational Mode > Control** from the main menu. Double-click DAPM6802+ in the Navigation pane. The **Control** dialog box opens displaying the module's controls.

Reading Hardware and Software Versions

To determine the hardware version number for DAPM6802+ modules, follow these steps:

1 Discover the frame that contains the module, and save results of your discovery.
2 In the **Navigation** pane, right-click the module and select **Configuration**.
3 In the **Configuration** dialog box, select the **Version** tab.
4 In the **Item Name** list, look at the **Hardware** menu item. If it is expandable and provides a version number, then the revised specifications apply to your module. If the **Hardware** menu item is non-expandable and provides no version information, then the original specifications apply.

Changing Parameter Settings Using a Web Browser

1 Access your frame using a web browser.
2 Login to the 6800 Control Interface.
3 In the Navigation pane, click the slot that contains your APM6803+ module.
4 Parameter categories will be displayed and you can drill down.

Most of the screenshots in this manual have been taken through the web browser interface for DAPM6802+.

Recalling Factory Default Parameter Settings

You can set all parameter settings back to the factory defaults by going to **Parameters > General** and selecting Yes from the **Factory Recall** drop down list. Once default settings are restored, you will see **Factory Recall** set to **---**

Reading Hardware and Software Versions

To determine the hardware version number for DAPM6802+ modules, follow these steps:

1 Enter the IP address of the frame that contains the module.

2 In the **Navigation** pane, click on the module to expand its menu, and then select **Configuration > Version**.

If the **Hardware** item displays a version number, then the revised specifications apply to your module. If the **Hardware** menu item provides no version information, then the original specifications apply.

Card Edge LEDs

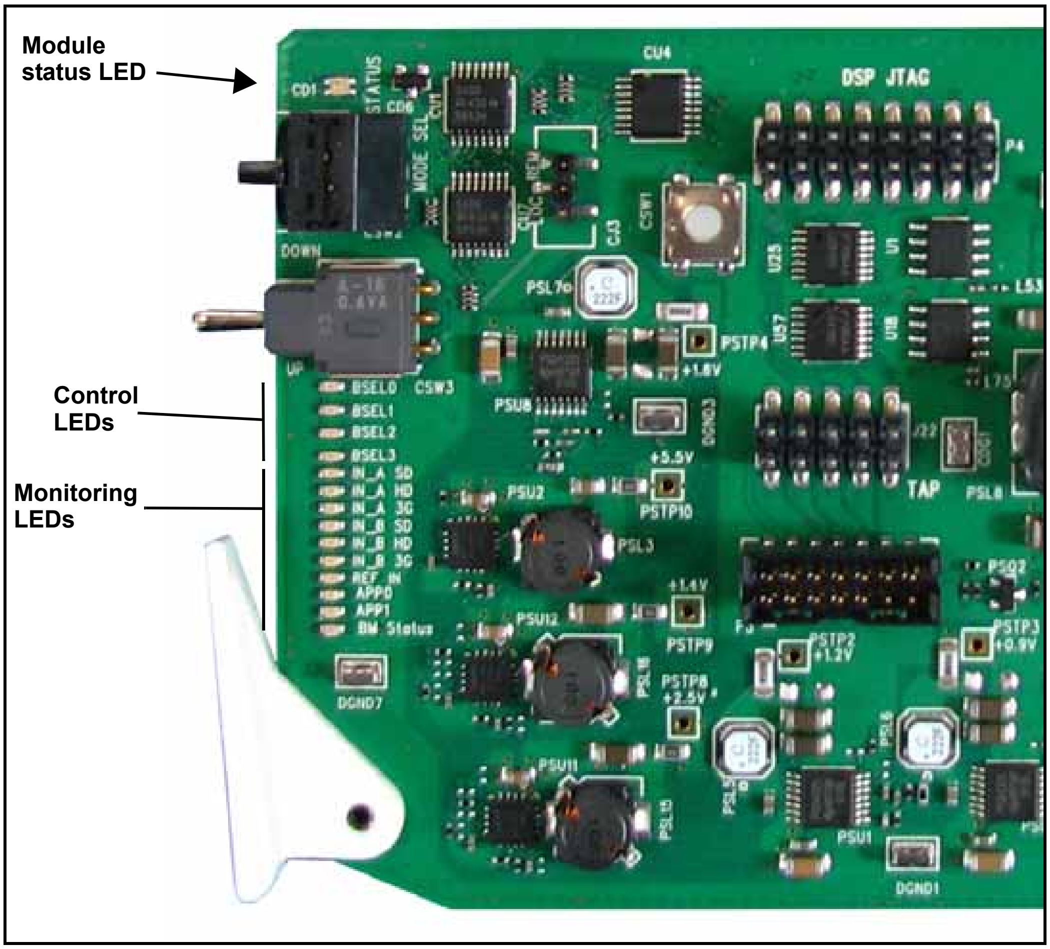


Figure 3-1 Location of DAPM6802+ LEDs

Table 3-3 Card Edge Monitoring and Control LEDs

LED	Color	Description
BSEL 0	Green	Bank LED 3-0 0000--Bank 0 0001--Bank1 1110-Bank 14 1111--Bank 15
BSEL 1	Green	
BSEL 2	Green	
BSEL 3	Green	

Table 3-3 Card Edge Monitoring and Control LEDs (Continued)

LED	Color	Description
IN_A SD	Green	Detects an SD-SDI signal at video input A ■ ON: SD-SDI A input present ■ OFF: SD-SDI A input absent
IN_A HD	Green	Detects an HD-SDI signal at video input A ■ ON: HD-SDI A input present ■ OFF: HD-SDI A input absent
IN_A 3G	Green	Not used
IN_B SD	Green	Detects an SD-SDI signal at video input B ■ ON: SD-SDI B input present ■ OFF: SD-SDI B input absent
IN_B HD	Green	Detects an HD-SDI signal at video input B ■ ON: HD-SDI B input present ■ OFF: HD-SDI B input absent
IN_B 3G	Green	Not used
REF IN	Green	Not used
AAP0	Green	Indicates the working status of AAP0 ■ ON: AAP0 operating normally ■ OFF: AAP0 not configured
AAP1	Green	Indicates the working status of AAP1 ■ ON: AAP1 operating normally ■ OFF: AAP1 not configured
BM Status	Green	Detects the back module type ON: Back module with relay OFF: Back module without relay

Module Status LED

The Module Status LED, located on the module's card edge, lights up if an error is detected. See **Figure 3-1** on page 19 for the location of this LED. The table below provides a definition of the LED colors.

Table 3-4 Module Status LED Descriptions

LED Color Sequence	Meaning
Off	There is no power to the module; the module is not operational.
Green	There is power to the module; the module is operating properly.
Red	There is an alarm condition.
Flashing red	The module has detected a hardware/firmware fault.
Amber	The module is undergoing configuration.

Note: *If the LED is flashing red, please contact your Customer Service representative.*

Alarms

If a major or minor alarm is triggered within your DAPM6802+ module, the Status LED lights red. Alarms are usually logged and monitored within available software control applications (for example, CCS Navigator). You can only differentiate between major and minor alarms within a software control application. See the appropriate software control user manual or online help for more information.

Alarm Options

The following settings can be made for each alarm within Navigator software.

Table 3-5 Alarm Options

Alarm Option	**Effect**
Enable/Disable	This option toggles between Enabled and Disabled. If the alarm is **Enabled**, an alarm condition generates an alarm; if it is **Disabled**, the alarm condition is ignored. By default, all alarms are disabled.
Alarm priority	This setting determines whether a triggered alarm is reported as major or minor. The range is 1–10. A priority of 6 or higher is a major alarm, and a priority of 5 or lower is a minor alarm.
Trigger (s)	This option determines how long an alarm condition must exist (in seconds) before the alarm is triggered. If the alarm level is reached for less time than the **Trigger** duration, then the alarm will not trigger. Choose any duration from 0 to 7200 seconds (or 2 hours). If this option is set to 0 and the alarm condition exists for any period of time, the alarm is triggered.
Clear (s)	Determines the amount of time the alarm condition must be in abatement in order for the alarm to be turned off. Choose any duration from 0 to 7200 seconds (or 2 hours). If this option is set to 0 and the alarm condition ceases for any period of time, the alarm is cleared.
Ack	When an alarm is active, click this option to allow other users on the network to see that you have acknowledged the alarm.

Alarm Definitions

Note: *For a complete list of alarms, see the DAPM6802+ Parameter list (available on the Harris Infrastructure and Networking DVD, or on the customer support website).*

4 DTS Neural Audio Processing

Overview

The following DTS Neural functions are available as options:

DTS Neural Surround Audio UpMix

Overview

The DTS Neural Surround UpMix renders any two channel audio source (stereo, matrix encoded stereo, LtRt, or DTS Neural Surround LwRw) as surround sound. The DTS Neural Surround UpMix can simultaneously position individual elements within the surround field, creating high levels of image stability and granularity.

The UpMix technology avoids taking “artistic license” with content by placing audio exactly where it would be heard in a professional LEDE (Live End Dead End) listening environment. For example, mono or pan-pot stereo will image in front of the listener, whereas stereo containing depth information will surround the listener.

You can use the DTS Neural Surround UpMix as a stand-alone unit to monitor stereo production, or you can use it in tandem with the DTS Neural Surround DownMix as a complete 5.1 transport solution. **Figure 1** below shows an UpMix taking a two-channel audio source (stereo, matrix encoded stereo, LtRt or DTS Neural Surround LwRw) and rendering a 5.1 multi-channel mix.

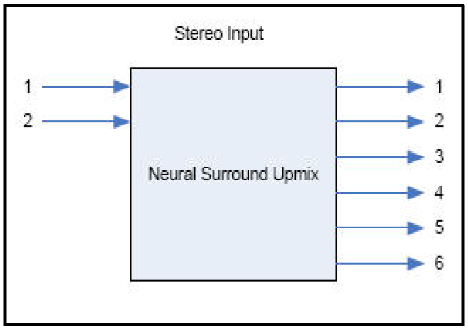


Figure 1. DTS Neural UpMix Block Diagram

Table 1. Channel Configurations for DTS Neural UpMix

Input Routing	Channel Name	2.1	3.1	4.1	5.1	6.1	7.1	Phantom 6.1	Phantom 7.1
1	Left (L)	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
2	Right (R)	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
3	Center (C)	Mute	Yes	Mute	Yes	Yes	Yes	Mute	Mute
4	Low Frequency Effects (LFE)	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
5	Left Surround (Ls)	Mute	Mute	Yes	Yes	Yes	Yes	Yes	Yes
6	Right Surround (Rs)	Mute	Mute	Yes	Yes	Yes	Yes	Yes	Yes
7	Left Back (LB) or Center Back (CB)	Mute	Mute	Mute	Mute	Yes (CB)	Yes (LB)	Yes	Yes
8	Right Back (RB)	Mute	Mute	Mute	Mute	Mute	Yes	Mute	Yes

DTS Neural UpMix Parameters

Table 4-1 UpMix Parameters

SDI x > Audio Configuration > Advanced Audio Processing > DTS Neural UpMix			
Parameter Name	**Function**	**Options**	**Default**
SDI x UpMix Status	Current status of the UpMix processing block.	■ Uninitialized ■ Running ■ Failed-function bypassed ■ Failed-output lost	Uninitialized
SDI x UpMix Channel Config	Controls the output channel. configuration	■ 2.1 ■ 3.1 ■ 4.1 ■ 5.1 ■ 6.1 ■ 7.1 ■ Phantom 6.1 ■ Phantom 7.1	5.1
SDI x UpMix Latency	Specifies the latency profile of the up-mix.	Low Latency High Latency	Low Latency
SSDI x UpMix DICE Process Level	Specifies the amount of DICE processing to perform.	0 to 100	50
SDI x UpMix Depth	Specifies the amount of front-to-back bias to apply to the standard soundstage.	-100 to 100	0
SDI x UpMix Front Width	Specifies the amount of narrowing or widening to perform on the front channels.	-100 to 100	0
SDI x UpMix Surround Width	Specifies the wideness of the surround channels.	0 to 100	100
SDI x UpMix LFE Cut off	Specifies the low-pass cutoff frequency of the LFE channel in Hz	60 Hz to 140 Hz	80 Hz
SDI x UpMix Final Limiter Ceiling	Specifies the threshold where final limiting on the up-mixed output occurs.	-12 dBFS to 0 dBFS	0 dBFS
SDI x UpMix Preset Save	Select a preset to save custom settings to.	■ (select save) ■ Custom 1 ■ Custom 2 ■ Custom 3 ■ Custom 4 ■ Custom 5	(select save)

DTS Neural UpMix Presets

You can create and save up to 5 presets with your custom UpMix settings.

1 Go to the **UpMix Settings** screen.
2 Define your preferred settings.
3 Save your settings to one of the available custom presets (**Custom1**-**Custom5)** by selecting it from the **S<1-2> UpMix Preset Save** drop down.
4 To load settings that you saved to a custom preset, select it from the **S<1-2> UpMix Preset Recall** drop down. For example, select **Custom1** to load UpMix settings saved to that preset.

DTS Neural DownMix

The DTS Neural Surround DownMix enables 5.1 surround sound to be transported through any stereo infrastructure. The DownMix process is based upon the principle that both natural stereo and 5.1 content are two-dimensional; both contain width and depth spatial attributes.

The DTS Neural Surround DownMix can represent six channels of discrete audio sources in a stereo downmix by transforming the sources into pure intensity and coherence encoding. By correcting overlaps of the signal sources in intensity, time, coherence, polarity, and phase before the six channels are combined, the DTS Neural Surround DownMix accounts for the problems suffered in traditional matrix encode systems—such as comb filtering, spatial location distortion, etc.

The proprietary Neural Audio “watermark process” faithfully reproduces surround information when it is rendered by the DTS Neural Surround UpMix or any LtRt system. In brief, the DTS Neural Surround DownMix produces a stereo downmix that accurately represents the original content whether monitored in mono, stereo, matrix or DTS Neural 5.1 Surround Sound.

Figure 4-1 shows a DownMix taking a multi-channel audio source. The downmix creates two-channel audio source using the Neural Audio approach of embedding a watermark signal within the stereo audio signal patch. The watermark signal contains spatial and steering positioning information. The resulting stereo audio signal is also known as LwRw.

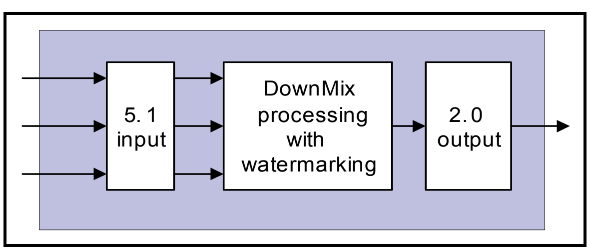


Figure 4-1 Neural Audio DownMix Block Diagram

DTS Neural DownMix Parameters

Table 4-2 DownMix parameters

SDI x > Audio Configuration > Advanced Audio Processing > DTS Neural DownMix			
Parameter Name	**Function**	**Options**	**Default**
SDI x DownMix Status	Current status of the DownMix processing block.	■ Uninitialized ■ Running ■ Failed-function bypassed ■ Failed-output lost	Uninitialized
SDI x DownMix L/R Encoding Mode	Specifies the encoding mode for the left and right channels.	Phantom Center Hard Center	Phantom Center
SDI x DownMix LFE Cut off	Specifies the low-pass cutoff frequency of the LFE channel in Hz.	60 Hz to 140 Hz	80 Hz
SDI x DownMix Final Limiter Ceiling	Specifies the threshold where final limiting on the down-mixed output occurs.	-20 dB to 0 dB	0 dB
SDI x DownMix Active Correction	Specifies correction to the DownMix ICLD, ICPD, and spectrum.	No Yes	No
SDI x DownMix Preset Save	Select a preset to save custom settings to.	■ Custom 1 ■ Custom 2 ■ Custom 3 ■ Custom 4 ■ Custom 5	(select save)

DTS Neural DownMix Presets

You can create and save upto 5 presets with your custom DownMix settings.

1 Go to the **DownMix Settings** screen.
2 Define your preferred settings.
3 Save your settings to one of the available custom presets (**Custom1**- **Custom5**) by selecting one from the **S<1-2> Downmix Preset Save** drop down.

To load settings that you saved to a custom preset, select it from the **S<1-2> DownMix Preset Recall** drop down. For example, select **Custom1** to load DownMix settings saved to that preset.

DTS Neural MultiMerge

The DTS Neural Surround MultiMerge enables broadcasters to transition from stereo to 5.1 surround sound, providing viewers with a 24/7 surround sound experience. With MultiMerge inline, 5.1 original content is passed unaffected to the viewer while original stereo content is upmixed to a 5.1 surround sound image. This provides the viewer with a consistent surround experience.

The transition between 5.1 and stereo occurs seamlessly without the need of operator intervention. By offering a 24/7 5.1 signal, AC3 metadata does not transition between 2/0 and 3/2 mode. This prevents audio clicks, pops, and dropouts. The process also avoids taking “artistic license” with content by placing audio exactly where it would be heard in a professional LEDE (Live End Dead End) listening environment. For example, mono or pan-pot stereo will image in front of the listener, whereas stereo containing depth information, or LtRt encoding, will surround the listener.

You can use MultiMerge in combination with the DTS Neural Surround DownMix device to pass 5.1 through stereo-only facilities and therefore eliminate the need for costly master control upgrades.

Figure 4-2 shows how the MultiMerge takes a two-channel audio source (stereo, matrix encoded stereo, LtRt or DTS Neural Surround LwRw) and renders a 5.1 multi-channel mix; in combination with taking original multi-channel content and creating a stereo downmixed signal, depending on the input configuration and content source used.

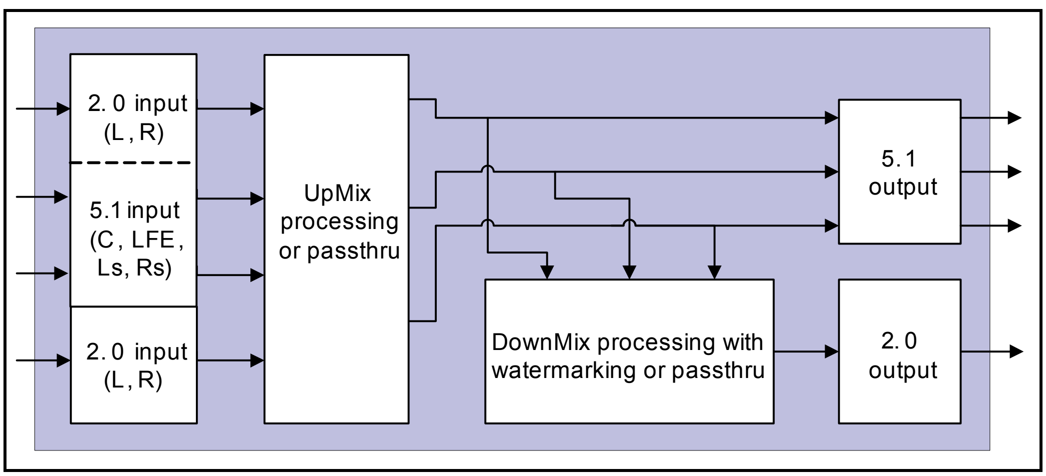


Figure 4-2 MultiMerge Block Diagram

DTS Neural MultiMerge Parameters

Table 4-3 MultiMerge parameters

SDI x > Audio Configuration > Advanced Audio Processing > DTS Neural MultiMerge			
Parameter Name	**Function**	**Options**	**Default**
SDI x MultiMerge Status	Current status of the MultiMerge processing block.	■ Uninitialized ■ Running ■ Failed-function bypassed ■ Failed-output lost	
SDI x MultiMerge Detected Input	Reports the current input(s) that are included in the output.	■ N/A ■ Mix ■ Multi ■ Stereo ■ Aux	Mix
SDI x MultiMerge Input Selection Mode	Specifies how input channels are selected. ■ **Mix** mixes the auxiliary 2.0 inputs into the multichannel 5.1 inputs. ■ **Multi** always uses the multichannel 5.1 inputs. ■ **Stereo** always uses the stereo L/R pair from the multichannel inputs. ■ **Aux** always uses the auxiliary 2.0 inputs. ■ **Multi Detect** uses multichannel 5.1 inputs if they are active. If not, the auxiliary 2.0 inputs are used. ■ **Stereo Detect** uses the stereo L/R pair from the multichannel 5.1 inputs if they are active. If not, the auxiliary 2.0 inputs are used. ■ **Aux Detect** uses the auxiliary 2.0 inputs if they are active. If not, the multichannel 5.1 inputs are used. ■ **Multi Aux Stereo** uses multichannel inputs if they are active, or uses the auxiliary inputs if they are active. If neither is active, it uses the stereo L/R pair from the multichannel inputs.	■ Mix ■ Multi ■ Stereo ■ Aux ■ Multi Detect ■ Stereo Detect ■ Aux Detect ■ Multi Aux Stereo	Multi
SDI x MultiMerge Input Noise Floor Threshold	Specifies the amount of signal required when detecting active inputs.	-80 dB to -50 dB	-60 dB
S<1-2> MultiMerge Input Cross Fade Time	Specifies the transition time between inputs when switching due to detected level changes.	50 ms to 750 ms	200 ms

Table 4-3 MultiMerge parameters

SDI x > Audio Configuration > Advanced Audio Processing > DTS Neural MultiMerge			
Parameter Name	**Function**	**Options**	**Default**
SDI x MultiMerge Mode	Current status of the MultiMerge mode. ■ **Auto** determines if content is stereo (2.0) or surround (5.1), and adapts to the correct UpMix/Passthrough mode for consistent 5.1 output. When in Auto, the Detect Threshold parameter controls the noise floor level for the detection. Any content above this threshold on the surround channel inputs 3-6 (C, LFE, Ls, Rs) is considered surround and MultiMerge will be in passthrough mode. ■ **Passthrough** forces MultiMerge to always pass through 5.1 content to the 5.1 outputs untouched, while creating a downmix for the Aux output. ■ **UpMix** forces MultiMerge to always upmix stereo content on both of the Left/Right 5.1 inputs and the Aux 2.0 inputs.	Auto Passthrough Upmix	Auto
SDI x MultiMerge Latency	Specifies the latency profile of the MultiMerge.	Low Latency High Latency	Low Latency
SDI x MultiMerge Noise Floor Threshold	Specifies the signal level that must be detected on any.	-80 dB to -50 dB	-60 dB
SDI x MultiMerge Cross Fade Time	Specifies the transition time between up-mix and passthrough when in auto mode.	50 ms to 750 ms	200 ms
SDI x MultiMerge UpMix Depth	Specifies the amount of front-to-back bias to apply to the standard soundstage.	-100 to 100	0
SDI x MultiMerge UpMix Front Width	Specifies the amount of narrowing or widening to perform on the front channels.	-100 to 100	0
SDI x MultiMerge UpMix Surround Width	Specifies the wideness of the surround channels.	0 to 100	100
SDI x MultiMerge UpMix LFE Cut off	The cutoff frequency for the generated LFE channel.	60 Hz to 140 Hz	80 Hz
SDI x MultiMerge DownMix L/R Encoding Mode	Specifies the encoding mode for the left and right channels.	Phantom Center Hard Center	Phantom Center
SDI x MultiMerge DownMix LFE Cut off	The cutoff frequency of the input LFE channel.	60 Hz to 140 Hz	80 Hz

Table 4-3 MultiMerge parameters

SDI x > Audio Configuration > Advanced Audio Processing > DTS Neural MultiMerge			
Parameter Name	**Function**	**Options**	**Default**
SDI x Multi Final Limiter Ceiling	Specifies the threshold where final limiting on the surround (5.1) output occurs.	-20 dBFS to 0 dBFS	0
SDI x Stereo Final Limiter Ceiling	Specifies the threshold where final limiting on the auxiliary stereo (2.0) output occurs.	-20 dBFS to 0 dBFS	0
SDI x MultiMerge Preset Save	Select a preset to save custom settings to.	■ Custom 1 ■ Custom 2 ■ Custom 3 ■ Custom 4 ■ Custom 5	(select save)

DTS Neural MultiMerge Presets

You can create and save upto 5 presets with your custom MultiMerge settings.

1. Go to the **MultiMerge Settings** screen
2. Define your preferred settings.
3. Save your settings to one of the available custom presets (**Custom1**- **Custom5**) by selecting one from the **S<1-2> MultiMerge Preset Save** drop down.

To load settings that you saved to a custom preset, select it from the **S<1-2> MultiMerge Preset Recall** drop down. For example, select **Custom1** to load MultiMerge settings saved to that preset.

DTS Neural Loudness Control

DTS Neural Loudness Control options manage loudness levels within a specific desired volume range. Advanced psychoacoustic and signal processing techniques detect and regulate the perceived loudness of stereo and 5.1 sources, for example to maintain audio perceived loudness between programming and commercials.

Neural Audio’s perceptual loudness measurement tool treats each audio channel (L, R, C, LFE, Ls, and/or Rs) as a separate mono channel. The tool accounts for spectral and density differences and temporal overlaps in modelling how the human ear perceives the loudness of the audio content. DTS Neural loudness measurement accommodates both stereo and multi-channel audio equally well.

After measurement, DTS Neural Loudness Control applies gain or attenuation to achieve the target loudness level (Dial Norm) while preserving the spectral balance of the original signal. It adapts the frequency response of the low and high frequencies to compensate for level differences within the original signal. You can use DTS Neural Loudness Control in the following roles:

- Protection—only affecting content that falls aggressively outside the desired target

- Management—tightly controlling loudness to guarantee intelligibility without the distracting side effects of traditional volume management solutions

Note: *Because of the single function nature of APM6803**+** modules, default settings provide an optimal configuration for the normal audio situation.*

Also see, ***How Loudness Control works*** on page 34.

DTS Neural Loudness Control Parameters

Table 4-4 Loudness Control Parameters

SDI x > Audio Configuration > Advanced Audio Processing > DTS Neural Loudness Control x			
Parameter Name	**Function**	**Options**	**Default**
SDI x Loudness Control <1-4> Function	Indicates whether or not to bypass the loudness control algorithm and this CODEC will behave as a delay only.	■ Always On ■ Bypass ■ On Alarm Only	Always On
SDI x Loudness Control <1-4> Function Feedback	Indicates whether or not bypass the loudness control algorithm and this CODEC will behave as a delay only.	■ Enabled ■ Bypassed	Enabled
SDI x Loudness Control <1-4> Status	Current status of the Loudness Control processing block.	■ Uninitialized ■ Running ■ Failed-function bypassed ■ Failed-output lost	Uninitialized
SDI x LC <1-4> Loudness Measurement Type	Specifies the type of loudness measurement to make prior to performing loudness control.	■ NLM ■ LEQ 1770	LEQ 1770
SDI x LC <1-4> Target Loudness Level	Specifies the Loudness control target level.	■ -40 dB Eq to 0 dB Eq	-27 dB Eq
SDI x LC <1-4> Ratio	Specifies the Loudness control Ratio.	0.00 to 1.00	0.98
SDI x LC <1-4>Upper Threshold	Specifies the Loudness control Upper Threshold	0 dB to 20 dB	0 dB
SDI x LC <1-4> Lower Threshold	Specifies the Loudness control Lower Threshold	-20 dB to 0 dB	0 dB
SDI x LC <1-4> Upper Alarm Threshold	Specifies the Loudness control Upper Alarm Threshold.	0 dB to 20 dB	0 dB
SDI x LC <1-4> Lower Alarm Threshold	Specifies the Loudness control Lower Alarm Threshold.	-20 dB to 0 dB	0 dB
SDI x LC <1-4> Upper Warning Threshold	Specifies the Loudness control Upper warning Threshold.	0 to 20 dB	0 dB
SDI x LC <1-4> Lower Warning Threshold	Specifies the Loudness control Lower warning Threshold.	-20 to 0 dB	-20 dB
SDI x LC <1-4>Freeze Window	Specifies the Loudness control Freeze Window.	0.0 to 10.0	2.0

Table 4-4 Loudness Control Parameters

SDI x > Audio Configuration > Advanced Audio Processing > DTS Neural Loudness Control x			
Parameter Name	**Function**	**Options**	**Default**
SDI x LC x Quiet Threshold	Specifies the Loudness control Noise Floor.	-80 dB Eq to -20 dB Eq	-55 dB Eq
SDI x LC x Attack Time	Specifies the Loudness control Attack Time.	5 ms to 150 ms	50 ms
SDI x LC x Release Time	Specifies the Loudness control Release Time.	20 ms to 500 ms	150 ms
SDI x LC x Compressor Threshold	Specifies the Loudness control Compressor Threshold.	0 dB to 16 dB	5 dB
SDI x LC x Compressor Ratio	Specifies the Loudness control Compressor Ratio.	0.00 to 1.00	0.50
SDI x LC x Shaping	Specifies the amount of loudness shaping desired. A value of zero is no loudness shaping.	0 to 10	0
SDI x LC x Final Limiter Ceiling	Specifies the Loudness control Output Limiter Threshold.	-20 dBFS to 0 dBFS	0 dBFS
SDI x LC x Metering	Activates real-time meter values in the status output.	■ Disabled ■ Enabled	Enabled
SDI x LC x Run Final Limiters	Indicates whether or not final limiters be applied according to the value of FinalLimiterCeiling_dBFS.	■ No ■ Yes	
SDI x LC x Preset Save	Selects to save the current LC setting.	■ (select save) ■ Custom1 ■ Custom2 ■ Custom3 ■ Custom4 ■ Custom5	(select save)
SDI x LC x Current Average Input Loudness	The smoothed input average loudness measurement suitable for metering.	■ -60 dBEq to 20 dBEq	0 dBEq
SDI x LC x Current Input Loudness	The smoothed input loudness measurement suitable for metering.	■ -60 dBEq to 20 dBEq	0 dBEq
SDI x LC x Current Input Peak	The input peak measurement.	■ -60 dBFS to 20 dBFS	0 dBFS
SDI x LC x Current Output Loudness	The smoothed output loudness measurement suitable for metering.	■ -60 dBEq to 20 dBEq	0 dBEq
SDI x LC x Current Output Peak	The output peak measurement.	■ -60 dBFS to 20 dBFS	0 dBFS
SDI x LC x Current Correction	The amount of correction actively being applied.	■ -60 dB to 60 dB	0 dB
SDI x LC x Current Compression	The amount of compression actively being applied.	■ -60 dB to 60 dB	0 dB

How Loudness Control works

The DAPM6802+ allows you to define virtual streams and Loudness Control can be enabled or disabled for a stream when defining the virtual output. Once enabled, Loudness Control can be configured as follows:

1. Go to the relevant output channel on which you want to enable loudness control **Parameters > SDI x > Audio Configuration > Advanced Audio Processing > Loudness Control 1** to **4.**
2. Set the **LC Control Function** to
   - Always On: LC correction is always enabled (depending on the incoming audio level and "Loudness Upper Threshold" and "Loudness Lower Threshold" settings).
   - Bypassed: LC correction is never applied and is only used for monitoring input streams.
   - On Alarm Only: LC correction gets enabled when the alarm "LC Alarm Level" is triggered. Otherwise, LC correction is bypassed.
3. Set the **Loudness Measurement Type** to either **LEQ1770** or **NLM**.
4. Define a **Target Loudness Level Source** (between -40 and 0 dBEq).
5. Set the **Loudness Control Ratio**.

   This is the amount of gain or attenuation applied when loudness differences are measured. When set to 1.00, 1 dB of gain/attenuation will be applied for every dB of loudness difference between the input signal and target level.
6. Set the **Loudness Upper Threshold**.

   This parameter represents the allowable distance the input signal can range above the Target Level before attenuation. If an input signal falls between the Upper and Lower Threshold in reference to the Target Level, no correction will be applied. For instance, if Target Level is -20 dBFS, Upper Threshold is 12 dBFS, anything over -8 dBFS will have attenuation applied.
7. Similarly, set the **Loudness Lower Threshold**.
8. Set the **Upper Alarm Threshold**

   If the long-term loudness average level "In Average Loudness" falls between the Upper and Lower Alarm thresholds in reference to the Target Level, the alarm "LC Alarm Level" is not triggered. Otherwise, the alarm is triggered (also affected by the trigger time) and it automatically controls the loudness control working mode if "LC Control Mode" is set as "On Alarm Only". (Also see **Loudness Protection**)
9. Similarly, set the **Lower Alarm Threshold**
10. Set the **Upper Warning Threshold** (0-20).

    This is the allowable (relative) distance between the Target Level and Upper Threshold and this range will not trigger a warning alarm. For instance, if Target Level is 20 dBFS, Upper Threshold is 12 dBFS, and Upper Warning Threshold is 6 dBFS - anything over 32 dBFS will have attenuation applied; anything between 26-32 dBFS will trigger an alarm, but no correction will be applied. (Also see **Loudness Protection**)
11. Similarly, set the **Lower Warning Threshold**.
12. Set the **Freeze Window**

    This is a window size between 0 and 10 dB where small loudness differences are allowed. Anything outside this window has gain/attenuation applied.
13. Set the **Attack time** to control how quickly processing responds to sharp onsets in loudness levels.
14. Set the **Release Time** to control how quickly processing responds to sharp drops in loudness levels.
15. Set the **Compressor Threshold** to detect and allow short term loudness peaks.

16 Set the **Compressor Ratio** to control the attenuation applied to short peaks that exceed the Compressor Threshold.

17 Set the **Loudness Shaping**.

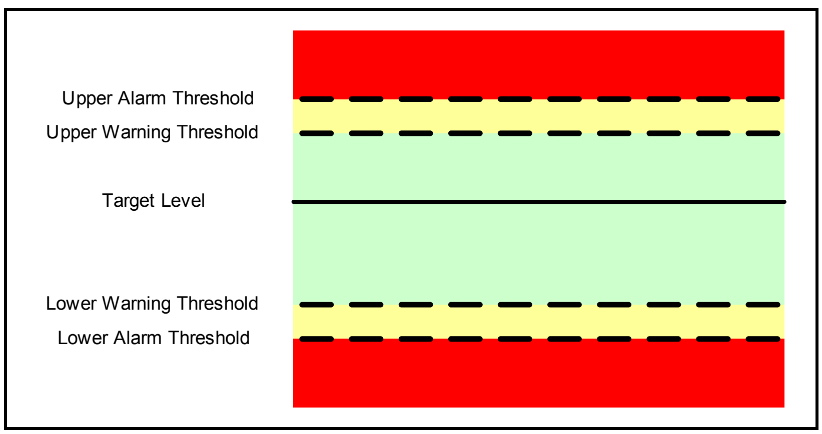
18 Set the **Output Limiter Threshold**.

19 Set the **Metering**.

Loudness Protection

The DAPM6802+ can provide certain warnings and alarms to indicate to the user that the incoming loudness exceeds pre-defined thresholds. This is particularly useful when using GPIs and automation to change the loudness control profile dynamically. The idea is to warn the user that the incoming loudness has been consistently low or high for an extended period, and that enabling loudness correction may be desired.

The first step is to average the incoming loudness on any Virtual Output that has Loudness Control enabled. The user specifies an averaging window size, and the DAPM6802+ will use it to average the loudness of the audio going into the Loudness Control block. This average is then sent to a series of alarms that have trigger and clear times. Two sets are provided, one to act as a warning, and the other to act as an alarm condition. The alarm condition has the added benefit that it can also trigger a loudness control preset.



By setting appropriate warning and alarm upper and lower thresholds, the APM6803+ can provide additional information to help protect against excessive loudness when very light or no real time correction is being applied. In addition, it can take action automatically and turn on a more aggressive loudness correction profile. Care must be taken to ensure the clear time in this case is longer than the trigger time in order to create hysteresis in the system, such that the system does not continuously enable and disable real time loudness correction. The clear time should provide sufficient time for a user to investigate and either keep the loudness correction profile or manually revert to another profile.

Bypass Audio Processing and Routing

To check the integrity of an input signal by letting it pass through the device (without processing), set the parameters as follows:

Parameters > SDI x > Audio Configuration > SDI x ADS Clean = No

Parameters > SDI x > Audio Configuration > Output Router > SDI x Audio Grp X Pari X Embed Control = Disabled

5 Custom Scripting

Overview

The DAPM6802+ provides custom scripting functionality for finer control over certain operations.

- **Common Scripting Guidelines**
- **Custom GPI Scripts**
- **Parameter Control Scripts**

Common Scripting Guidelines

These scripting guidelines are common to both GPI (input and output) scripts and parameter scripts. Note these requirements when writing your custom scripts.

Statements

- Scripts are created from several statements.
- Each statement consists of one condition and several assignments.
- In each statement, if the condition is satisfied, the assignments will take effect.
- The number of characters (including spaces) in one command line is limited to 251

Note: *If the script is greater than 251 characters, the module will automatically truncate it without warning.*

IF Condition

- The number of **IF** command combinations is limited to 30
- The number of condition combinations in each **IF** command is limited to 10

IF [condition] **THEN** [assignment] [assignment]...

Comparisons

- A condition is created from one or more comparisons.
- Comparisons can be **ANDed** together using **&&** and **ORed** together using **||** to form a condition.
- The AND operation always has precedence over the OR operation when AND and OR both exist in a condition.
- The following comparison operators can be used:
  - == equal
  - > greater than
  - < less than
  - >= greater or equal than
  - <= less than or equal than
  - != not equal

Notes

- Physical GPIs are labeled using natural numbers, from **GPI In 1** to **GPI In 8**, and **GPI Out 1** to **GPI Out 4**. However, the IDs of GPIs used in scripting start from **0**.

  For instance, **GPI In 1** is called **GPI0** in the script and **GPI Out 4** is called **GPO3**.
- Refer the DAPM6802+ parameter list for parameter IDs used in scripting.

Error Diagnosis

Whenever scripting is used, ensure the status feedback (Custom Input Status/Custom Output Status/Script Status) is **Active**.

- If Parameter scripting is in use, ensure the parameter **Activate Script** (Parameters > General > Parameter Control Script) is set to **Enabled**.
- If GPI scripting is in use, ensure the targeted GPI ports are configured as **Custom GPI**. (Parameters > General > GPI Input/Output)

Keywords

Keywords for the scripting parser are limited to the following and must be in all caps:

IF, THEN, PARAM[,], =, (,), ==, &&, ||, >=, <=, ==, !=, >, <,
0, 1, 2, 3, 4, 5, 6, 7, 8, 9,

The following keywords are only acceptable in GPI scripts.

GPI0, GPI1, GPI2, GPI3, GPI4, GPI5, GPI6, and GPI7
GPO0, GPO1, GPO2, and GPO3

Error Reporting Rules

In case of script errors, the index number where the error is detected is reported. The index starts from 0 for the first character of the first illegal keyword.

Examples

The following is an example of correct syntax:

IF GPI1==0 THEN PARAM[34]=0

The table below illustrates errors in syntax and how errors are reported.

Table 5-1 Error Reporting

Incorrect Command	Status	Comments
IF GPI1=0 THEN PARAM[34]=0	Error:7	“==” must be used in comparison commands. “=” is incorrect.
IF GPI1==0 THEN PARAM[34]==0	Error:28	“=” must be used in assignment commands. “==” is incorrect.
IF GPI1==0 THEN PARAM [34]=0	Error:16	There must be no space between “PARAM” and “[“
IF GPI1==0 THEN pARAM[34]=0	Error:16	PARAM must be in all caps. pARAM is incorrect.
IF GpI1==0 THEN PARRAM[34]=0	Error:3	There are 2 errors in this command - GpI is not in all caps and PARRAM has been misspelled. However, only the index of the first error found is reported.

Custom GPI Scripts

The DAPM6802+ supports custom GPI scripts to enable you to configure GPIs. Refer **Common Scripting Guidelines** for more details on writing your GPI script.

GPI scripting is available by going to:
Parameters > General > Custom GPI

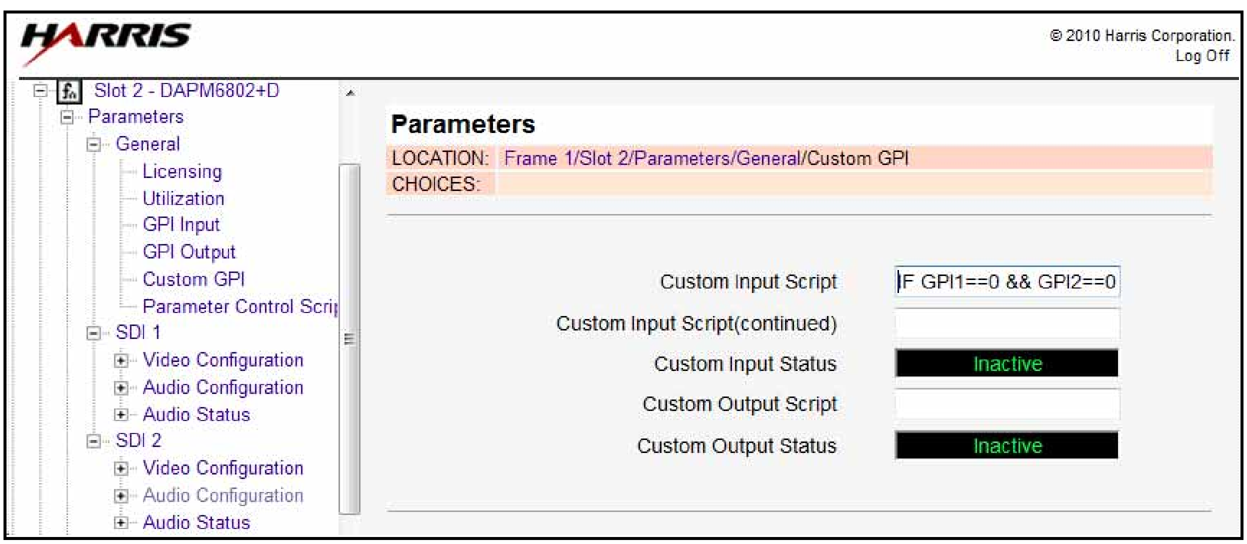


Figure 5-1 Custom GPI Script Settings

Setting up Custom GPI

1. In the Parameter tree, go to **General** > **GPI Input**.
2. Set **GPI Input To Edit** to **Input x**
3. Set **Input Trigger** to any option except Off (see Note below). For instance, set it to **Rising Edge**
4. Set **Input Event** to **Custom GPI**
5. In the Parameter tree, go to **General** > **Custom GPI**
6. Enter the GPI Input script in the **Custom Input Script** text box.

   If necessary, the script will be continued in **Custom Input Script (continued)** text box.

Note: The *script in **Custom Input Script** and **Custom Input Script(continued)** should not conflict logically. Otherwise, the script in the **Custom Input Script (continued)** box has priority over the script in **Custom Input Script** box.*

7. Enter the GPI Output script in the **Custom Output Script** text box.

Note: *When you select a Custom GPI input or output, the Trigger control in GPI Input is disabled for that specific GPI.*

Comparisons

In a **GPI input script**, the comparison could be either GPIx == 0 or GPIx == 1, where x is in the range of [0, (number of GPI inputs - 1)], and 0 represents low and 1 represents high.

In a GPI output script, the comparison is in the format of PARAM[x]==y, where x is the ID of a device parameter and y is a value for that parameter.

Assignments

- In a GPI input script, an assignment is written as:
  PARAM[x]=y
  where x is the ID of a device parameter and y is a value for that parameter.

- In a GPI output script, an assignment is written as:
  GPOx=0 or GPOx=1
  where x is the range [0, number of GPI outputs -1] and 0 represents low and 1 represents high.

Custom GPI Example

The following example indicates how to use the combination of GPI In 2 and GPI In 3 to change audio output sources, and trigger GPI Outputs based on the source of **SDI_A Out 1 Surround In**.

Table 5-2 Custom GPI Example Inputs

Action		Result	
GPI Inputs		**Parameters**	
12 (GPI1)	I3 (GPI2)	S1 Out 1 Surround In (PARAM [346])	S1 Out 1 Stereo In (PARAM [347])
0	0	Virtual In 1 Surround 5.1	Virtual In 1 Stereo
0	1	Virtual In 4 Surround 5.1	Virtual In 4 Stereo
1	0	AAP Surround 5.1	AAP Aux
1	1	Virtual In 2 Surround 5.1	Virtual In 2 Stereo

Table 5-3 Custom GPI Example Outputs

Action	Result			
Parameters	**GPI Outputs**			
SDI_A Out 1 Surround In (PARAM [346])	O1 (GPO0)	O2 (GPO1)	O3 (GPO2)	O4 (GPO3)
Virtual In 1 Surround 5.1	1	0	0	0
Virtual In 4 Surround 5.1	0	1	0	0
AAP Surround 5.1	0	0	1	0
Virtual In 2 Surround 5.1	0	0	0	1

Pre-Requisites

Make the following configuration settings:

- Go to **SDI_A > Audio Configuration** > **Input Configuration**
  - Set **Virtual In 1** > **SDI_A In 1 Virtual Type** to **Surround-5.1+2.0**
  - Set **Virtual In 2** > **SDI_A In 2 Virtual Type** to **Surround-5.1+2.0**
  - Set **Virtual In 4** > **SDI_A In 4 Virtual Type** to **Surround-5.1+2.0**
- Go to **SDI_A > Audio Configuration** > **Advanced Audio Processing**
  - Set **SDI_A Processing Mode to M3: MultiMerge [Credit4]**
- Go to **SDI_A > Audio Configuration** > **Output Configuration**
  - Set **Virtual Out 1** > **SDI_A Out 1 Virtual Type** = **Surround-5.1+2.0**
  - Set **Virtual Out 1** > **Source Select > SDI_A Out 1 Virtual Source Routing Type = Group**
- Go to **General** > **GPI Input**
  - Set **GPI Input to Edit** to **Input 2**
  - Set **Input Trigger** to **RisingEdge**
  - Set **Input Event** to **Custom GPI**
  - Repeat the previous steps to configure **Input 3**

- Go to **General** > **GPI Output**
    - Set **GPI Output to Edit** to **Output 1**
    - Set **Output Trigger** to **Active High**
    - Set **Output Event** to **Custom GPI**
    - Repeat the above steps to configure **Output2**, **Output3**, and **Output4**

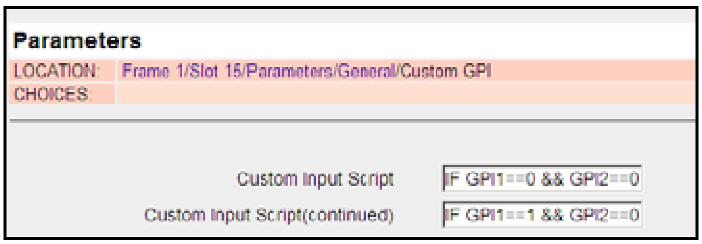
Custom GPI Input Script

Enter the following in the **Custom Input Script** text box:

IF GPI1==0 && GPI2==0 THEN PARAM[346]=0 PARAM[347]=0 IF GPI1==0 && GPI2==1 THEN PARAM[346]=3 PARAM[347]=3

Enter the following in the **Custom Input Script (continued)** text box:

IF GPI1==1 && GPI2==0 THEN PARAM[346]=4 PARAM[347]=9 IF GPI1==1 && GPI2==1 THEN PARAM[346]=1 PARAM[347]=1



Custom GPI Output Script

Enter the following in the Custom Output Script text box:

IF PARAM[346]==0 THEN GPO0=1 GPO1=0 GPO2=0 GPO3=0 IF PARAM[346]==3 THEN GPO0=0 GPO1=1 GPO2=0 GPO3=0 IF PARAM[346]==4 THEN GPO0=0 GPO1=0 GPO2=1 GPO3=0 IF PARAM[346]==1 THEN GPO0=0 GPO1=0 GPO2=0 GPO3=1

Parameter Control Scripts

Smart scripting control over parameters enables you to control certain operations programmatically. Set **Activate Script** as **Enabled** whenever you want to use scripting control.

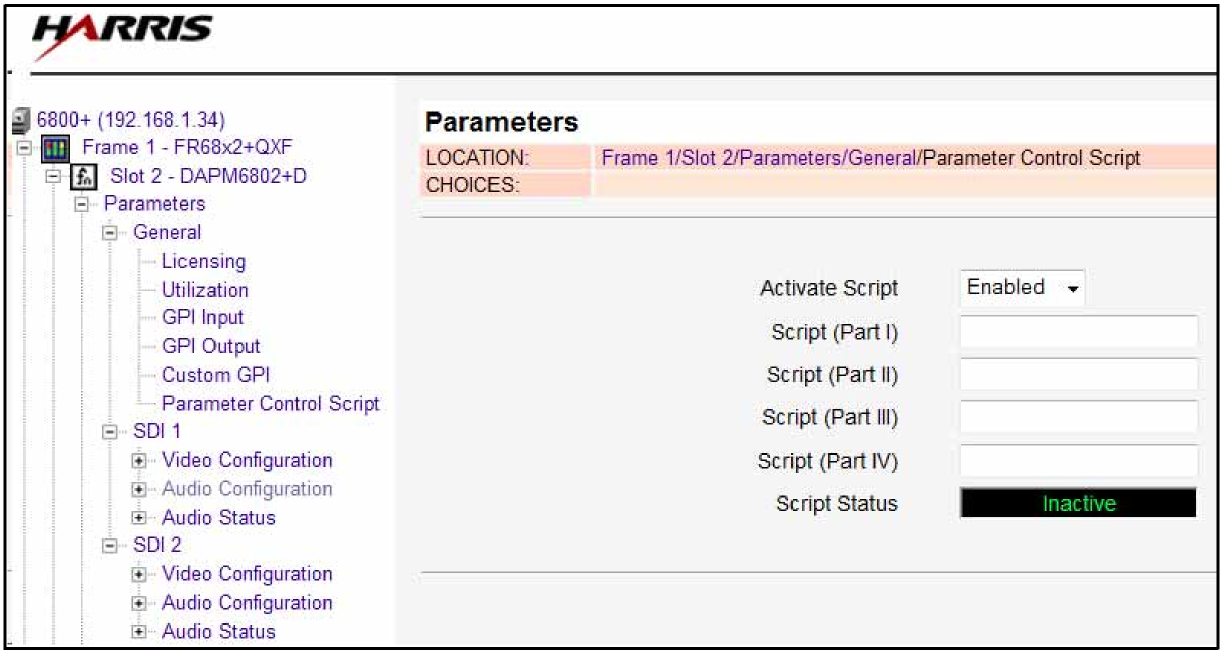


Figure 5-2 Enabled Activate Script

Parameter Control Example

Description

In this example, the module detects if SDI_A embedded audio group 1 is present.

- If the embedded audio is present, the module routes **Virtual In 1** to **Virtual Out 1**.
- If the embedded audio is not present, it routes **Virtual In 4** to **Virtual Out 1**.

Pre-Requisites

- Set **SDI_A > Audio Configuration > Input Configuration > Virtual In 1 > SDI_A In 1 Virtual Type** to **Surround-5.1+2.0**
- Set **SDI _A > Audio Configuration > Input Configuration > Virtual In 4 > SDI_A In 4 Virtual Type** to **Surround-5.1+2.0**
- Set **SDI_A > Audio Configuration > Output Configuration > Virtual Out 1 > SDI_A Out 1 Virtual Type** to **Surround-5.1+2.0**
- Set **Virtual Out 1 > Source Select > SDI_A Out 1 Virtual Source Routing Type = Group**

Parameter Control Script

- Go to **Parameters > General > Parameter Control Script**
- Enter the following into the **Script (Part 1)** textbox:
- **IF PARAM[57]==1 THEN PARAM[346]=0 IF PARAM[57]!=1 THEN PARAM[346]=3**

6 Specifications

Video Input

Table 6-1 Video Input Specifications

Item	1.5G HD-SDI Specification*	SD-SDI Specification*
Number of Inputs	2	2
Standard	■ 1080i/p (SMPTE 274M) ■ 720p (SMPTE 296M)	SMPTE 259M-C (270Mb/s, 525/625 component)
Connector	BNC (IEC 169-8)	BNC (IEC 169-8)
Impedance	75Ω	75Ω
Frame Rate	■ 1080p: 23.98, 24, 25, 29.97, 30Hz ■ 1080psf: 23.98, 24 Hz ■ 1080i: 50, 59.94, 60 Hz ■ 720p: 50, 59.94, 60 Hz	■ 525: 59.94 Hz ■ 625: 50 Hz
Return Loss	>15 dB from 5 MHz to 1485 MHz	>15 dB up to 270 MHz
Equalization	Up to 120 m (393 ft) for Belden 1694A	Up to 280 m (918 ft) for Belden 1694A and 8281
*Standard and relay back modules		

Video Output

Table 6-2 Video Output Specifications

Item	**1.5G HD-SDI Specification***	**SD-SDI Specification***
Number of Outputs	4	4
Standard	■ 1080i (SMPTE 274M) ■ 720p (SMPTE 296M) ■ SMPTE 292M with SMPTE299M embedded audio	■ SMPTE 259M-C,270 Mb/s, 525/625 component
Frame Rate	■ 1080i: 50, 59.94, 60 ■ 1080p: 23.98 (p/psf), 24 (p/psf), 25, 29.97, 30 Hz ■ 720p: 50, 59.94, 60 Hz	525, 625
Connector	BNC (IEC 169-8)	BNC (IEC 169-8)
Impedance	75Ω	75Ω
Return Loss	>15 dB, typical, from 5 MHz to 1485 MHz	>15 dB up to 270 MHz
Signal Level	800 mV ± 10%	800 mV ± 10%
D.C. Offset	0.0 V ± 0.5 V	0.0 V ± 0.5 V
Rise and Fall Time	<270 ps (20% to 80%)	400 to 1500 ps (20% to 80%)
Overshoot	<10% of amplitude (all outputs terminated)	<10% of amplitude (all outputs terminated)
Timing Jitter	■ HD-SDI: <1 UI (pk-to-pk)	■ SD-SDI: < 0.2 UI (pk-to-pk)
Alignment Jitter	■ >100 kHz: <0.2 UI (135 ps) pk-to-pk ■ >10 Hz: <1 UI (675 ps) pk-to-pk	■ <0.2 UI (740 ps) pk-to-pk
*Standard and relay back modules		

GPI Inputs and Outputs

Table 6-3 GPI Inputs and Output Specifications

Item	Specification
Inputs	
Number of Inputs	8
Connector	SAMTEC mini mate header
Trigger Action	RisingEdge or FallingEdge, configurable
Internal Pull-Up	+5 V
Baud Rate	<10 Kbps
Outputs	
Number of Outputs	4
Signal Standard	TTL Active low or high
Connector	SAMTEC mini mate header
Baud Rate	<10 Kbps

Propagation Delays

Table 6-4 Audio and Video Propagation Delays

AAP Mode	Audio Delay	Video Delay	
		Auto Match A/V Delay Enabled	**Auto Match A/V Delay Disabled**
UpMix With Low Latency	64.20 ms	64.20 ms	▪ HD-SDI (1080i, 1080p, 720p-50/59.94): approx 1.60 ms ▪ SD-SD (525/625): approx.1.80 ms
UpMix With High Latency	85.53 ms	85.53 ms	
DownMix	48.20 ms	48.20 ms	
MultiMerge With Low Latency	64.20 ms	64.20 ms	
MultiMerge With High Latency	85.53 ms	85.53 ms	
Loudness Control	96.20 ms	96.20 ms	
UpMix With Low Latency + Loudness Control	112.20 ms	112.20 ms	
UpMix With High Latency + Loudness Control	133.53 ms	133.53 ms	
Downmix + Loudness Control	96.20 ms	96.20 ms	
MultiMerge With Low Latency + Loudness Control	112.20 ms	112.20 ms	
MultiMerge With High Latency + Loudness Control	133.53 ms	133.53 ms	

Power Consumption

Table 6-5 Power Consumption Specification

Module	**Power Consumption**
DAPM6802+D or DAPM6802+RLYD	10.9W

Start-Up Time

The start-up time for the DAPM6802+ module is 12 to 14 seconds.

A Audio Bit Manipulation

Overview

The tables in this appendix contain information on the manipulation of bits that occur when using APM6802+ modules.

The following items are documented:

- ***Manipulating Channel Status Bits (C-Bit)*** on page 50
- ***Manipulating Validity and User Bits (V-Bit and U-Bit)*** on page 52
- ***Identifying Audio Characteristics (Audio Sampling Frequency and Word Length)*** on page 52

Table A-1 Description of Short Forms in the Appendix

RX Key	TX Key	Sample Rate Indication	Audio Word Length
N=Not recognized	N=Not transmitted	Byte 0 Bits [6,7], Byte 4 Bits [3,4,5,6] ■ [00,0100] = 96 kHz ■ [01,0000] = 48 kHz ■ [11,0000] = 32 kHz ■ [00,1000] = 24 kHz ■ [00,0101] = 88.2 kHz ■ [10,0000] = 44.1 kHz ■ [00,1001] = 22.05 kHz	Byte 2 Bits [0,1,2] ■ [000] = Maximum word length 20 bits (auxiliary bit use not indicated) ■ [001] = Maximum word length 24 bits (auxiliary bits used for audio) Byte 2 Bits [3,4,5] ■ [100] = Encoded word length =Maximum word length–4 bits ■ [101] = Encoded word length =Maximum word length–0 bits
Y=Recognized	Y=Transmitted		
S=Recognized and stored, passed-through, or both			

Manipulating Channel Status Bits (C-Bit)

Table A-2 Channel Status (C-Bits) Data Description

Byte	Bit	Function	RX	TX	Remarks
0	0	[0] Consumer Use [1] Professional Use	N Y	N Y	■ RX ignores bit ■ TX sets bit to 1
0	1	[0] Audio [1] Non-Audio	S S	Y Y	■ RX sets up audio channel to pass data (Gain=0 dB, Invert=off) ■ TX bit passed unmodified or forced, according to **Output Chxx Format** and **Out Chxx Format Fb** parameters
0	2 to 4	[000] Not Indicated [100] No Emphasis [110] 50/15 µs [111] CCITTJ17	S S S S	Y Y Y Y	■ RX ignores bits ■ TX passed bits unmodified
0	5	[0] Locked [1] Unlocked	N N	Y N	■ RX ignores bits ■ TX sets bit to [0]
0	6 to 7	[00] Not indicated [01] 48 kHz [10] 44.1 kHz [11] 32 kHz	Y Y Y Y	N Y N N	■ RX ignores bits ■ TX sets bits to [01]
1	0 to 3	[0000] Not indicated [0001] Two channel [0010] Mono [0011] Prim/sec [0100] Stereo [0101] to [1111] Undefined	N N N N N N	Y N N N N N	■ RX ignores bits ■ TX sets bits to [0000]
1	4 to 7	[0000] Not indicated [0001] 192 bit block [0010] AES18 (HDLC) [0011] User defined [0100] to [1111] Undefined	S S S S S	Y Y Y Y Y	■ RX ignores bits ■ TX passes bits unmodified
2	0 to 2	[000] Aux. bit use is not indicated [001] Aux. bit use is audio data [010] Aux. bit use is co-ordination signal [011] to [111] Undefined	N N N N	Y Y N N	■ RX ignores bits ■ TX sets bits according to transmitted word length
2	3 to 5	[000] Not indicated [001] Max Length–1 [010] Max Length–2 [011] Max Length–3 [100] Max Length–4 [101] Max Length [110] to [111] Undefined	N N N N N N N	N N N N Y Y N	■ RX ignores bits ■ TX sets bits according to transmitted word length

Table A-2 Channel Status (C-Bits) Data Description

Byte	Bit	Function	RX	TX	Remarks
2	6 to 7	[00] Alignment level not indicated [01] Alignment to SMPTE RP155 [10] Alignment to EBU R68 [11] Reserved	N	Y	▪ RX ignores bits ▪ TX sets bits to [00]
3	0 to 6	bit 7 = 0: Channel number bit 7 = 1: [0,1,2,3] Channel number [4,5,6] Multi-channel mode	N	Y	▪ RX ignores bits ▪ TX sets bits to [0000000]
3	7	[0] Undefined multi-channel mode [1] Defined multi-channel mode	N	Y	▪ RX ignores bits ▪ TX sets bits to [0]
4	0 to 1	[00] Not a reference [01] Grade 1 reference [10] Grade 2 reference [11] Undefined	N N N N	Y N N N	▪ RX ignores bits ▪ TX sets bits to [00]
4	2	Reserved	N	Y	▪ RX ignores bits ▪ TX sets bits to [0]
4	3 to 6	[0000] Not indicated [1000] 24 kHz [0100] 96 kHz [1100] 192 kHz [1001] 22.05 kHz [0101] 88.2 kHz [1101] 176.4 Hz [1111] User defined	N	Y	▪ RX ignores bits ▪ TX sets bits to [0000]
4	7	[0] Sample frequency not scaled [1] Sample frequency scaled by 1/ 1.001	N	Y	▪ RX ignores bits ▪ TX sets bits to [0]
5	0 to 7	Reserved	N	Y	▪ RX ignores bits ▪ TX sets bits to [00000000]
6 to 9	0 to 7	Alphanumeric channel origin data	S	Y	▪ RX ignores bits ▪ TX passed bits unmodified
10 to 13	0 to 7	Alphanumeric channel destination data	S	Y	▪ RX ignores bits ▪ TX passed bits unmodified
14 to 17	0 to 7	Local sample address code	S	Y	▪ RX ignores bits ▪ TX passed bits unmodified
18 to 21	0 to 7	Time-of-day sample address code	S	Y	▪ RX ignores bits ▪ TX passed bits unmodified
22	0 to 3	Reserved	N	Y	▪ RX ignores bits ▪ TX passed bits unmodified
22	4	Bytes 0 to 5 reliability flag	N	Y	▪ RX ignores bits ▪ TX passed bits unmodified

Table A-2 Channel Status (C-Bits) Data Description

Byte	Bit	Function	RX	TX	Remarks
22	5	Bytes 6 to 13 reliability flag	S	Y	■ RX ignores bits ■ TX passed bits unmodified
22	6	Bytes 14 to 17 reliability flag	S	Y	■ RX ignores bits ■ TX passed bits unmodified
22	7	Bytes 17 to 21 reliability flag	S	Y	■ RX ignores bits ■ TX passed bits unmodified
23	0 to 7	CRC	Y	Y	■ RX ignores bits ■ TX calculates CRC on output

Manipulating Validity and User Bits (V-Bit and U-Bit)

Table A-3 V-Bit and U-Bit Data Descriptions

Item	RX	TX	Remarks
Validity (V) bit	S	Y	■ RX optionally mutes data if enabled ■ TX passes bit unmodified
User (U) bit	S	Y	■ RX ignores bits ■ TX passed bits unmodified

Identifying Audio Characteristics (Audio Sampling Frequency and Word Length)

Table A-4 Audio Sampling Frequency and Word Length

Item	Remarks
Audio sampling frequency	■ RX: 32 to 108 kHz ■ TX: 48 kHz
Audio word length	■ RX: 16 to 24 bits ■ TX: 16 to 24 bits

B Communication and Control Troubleshooting Tips

Software Communication Problems

Problem

The frame is powered up, but the module does not communicate with CCS Navigator or the web GUI interface.

Solutions

- Ensure you have specified the proper module slot.

  See your *6800***+** *Frame Installation and Operation Manual* for more information about slot identification.
- Confirm there is an 6800**+**ETH module installed in the frame.
- Remove any legacy 6800 series product that is in the frame.

  CCS software cannot communicate with legacy 6800 series products, even if these modules may operate with card-edge controls in the frame. Legacy 6800 products do not have the “**+**” symbol on their extractor handles.
- Check for damaged pins on the back module by following this procedure:
  i. Unplug the front module.
  ii. Unscrew and remove the back module.
  iii. Inspect the 20- or 30-pin spring connector at the bottom of the back module, and verify that the connector does not have any slightly bent pins.
  iv. Carefully reposition any bent pins. If this is not possible, contact Harris Customer Support.

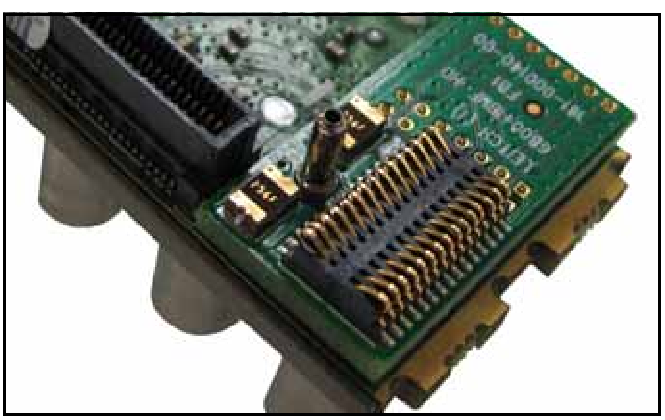


Figure B-1 Typical Back Module Spring Connector

Problem

The frame is powered up, but Pilot Lite does not communicate with the module.

Solution

Verify whether there is a 6800+ETH module installed in the frame. (Pilot Lite serial control is disabled if a 6800+ETH control module is installed in the frame.)

Problem

The IP address of the frame has been forgotten.

Solution

Follow this procedure:

1. Remove the ETH6800**+** module from the frame.
2. Select DIP switch 2 on the ETH6800**+** module and slide the tab to the forward position.
   This sets the ETH6800**+** module to its default IP address of **192.168.100.250**.

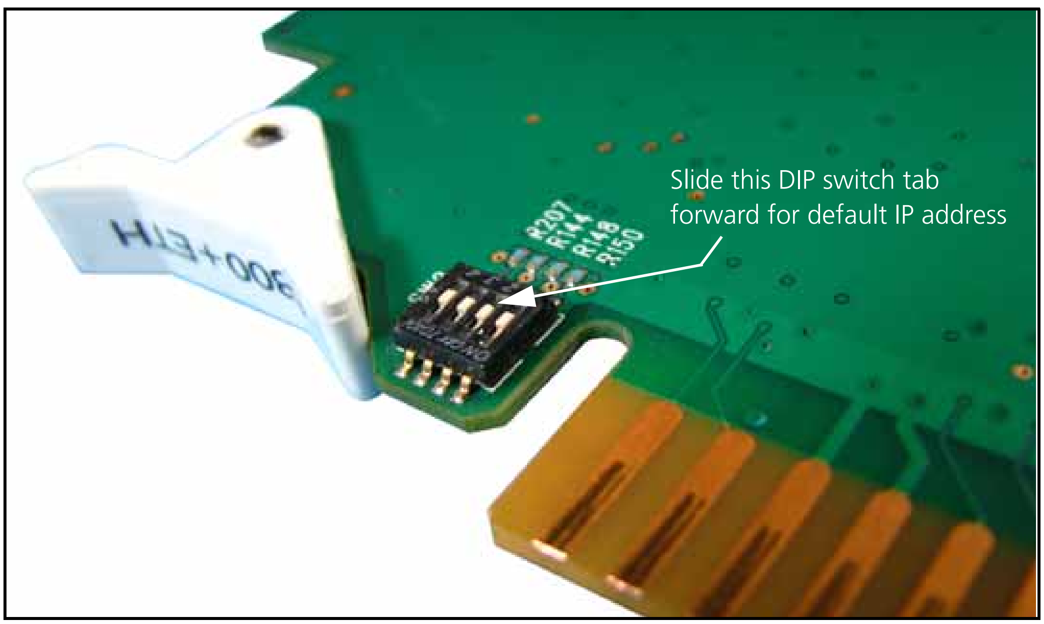


Figure B-2 ETH6800**+** DIP Switch

3. Use CCS Navigator or the web browser to interface with the ETH6800**+** module and then set the desired IP address.
4. Set the DIP switch back to its normal position and then re-insert the module.

Problem

There is no Ethernet connection to the frame.

Solutions

- Verify that the correct IP address is being used. If you are not certain, refer to the above procedure to set to default IP.

- Verify that the ethernet cable is the correct type (a *crossover* cable is required for direct connection from a PC).
- Verify the **Link** and **Activity** (left and right) status LEDS are active on the RJ-45 connector at the rear of the frame.
- Verify the **Status** LED is lit and green on the ETH6800+ card, as viewed from the front of the frame with the door open.
- Verify that the security settings on the PC allow for connection to a frame (TCP port 80 and UDP ports 4000/4050 must be open).
- Verify that the PC is configured for, and can communicate on the desired subnet.

Problem

CCS software sees the frame, but does not find all of the modules.

Solutions

- Remove any legacy 6800 series products.
- Plug your modules in before starting the discovery.
- Start your discovery after the frame and all modules have fully powered up.
- Refresh the CCS software and ensure that the installed modules are fully powered up first before discovery.

Problem

CCS Software does not respond after it is launched.

Solutions

Close any CCS software that is already launched.

Problem

CCS software shows a module in the **Control** window, but cannot control it.

Solution

Follow this procedure:

1. Set the module's Local/Remote jumper to **Remote.**
2. Ensure the module name in the **Control** window matches the module type in the frame.
3. Gently push the module into its slot in the frame to ensure it is seated properly and powered up.
4. Verify that the **Control** window indicates the device is ready.

Hardware Communication Problems

Problem

After a power failure, the frames and PC do not communicate.

Solution

Follow this procedure:

1. Wait four minutes for the frames to recover from the power failure.
2. Close the CCS software, and then restart the PC.
3. Restart the software application.

Problem

The module does not seem to work.

Solutions

- Ensure the correct frame is powered up.
- Verify that all appropriate rear connections are secure.
- Gently push the module into its slot in the frame to ensure it is seated properly. Then verify the **Status** LED on the module is lit and green.
- Follow this procedure to ensure the back module does not have bent pins:
  i. Unplug the front module.
  ii. Unscrew and remove the back module.
  iii. Carefully reposition any bent pins. If this is not possible, contact Customer Support.

  Note: *Pressed* pins are will not affect the functionality of the product.

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For more information, please visit www.broadcast.harris.com.

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