Dolby®

Cat. No. 778

Digital Input/Output Card
For CP650 Cinema Processor

Installation and Operation

Issue 2 Part No. 91836

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Chapter 1 Installation

1.1 Introduction

The Dolby® Cat. No. 778 supports four balanced AES/EBU digital audio input pairs and five balanced AES/EBU digital audio output pairs. It provides Dolby Digital Surround EX™ decoding when playing Dolby Digital soundtracks from 35 mm film. The card also supports decoding of consumer Dolby Digital and Dolby Digital Surround EX bitstreams as found on DVDs and digital television broadcasts.

The digital inputs accommodate bit depths of 16, 20, and 24 bits, with sampling frequencies between 44.1 and 48 kHz. The system is synchronized to the first digital input (L/R) and requires the other digital inputs to be phase and sample synchronous. The digital inputs are designated to support Left/Right, Center/Subwoofer, Left surround/Right surround, and Back surround left/Back surround right channels. The digital audio output channels are designated to support the Left/Right, Center/Subwoofer, Left surround/Right surround, Back surround left/Back surround right, and Left extra/Right extra channels. The digital inputs and outputs are located on the 25 pin female D-connector labeled **Option Card I/O** on the rear panel of the CP650.

Note: The analog outputs of the CP650 are disabled and the CP650 settings are returned to their factory default values when a Cat. No. 778 card is installed. All CP650 outputs are digital.

In addition to the other inputs and formats available on the CP650, the following digital audio formats are supported when the Cat. No. 778 card is installed:

Format 80, Master Digital (auto-switching)
Dolby Digital (consumer)
PCM two-channel: L, R
PCM two-channel: L, R with Dolby Pro Logic® decoding

Format 81, Master Digital with Surround EX decoding

Format 85, Five-Screen Channel
PCM In: L/R, C/LFE, Ls/Rs, Bsl/Bsr (Le/Re input on Bsl/Bsr)
PCM Out: L/R, C/LFE, Ls/Rs, Bsl/Bsr (copy of Ls/Rs), Le/Re

Format 86, Five-Screen Channel with Surround EX decoding
PCM In: L/R, C/LFE, Ls/Rs, Bsl/Bsr (Le/Re input on Bsl/Bsr)
PCM Out: L/R, C/LFE, Ls/Rs, Bsl/Bsr, Le/Re

Format 88, Six-Channel
PCM In: L/R, C/LFE, Ls/Rs
PCM Out: L/R, C/LFE, Ls/Rs

Format 89, Six-Channel with Surround EX decoding
PCM In: L/R, C/LFE, Ls/Rs
PCM Out: L/R, C/LFE, Ls/Rs, Bsl/Bsr

Format 90, Seven-Channel
PCM In: L/R, C/LFE, Ls/Rs, Bs/–
PCM Out: L/R, C/LFE, Ls/Rs, Bsl/Bsr

See Section 3.1 for complete descriptions of each format.

1.2 Installation

Some steps in this upgrade involve handling printed circuit boards. Many components are very sensitive to static electricity. These components can be destroyed if static charge on your body discharges through the component. You do not even have to touch the component to damage it. Before touching the components on the PC boards, ground yourself by rubbing the frame of the unit with each hand or by wearing a grounding strap.

The upgrade kit consists of

- A Cat. No. 778 Card
- Mounting hardware
- 8-Channel digital I/O cable assembly (Dolby Part No. 83435)
- Cinema Resources CD-ROM (Dolby Part No. 82055)

1.2.1 Check CP650 Software Version

The CP650 operating system software must be version 2.2 or higher (versions 2.0 and 2.1 offered preliminary support for the Cat. No. 778). With the CP650 operating normally, follow these steps:

	Press the left **menu** button multiple times to step through the menus to **About this CP650.** **Note:** You can also press and hold the left menu button while rotating the front-panel fader knob clockwise to step through the menu items.
	**About this CP650** is made up of four menu screens. The first screen displays the version number of the installed operating system software. If the version reads “2.1.x.x”(x = any number) or earlier, update the operating system software to version 2.2.x.x or higher.
	Press the illuminated format button to return to the top menu screen.

1.2.2 Installation Steps

1. Remove mains power from the CP650 by unplugging the rear-panel power cord.
2. Open the setup control panel access door.

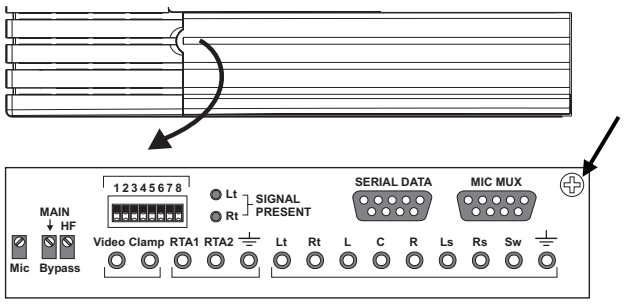


Figure 1-1 CP650 Control Panel

3. Remove the front-panel mounting screw located in the upper right-hand corner of the setup control panel, and carefully pull the front panel toward you to remove it.
4. Remove the seven subpanel mounting screws and carefully pull the subpanel toward you to remove it. Be sure to support the panel while you perform the next step.

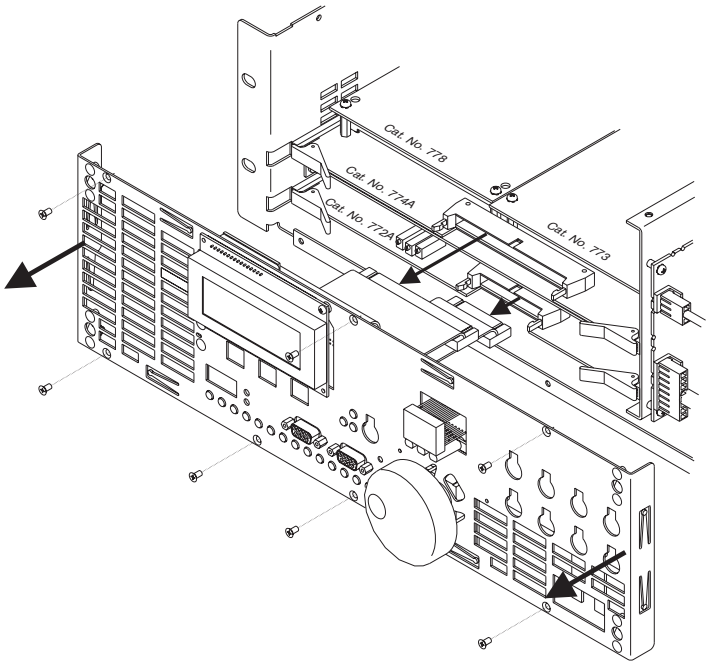


Figure 1-2 Removal of Subpanel

5. Unplug the two ribbon cables connected to the internal circuit boards.
6. Remove the Cat. No. 774A circuit board using the left and right card ejectors. Place the board on a flat surface (for example, on a platter disk). The board should be oriented with the card ejectors close to you.

7. Remove the Cat. No. 778 board from its anti-static bag, and plug it into the **left-hand** side of the main board by aligning the two connectors shown in Figure 1-3. Press down on each side, making sure the connectors are fully seated. The board can be oriented only one way for the connectors to match.

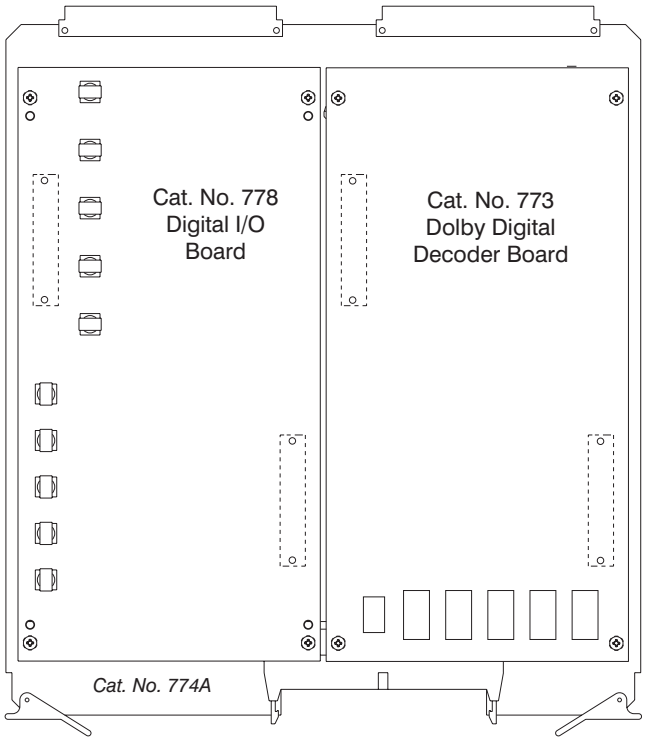


Figure 1-3 Cat. No. 778 Board Location

Note: The Cat. No. 778 will not function if it is installed in the right-hand location. This area is reserved for the Cat. No. 773 Dolby Digital Decoder Board.

8. Turn over the board combination, and install screws through the Cat. No. 774A board and into standoffs at each corner of the Cat. No. 778 board.
9. Reinstall the assembly into the CP650. Push the board in firmly until it is fully seated.
10. Reinstall the two ribbon cables, inner panel, and front panel.

1.2.3 Connections

With the Cat. No. 778 board installed, all digital inputs/outputs appear at the **Option Card I/O** connector on the rear panel of the CP650. The analog outputs of the CP650 are disabled when a Cat. No. 778 card is installed.

Note: When wiring to the digital inputs on the Cat. No. 778, be sure to ground all unused digital inputs. Failure to do so will cause unused channels to show signal due to crosstalk between the used and unused digital inputs.

Table 1 Option Card I/O Connector Pinout with Cat. No. 778 Installed

Pin	Digital Signal I/O with Cat. No. 778 Installed	
1	AES1 (L/R) In +	This digital input pair can receive either PCM or Dolby Digital bitstreams. The remaining AES pairs receive PCM only.
2	AES1 (L/R) In –	
3	AES3 (Ls/Rs) In +	
4	AES2 (C/SW) In +	
5	AES4 (Bsl/Bsr) In + (Le/Re In+ Format 85)	
6	AES5 (X1/2) ½ In +	
7	AES1 (L/R) Out +	
8	AES3 (Ls/Rs) Out +	
9	Chassis Gnd	
10	AES5 (Le/Re) Out +	
11	Chassis Gnd	
12	AES2 (C/SW) In –	
13	AES3 (Ls/Rs) In –	
14	Chassis Gnd	
15	AES1 (L/R) Out –	
16	AES3 (Ls/Rs) Out –	
17	AES5 (Le/Re) Out –	
18	Chassis Gnd	
19	AES5 (X1/2) In –	
20	AES4 (Bsl/Bsr) In – (Le/Re In- Format 85)	
21	AES4 (Bsl/Bsr) Out –	
22	AES4 (Bsl/Bsr) Out +	
23	AES2 (C/SW) Out –	
24	AES2 (C/SW) Out +	
25	Chassis Gnd	

Chapter 2
Operation

Installation of a Cat. No. 778 card along with system software version 2.2 (or higher) provides greater control of system headroom and adds the Unity Gain feature.

2.1 Headroom Configuration

Headroom is the difference (in dB) between the highest peaks of the signal and the highest peak that can be passed through the system without distorting. Use the Headroom Configuration setup menu item to specify how this additional capacity is to be used. The settings available are:

2.1.1 Typical (Default)

This is the recommended setting. All headroom is available for use by the output level fader and/or equalization boost. This setting should not be changed without a thorough understanding of the consequences.

2.1.2 Noise Floor Optimization

Noise Floor Optimization calculates and utilizes any available leftover capacity in each channel to improve the overall noise floor of the CP650. It operates by measuring the extent to which the unit uses the available headroom in the B-chain, then boosts the overall gain in the digital domain to use all available headroom. The analog output level trims are then reduced by the amount of that boost, resulting in an optimally reduced noise floor. This adjustment also factors in the headroom available to the fader. Changes to the output trim values are transparent to the user and no changes will be seen in the fader values in the setup software. Clipping may occur on high level signals if the fader is set above 7.0.

Noise Floor Optimization mode is indicated by an ‘N’ in the top right corner of the front-panel display.

Example

In general, the more headroom that is required by a system for processing, the higher the system noise floor will become relative to the signal.

Consider this example. With Headroom Configuration set to **Typical,** the signal headroom above Dolby® level required for a channel in any processor is, for example, 20 dB. Therefore the optimum Dolby level setting within a digital system is –20 dBFS. In the CP650 with a Cat. No. 778 card installed, this would be the required Dolby level setting at the digital inputs.

The CP650 is required to provide room equalization and fader control, which involves boosting and cutting signals. To provide the required headroom for this potential signal boost, the input level is dropped by, for example, 12 dB within the processor as shown in Figure 2-1. This drops the Dolby level within the CP650 to –32 dBFS. With Headroom Configuration set to **Typical**, Dolby level at the output of the CP650 would be –32 dBFS (with the fader set to 7 and no equalization). This 12 dB drop must be regained in the analog section of the monitor chain (at the amplifiers, for example) resulting in a potential 12 dB increase in noise floor.

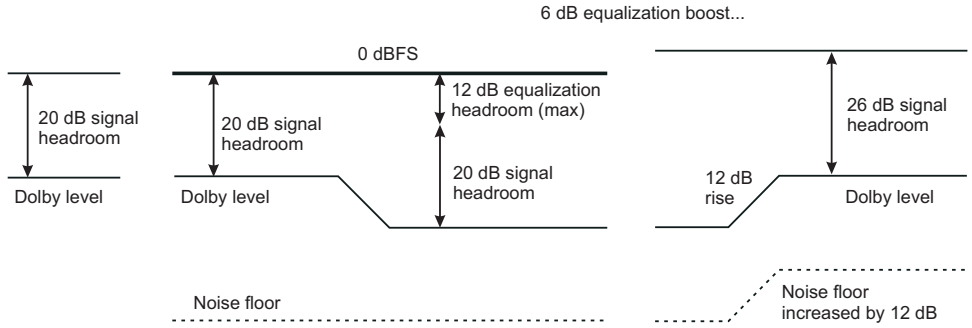


Figure 2-1 Typical Gain Structure With 12 dB Room EQ Required

If the additional headroom required for room equalization is 6 dB rather than 12 dB, the input signal needs be dropped by only 6 dB (to –26 dBFS) in order to maintain sufficient internal headroom. Figure 2-2 shows how, in this example, the noise floor only increases by a potential 6 dB at the amplifiers. “saving” 6 dB of signal-to-noise ratio.

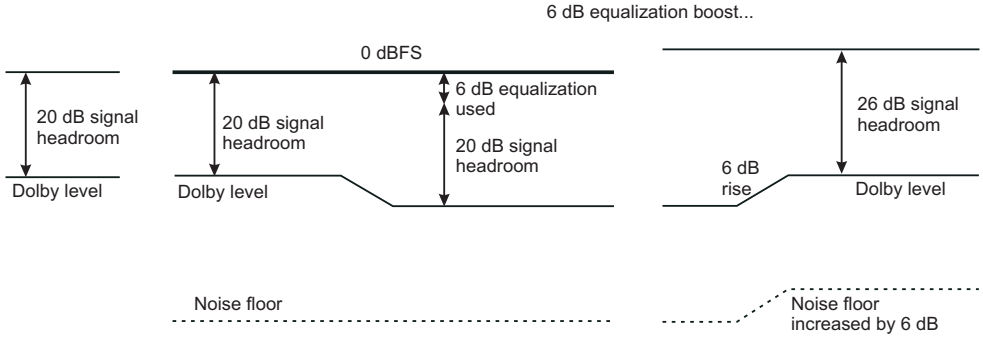


Figure 2-2 Typical Gain Structure with 6 dB Room EQ Required

The Noise Floor Optimization mode is a process that looks at the maximum headroom required for each channel independently, and adjusts the internal Dolby level setting for that channel, dependent on the headroom required. The process maintains the maximum signal-to-noise ratio of each channel.

Note: Sound pressure level must be regularly checked during equalization setting because the effective output level of the CP650 will change whenever equalization is adjusted.

Noise Floor Optimization can also be activated in analog CP650 variants, however the process becomes transparent since the analog output is automatically adjusted to compensate for the change in internal gain during room equalization.

2.1.3 Unity Gain

Unity Gain allows a digital signal to pass through the CP650 while maintaining its original level. In this condition, the CP650 will pass digital signal inputs up to full-scale on all channels without clipping, provided that the Graphic EQ, Subwoofer EQ, Treble Bulk EQ, Bass Bulk EQ, Optical Surround Level Trim, Subwoofer Level Trim, and/or Fader provide no boost in level. (When the Cat. No. 778 card is initially installed, these settings are automatically set to flat, but may have been changed subsequently.)

The CP650 is designed to process a +0 dB signal as a full-scale digital signal (+0 dBFS). Normal operation produces a –12 dB insertion loss in the B-chain. This allows for +6 dB boost in the EQ, and up to +6 dB additional boost by the fader (above nominal 7.0) without causing numerical overflows (digital clipping) within the internal digital signal processing. The fader algorithm produces a ‘soft’ clip when it encounters an overflow condition, minimizing the objectionable effects of unmodified digital clipping. While the Unity Gain feature is enabled, the CP650 can, at most, deliver a +0 dBFS signal output from a +0 dBFS input. Any desired additional boost must occur outside the unit.

Unity Gain mode is indicated by a ‘U’ in the top right corner of the front-panel display.

Note: Noise Floor Optimization and Unity Gain cannot be selected simultaneously.

Test

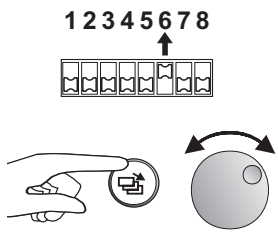
The Test function within Unity Gain mode will place the CP650 in a temporary configuration which guarantees that full-scale digital signal inputs (0dBFS) appear at the same level on the outputs. The Test function applies several configuration changes:

- The level adjustments for surround and subwoofer levels are suspended. These channels leave a 0 dBFS signal at 0 dBFS.
- All B-Chain equalizations are held flat to avoid inadvertent clipping.
- The fader is held at 7.0.
- The output level trims are set to 0 dB (no reduction in level).

The CP650 passes a signal at the Cat. No. 778 digital inputs to its outputs without a net change in level. Overall console calibration and reference measurements are possible.

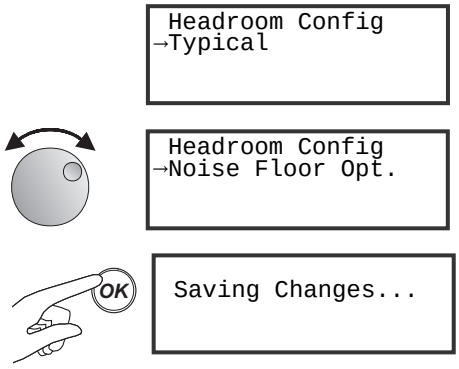
The system restores the previous settings after exiting the Test mode.

2.2 Headroom Configuration Menus



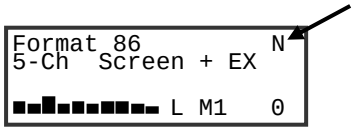
Slide DIP switch 6 to the up position. This switches the CP650 into Setup mode.

Press the left menu button multiple times (or press and hold the left menu button while rotating the front-panel fader knob) to move to the **Headroom Config** menu.

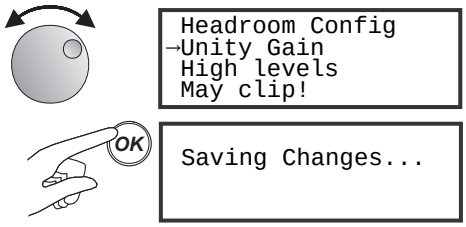


See the description of the **Typical** and **Noise Floor Optimization** settings in the previous section.

Rotate the front-panel knob to select the desired setting then press OK to save this setting.

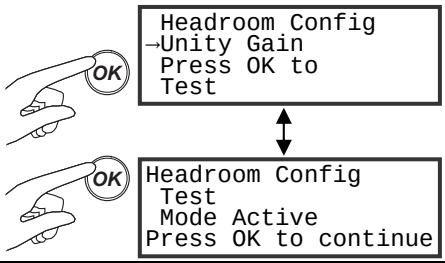


Noise Floor Optimization mode is indicated by an 'N' in the top right corner of the front-panel display.

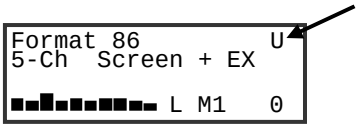


The **Unity Gain** setting is described in Section 2.1.3.

Press OK to set the unit to Unity Gain.



With Unity Gain selected, pressing the OK button toggles the unit between the Unity Gain select screen and Test Mode. With Test mode active, EQ and output level trims are forced to their default unity gain settings to ensure true unity gain for a temporary confidence check.



Unity Gain mode is indicated by a 'U' in the top right corner of the top-level menu.

Chapter 3
Reference Information

3.1 Format Descriptions

If a format name is too long to display on the CP650 front panel, a second, shorter name is given:

Format 80
Bitstream Master Digital Input
(Short name: **Master Digital In**)

Audio Description: The input is a digital bitstream into the Option Card I/O connector, AES 1 (L/R) pins. This format automatically detects whether the bitstream is PCM or Dolby® Digital (AC-3), and configures the decoder appropriately. For a PCM (two-channel) input signal, matrix decoding may be applied using the "Format 80/81 PCM Options" screen. This screen also contains a "PCM Auto Mute" option which causes the CP650 to mute automatically if it does not detect a Dolby Digital bitstream. This prevents noise when working only with encoded bitstream audio. For Dolby Digital, the metadata contained in the bitstream specifies the channel format and surround processing to be applied.

This format does not support more than two channels of PCM.

Format 81
Bitstream Dolby Digital Surround EX
(Short name: **Fmt 80 w/Surr. EX**)
Audio Description: The input signal is a digital bitstream into the Option Card I/O input. This audio format is the same as format 80, but it applies Dolby Digital Surround EX™ decoding to any bitstream with exactly two surround channels (3/2, 3/2L, or 2/2) regardless of the Surround EX metadata information. This produces three surround outputs, Ls, Bs, and Rs. Bs is copied to both the Bsl and Bsr outputs at the appropriate levels.

Format 85
5-Screen Channel
(Short name: **5-Ch Screen**)
Audio Description: This format uses eight discrete input channels: L/R, C/LFE, Ls/Rs, and Bsl/Bsr (Le/Re input on Bsl/Bsr). The audio clock source is the L/R input (first pair).

The output channels are L/R, C/LFE, Ls/Rs, Bsl/Bsr (a copy of Ls/Rs), and Le/Re.

The surround levels are internally adjusted, where Ls/Bsl together produce 82 dBC SPL, and Rs/Bsr together produce 82 dBC SPL.

Format 86
5-Screen Channel with Surround EX
(Short name: **5-Ch Screen + EX**)
Audio Description: This format is identical to Format 85, with the exception that Surround EX decoding is applied to Ls and Rs, producing additional Bsl/Bsr (identical) outputs. The surround levels are internally adjusted where Ls/Rs each produce 82 dBC SPL and Bsl/Bsr each produce 79 dBC SPL (which total 82 dBC SPL).

Format 88
6-Channel PCM
Audio Description: This format uses six discrete input channels (L/R, C/LFE, and Ls/Rs) and processes them in the following manner:

1. The master audio clock source is the L/R input (first AES pair).
2. Dolby Digital bitstream decoding is disabled.
3. Digital subwoofer EQ and level processing is applied, but no low-pass filter is used.
4. The Ls input is duplicated on the Bsl output, and the Rs input is duplicated on the Bsr output.
5. The surround level adjustments are similar to those for 5.1 film formats, where Ls/Bsl together produce 82 dBC SPL, and Rs/Bsr together produce 82 dBC SPL.

Format 89
6-Channel PCM + Surround EX
(Short name: **6-Channel PCM + EX**)
Audio Description: This format is identical to Format 88, with the exception that Surround EX decoding is applied to Ls and Rs, producing additional Bsl/Bsr (identical) outputs.

Format 90
7-Channel PCM
Audio Description: This format is used to input seven discrete channels: L/R, C/LFE, Ls/Rs, and Bs/- , as would exist in a pre-encoded Surround EX mix. Processing is identical to Format 88 with the following exceptions:

1. The Back Surround (Bs) input (Channel 1 of AES input 4) is copied to Bsl and Bsr outputs.
2. No Surround EX decoding is performed.

The output levels are adjusted to produce reference 85 dB SPL in Ls and Rs, and 82 dB SPL in Bsl and Bsr. (This produces a net 85 dB SPL from the combined Back Surround outputs.)

3.2 Digital Audio Inputs

There are two professional interface formats used for digital audio: **AES/EBU** (also known as AES3) and **AES-3id**. These stream the same digital data and professional audio header information over copper conductor links, but use different types of conductors and connectors. The Cat. No. 778 digital inputs are AES/EBU.

AES/EBU uses a balanced connection (two conductors plus shield) with a characteristic input impedance of 110Ω, nominal peak-to-peak signal level of 5 V, and, most commonly, XLR connectors. The typical maximum transmission distance is 100 meters (328 feet). AES-3id uses an unbalanced connection (one signal conductor plus shield) with a characteristic input impedance of 75Ω, peak-to-peak signal level of 1 V, and BNC ("push and twist") connectors. The typical maximum transmission distance is 1,000 meters (3,280 feet).

Professional digital audio equipment usually uses the AES/EBU format because balanced operation yields superior noise immunity, as it does with analog audio signals, and because XLR connectors have been standard on analog professional audio equipment.

Professional video equipment usually uses the AES-3id variation of this interface, with BNC connectors. As with the use of XLR connectors on pro audio equipment, the adoption of BNC connectors for the audio on professional video equipment stems from their existing use for the video signal. Also, the unbalanced AES-3id signal can connect to more than one piece of equipment by using the loop-through connectors available on some devices. The signal is robust for long cable runs.

3.2.1 Consumer Interface Standards for Digital Audio

The consumer interface standard for digital audio is **S/PDIF** (IEC61937). S/PDIF is found using either coaxial unbalanced connections (one signal conductor plus shield) with a characteristic input impedance of 75Ω with RCA (phono) connectors, or a fiber-optic cable with Toslink™ connectors. The unbalanced coaxial connection has a peak-to-peak signal level of 0.5 V. The typical maximum transmission distance is 10 meters (33 feet). Although S/PDIF-specific cables with suitable connectors can be

purchased, you can also get good results using high-quality 75Ω video cable with the appropriate connectors and/or adapters.

3.2.2 Cable Issues

Even in digital audio, noise-free signals are still very important. The cable used for digital signals is specifically designed for digital audio use even though it appears to be the same as that used for analog audio or video signals. Any professional audio equipment or broadcast supply company can provide 110Ω cable with connectors (or without, if you wish to terminate them yourself) for AES/EBU connections, and high-quality 75Ω video cables with BNC connectors for AES-3id connections. Use of cables or connectors not designed for digital transmission or with incorrect impedance compromises the integrity of the bitstream and may create an unreliable link between pieces of equipment, particularly those with long cable runs.

3.2.3 Multiple Sources: Conversion Between Interface Standards

Although some details of the bitstreams used in the AES and S/PDIF standards are different, the audio information is exactly the same. As a consequence, most audio equipment accepts either standard with no need to convert the bitstream itself; this is the case with the CP650. However, if you intend to connect sources across different types of digital audio inputs, do **not** attempt to convert a digital interface type by, for example, directly wiring an XLR connector to a BNC or RCA plug. This causes an impedance mismatch and signal reflections, resulting in degradation of the digital waveform. It may seem to work, but the results are unreliable and dropouts occur.

For conversion between the AES-3id and S/PDIF formats, you can use high-quality RCA phono plug-to-BNC adapters because the cable and impedance are both the same (75Ω).

For conversion between the AES/EBU and AES-3id or AES/EBU and S/PDIF formats, a simple and economical method is to use inline transformers. These devices perform the necessary impedance and balanced/unbalanced conversion. Table 2 shows some examples of suitable adapters. The unbalanced connector in these examples is a BNC. BNC-to-RCA adapters can be added to connect to consumer S/PDIF connections. The units listed use passive circuitry.

Table 2 Examples of Available Balanced ↔ Unbalanced Adapters

Adapter Type	Neutrik®	Canare™
XLR female 110Ω in to BNC female 75Ω out	NA-BF	BCJ-XJ-TRA
BNC female 75Ω in to male XLR 110Ω out	NA-BM	BCJ-XP-TRA

Higher-priced units incorporating active circuitry are also available. These offer additional features like multiple inputs, inputs for Toslink digital connections, and multiple outputs.