Dolby

Model DP562

Multichannel Dolby Digital / Dolby Pro Logic Decoder

User's Manual

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If you purchased the Model DP562 through a dealer or systems integrator, please contact them for technical support.

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Regulatory Notices

FCC

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with this instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his or her own expense.

Canada

This Class A digital apparatus complies with Canadian ICES-003.

UL

Troubleshooting must be performed by trained technicians. Do not attempt to service this equipment unless you are qualified to do so.

⚠ **WARNING**: *Check that the correct fuses have been installed. To reduce the risk of fire, replace the fuses only with the same type and rating.*

Exposed portions of the power supply are electrically "hot". In order to reduce the risk of electrical shock, the power cord MUST be disconnected when the cover of this equipment is removed.

The ground terminal of the power plug is connected directly to the chassis of the unit. For continued protection against electric shock, a correctly wired and earthed three-pin power outlet must be used. Do not use a ground-lifting adapter and never cut the ground pin on the three-prong plug.
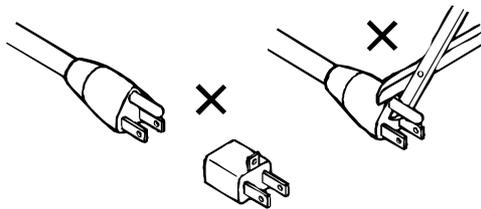
UK

As the colours of the cores in the mains lead may not correspond with the coloured markings identifying the terminals in your plug, proceed as follows:

- The core which is coloured green and yellow must be connected to the terminal in the plug which is marked with the letter **E** or by the earth symbol ⏚ or coloured green or green and yellow.

- The core which is coloured blue must be connected to the terminal which is marked with the letter **N** or coloured black.

- The core which is coloured brown must be connected to the terminal which is marked with the letter L or coloured red.

This apparatus must be earthed.

IEC Notices

This equipment complies with the EMC requirements of EN55103-1 and EN55103-2 when operated in an E2 environment in accordance with this manual.

IMPORTANT SAFETY NOTICE

This unit complies with the safety standard EN60065. To ensure safe operation and to guard
against potential shock hazard or risk of fire, the following **must** be observed:

o   Ensure that your mains supply is in the correct range for the input power requirement of the unit.
o   Ensure **fuses** fitted are the **correct rating and type** as marked on the unit.
o   The unit **must be earthed** by connecting to a correctly wired and **earthed** power outlet.
o   The **power cord** supplied with this unit must be wired as follows:

<div align="center">Live—Brown   Neutral—Blue   Earth—Green/Yellow</div>

(GB)

IMPORTANT – NOTE DE SECURITE

Ce materiel est conforme à la norme EN60065. Pour vous assurer d'un fonctionnement sans danger et de prévenir
tout choc électrique ou tout risque d'incendie, veillez à observer les recommandations suivantes.

o   Le selecteur de tension doit être placé sur la valeur correspondante à votre alimentation réseau.
o   Les fusibles doivent correspondre à la valeur indiquée sur le materiel.
o   Le materiel doit être correctement relié à la terre.
o   Le cordon secteur livré avec le materiel doit être câblé de la manière suivante:

<div align="center">Phase—Brun   Neutre—Bleu   Terre—Vert/Jaune</div>

(F)

WICHTIGER SICHERHEITSHINWEIS

Dieses Gerät entspricht der Sicherheitsnorm EN60065. Für das sichere Funktionieren des Gerätes und zur
Unfallverhütung (elektrischer Schlag, Feuer) sind die folgenden Regeln unbedingt einzuhalten:

o   Der Spannungswähler muß auf Ihre Netzspannung eingestellt sein.
o   Die Sicherungen müssen in Typ und Stromwert mit den Angaben auf dem Gerät übereinstimmen.
o   Die Erdung des Gerätes muß über eine geerdete Steckdose gewährleistet sein.
o   Das mitgelieferte Netzkabel muß wie folgt verdrahtet werden:

<div align="center">Phase—braun   Nulleiter—blau   Erde—grün/gelb</div>

(D)

NORME DI SICUREZZA – IMPORTANTE

Questa apparecchiatura è stata costruita in accordo alle norme di sicurezza IEC 65. Per una perfetta
sicurezza ed al fine di evitare eventuali rischi di scossa êlettrica o d'incendio vanno osservate le
seguenti misure di sicurezza:

o   Assicurarsi che il selettore di cambio tensione sia posizionato sul valore corretto.
o   Assicurarsi che la portata ed il tipo di fusibili siano quelli prescritti dalla casa costruttrice.
o   L'apparecchiatura deve avere un collegamento di messa a terra ben eseguito; anche la connessione rete deve
    avere un collegamento a terra.
o   Il cavo di alimentazione a corredo dell'apparecchiatura deve essere collegato come segue:

<div align="center">Filo tensione—Marrone   Neutro—Blu   Massa—Verde/Giallo</div>

(I)

AVISO IMPORTANTE DE SEGURIDAD

Esta unidad cumple con la norma de seguridad EN60065. Para asegurarse un funcionamiento
seguro y prevenir cualquier posible peligro de descarga o riesgo de incendio, se han de observar
las siguientes precauciones:

o   Asegúrese que el selector de tensión esté ajustado a la tensión correcta para su alimentación.
o   Asegúrese que los fusibles colocados son del tipo y valor correctos, tal como se marca en la unidad.
o   La unidad debe ser puesta a tierra, conectándola a un conector de red correctamente cableado y puesto a tierra.
o   El cable de red suministrado con esta unidad, debe ser cableado como sigue:

<div align="center">Vivo—Marrón   Neutro—Azul   Tierra—Verde/Amarillo</div>

(E)

VIKTIGA SÄKERHETSÅTGÄRDER!

Denna enhet uppfyller säkerhetsstandard EN60065. För att garantera säkerheten och gardera mot
eventuell elchock eller brandrisk, måste följande observeras:

o   Kontrollera att spänningsväljaren är inställd på korrekt nätspänning.
o   Konrollera att säkringarna är av rätt typ och för rätt strömstyrka så som anvisningarna på enheten föreskriver.
o   Enheten måste vara jordad genom anslutning till ett korrekt kopplat och jordat el-uttag.
o   El-sladden som medföljer denna enhet måste kopplas enligt följande:

<div align="center">Fas—Brun   Neutral—Blå   Jord—Grön/Gul</div>

(S)

BELANGRIJK VEILIGHEIDS-VOORSCHRIFT:

Deze unit voldoet aan de EN60065 veiligheids-standaards. Voor een veilig gebruik en om het gevaar van electrische
schokken en het risico van brand te vermijden, dienen de volgende regels in acht te worden genomen:

o   Controleer of de spanningscaroussel op het juiste Voltage staat.
o   Gebruik alleen zekeringen van de aangegeven typen en waarden.
o   Aansluiting van de unit alleen aan een geaarde wandcontactdoos.

<div align="center">Fase—Bruin   Nul—Blauw   Aarde—Groen/Geel</div>

(NL)

Chapter 1
Introduction and Pre-Installation

Thank you for purchasing the DP562 Multichannel Dolby Digital and Dolby Pro Logic Decoder. The DP562 is designed to serve as the reference decoder for all high-performance monitoring applications. We hope that the DP562 will offer you the exceptional quality that you expect from all of our professional audio products.

1.1   About This Manual

This User's Manual has been prepared specifically to help you get the most from the DP562, and includes the proper steps to follow before operating the DP562 in order to take advantage of all of its features. We suggest that you keep this manual readily available for reference. Two additional manuals, *Dolby Digital Broadcast Implementation Guidelines* (Part No. 91549) and *Dolby Digital Professional Encoding Manual* (Part No. 91535) provide in-depth operational details for specific applications. Contact your dealer or Dolby Laboratories for availability.

Advanced Television Systems Committee (*ATSC*) specifications are often referenced in this manual to explain methods of standardization applied to the DP562. You can find a link to the *ATSC A/52 Annex B* specification under "Technical Information" at our Web site, www.dolby.com, or you can link directly to: www.atsc.org and select **standards**.

Conventions

The DP562 User's Manual employs the following conventions:

- The term AC-3 in figures and tables refers to Dolby Digital.

- "Warning" and "Note" annotations contain important information specific to the text that it follows.
  Examples:

  > ⚠ *WARNING: Check that the correct fuses have been installed. To reduce the risk of fire, replace the fuses only with the same type and rating.*

NOTE: Both audio pins of the XLRs (pins 2 and 3) must be connected; neither may be left open.

- The DP562 controls are represented by bold-faced capital letters enclosed in brackets.

- Example:
  Press **[ENTER]**.

- Menu selections shown in the liquid crystal display (LCD) of the DP562 are represented as follows:

  In the body of the text, they are written in bold-face type:

  User menu

  In figures, a frame surrounds the text:

  ```
  ┌─────────────────────┐
  │     USER MENU       │
  │   USER FUNCTION     │
  └─────────────────────┘
  ```

1.2   About the Dolby Model DP562

The Dolby DP562 is a reference decoder for monitoring audio, video, and multimedia programs with multichannel sound. The unit decodes Dolby Digital (AC-3) program material with up to 5.1 channels, and provides four-channel Dolby Pro Logic decoding for matrix-encoded surround material (see ***Appendix A***, ***About Dolby Digital and Dolby Pro Logic***). It can also be used to downmix a multichannel Dolby Digital input signal to a two-channel (Lt/Rt) Dolby Surround signal for simultaneous monitoring through the Pro Logic decoder.

Unlike previous professional surround decoders, the DP562 can decode Dolby Pro Logic in the digital domain; therefore, it is now possible to monitor Dolby Surround-encoded PCM soundtracks without converting them to analog. The unit occupies 1U (44 mm, 1.75") of standard (19") rack space.

Features

- Dolby Digital and Dolby Pro Logic decoding
- Dolby Digital (IEC 1937) or PCM (AES/EBU) input via XLR connector
- Automatic Dolby Digital bitstream detection
- Three digital (AES/EBU) and six balanced analog outputs (XLR connectors)
- 20-bit digital to analog converters
- Analog output level control (+10 dB to –95 dB), level trims (+6 dB to –18 dB), and output channel assignment
- Two-line, 16-character alphanumeric front-panel LCD
- Front-panel channel activity indicators
- Built-in test noise sequence generator for balancing playback system
- Adjustable center and surround channel delays
- Serial remote control capability
- Front-panel headphone jack with adjustable level
- Custom Dolby Digital compression parameter adjustment
- Bass management

1.3   Specifications and Block Diagram

Audio Sampling Rates
32, 44.1, or 48 k samples/second
Output sample rate follows input

Frequency Response
Digital outputs: 20 Hz–20 kHz ±0.01 dB
Analog outputs: 20 Hz–20 kHz ±0.1 dB

Distortion
Less than 0.005%, 20 Hz–20 kHz

Dynamic Range
Greater than 100 dB

Trim Level Adjust
+6 to –18 dB in 0.125 dB steps

Crosstalk
Better than –100 dB @ 1 kHz

Digital Audio Input
Dolby Digital (AC-3) in IEC 1937 format (formerly
IEC 958 Annex B) as specified in ATSC A/52
Annex B. PCM in AES/EBU format
3-pin female XLR connector

Digital Audio Outputs
AES/EBU PCM. 5 Vpp, 110 Ω. Three 3-pin male
XLR connectors

Analog Audio Outputs
0 dBFS = +24 dBu, balanced floating, 25 Ω output
impedance. 20-bit DAC. 3-pin male XLR connectors

Headphone Output
+11.5 dBu maximum output into 600 Ω nominal.
1/4" standard audio headphone jack. Level adjustable
over 60 dB range.

Serial I/O
RS-232, 9600 bps. 9-pin female D connector.
The DP562 is configured as a DCE device.

Status/Remote Input
9-pin female D connector

Front-Panel Displays
Two-line by 16-character alphanumeric LCD with
back lighting

Six-channel Activity/Signal Presence LEDs
(L, C, R, LS, RS, SW):
Off: input not present and not selected for playback
Yellow: input present and selected for playback
Green: signal present with level above –60 dBFS
Red: signal present with level above –0.1 dBFS (clipped)
NOTE: LS and RS illuminate simultaneously to indicate
presence of mono surround channel in Pro Logic-decoded
signal.

Decoding Modes
Dolby Digital, Dolby Pro Logic, Dolby Digital +
Dolby Pro Logic, and Pass-through

Listening Modes
Full, 3 Stereo, Phantom, Stereo, Mono

Compression Modes
None, Custom, Line, RF. Custom and Line modes
have adjustable parameters.

Test Noise
Auto: Sequences wide-band or band-limited pink
noise through enabled channels at two-second
intervals
Manual: Feeds continuous noise to selected channel
when unit is in trim level mode.

Power Requirements
90–264 VAC, 50–60 Hz, auto sensing,
30 W maximum

Dimensions and Weight
45 mm x 483 mm x 305 mm (1.75" x 19" x 12").
Net: 4.9 kg (10.5 lbs)

Operating Conditions
5–45° C, natural convection cooling.
0–90% relative humidity (non–condensing).

Regulatory Notices
North America: This unit complies with the limits for
a Class A digital device, pursuant to Part 15 of the
FCC Rules. It is UL listed for the U.S. and Canada.

Europe: This product complies with the requirements
of Low Voltage Directive 73/23/EEC and EMC
Directive 89/336/EEC and carries the CE marking
accordingly.

Warranty
One-year limited, parts and labor

Specifications subject to change without notice

System Block Diagram

				External
Front Panel	Microcontroller			Level control
				RS232
				Status Out

Signal Presence

AES/EBU Receiver

DSP

AC-3 Decoder

DSP

Pro Logic Decoder

Assignable Digital Outputs

AES/EBU transmitter — 1,2

AES/EBU transmitter — 3,4

AES/EBU transmitter — 5,6

Six Channel DAC

6

Six Channel Level Control

Assignable Analog Outputs
1
2
3
4
5
6

Power Supply

Channel Select

2

Level Control

Headphone Out

1.4   Controls and Connectors

Dolby Decoder Keys
Select among four decoding modes:
Dolby Digital, Dolby Pro Logic, Dolby Digital + Pro Logic, PCM pass-through.

Listening Modes
Five listening (downmix) modes are available:
Full - all input channels.
3 Stereo - no surround channels.
Phantom - Left and Right create a "phantom" Center channel.
Stereo - only Left and Right front channels.
Mono - Center channel only.

Output Activity LEDs
Tri-color LEDs indicate individual channel status.
Yellow - Channel enabled.
Green - Signal above -60dB FS.
Red - Signal above -0.1dB FS (saturation condition).

Headphone Jack
1/4" standard audio headphone jack.

Headphone Volume Control
270 degree rotation control;
controls headphone output level.

Dim
Attenuates analog outputs by -20 dB.

Setup
Invokes Setup menu which provides configuration
of DP562 to individual user's environment.

User
Custom binary switch defined through Setup menu.

Test Noise
Sequences pink noise through all channels at two second intervals.
Continuous generation from a single channel is enabled if
Test Noise is ON when the unit is in trim level mode.

Dynamic Compression
Select Dolby Digital dynamic compression mode:
None - no compression.
Custom - dialog normalization selectable; custom scaling of low and high level compression parameters.
Line - dialog normalization on; custom scaling of low and high level compression parameters.
RF - dialog normalization and compression on, 11 dB gain.

Menu Navigator Keys
Arrow keys navigate through status screens and Setup menu.

Enter
Enter key selects Setup menu entry.

Display
Displays PCM and Dolby Digital status.
Displays entries in Setup menu.

Figure 1 - 1. Front panel.

Status / Remote
Connections for remote status LEDs.
Connections for potentiometer which function as an external analog output level controller.

Digital Input
Encoded Dolby Digital, Dolby Surround-encoded PCM, or stereo PCM encapsulated within AES/EBU / IEC 1937-compatible bit stream.

AC Mains Connector
90 - 264Vac.
Main fuse: 1A 20mm time-lag.
Internal fuse: 2A 20mm fast-blow.

Digital Outputs CH 1/2, 3/4, 5/6
Decoded PCM outputs formatted in AES/EBU / IEC1937 bit stream standard.
Output channel connector can be reassigned to arbitrary audio output channels.

Analog Outputs CH1 - 6
Six balanced analog outputs; drives 2k load impedance at 0 dBFS = 24.0 dBu. Channel assignment follows assignment for digital output channels.

Serial I/O
Serial Interface to exchange commands / status with a remote terminal or PC.

WARNING: Check fuse rating before applying power.

WARNING: No user-serviceable parts inside. Refer all service to qualified personnel.

Digital Input

Digital Audio Outputs

Ch 5/6   Ch 3/4   Ch 1/2

Model DP562   Multichannel Decoder   Status/Remote   Serial I/O

Analog Audio Outputs

Ch 1   Ch 2   Ch 3   Ch 4   Ch 5   Ch 6

Dolby and the double-D symbol are trademarks of Dolby Laboratories Licensing Corporation.

Figure 1 - 2. Rear panel.

1.5   Pre-Installation

Unpacking

Before unpacking the DP562, inspect the outer carton for shipping damage. If the carton shows damage, inspect the unit especially carefully in those areas.

Inspection

Carefully remove the unit from its carton, remove the plastic wrapping, and place on a flat surface. If there are no signs of damage, proceed to "Fusing Information" below.

1.6   Fusing Information

WARNING*: To reduce the risk of fire, replace fuses only with the same type and rating.*

The DP562 uses a universal switching power supply that handles the full range of nominal mains voltages between 90 VAC and 264 VAC and any frequency between 50 Hz and 60 Hz.

1.6.1   Check Main Fuse

1. With a small flat-blade screwdriver, open the fuse compartment door in the AC power input housing as shown below in Figure 1-1.
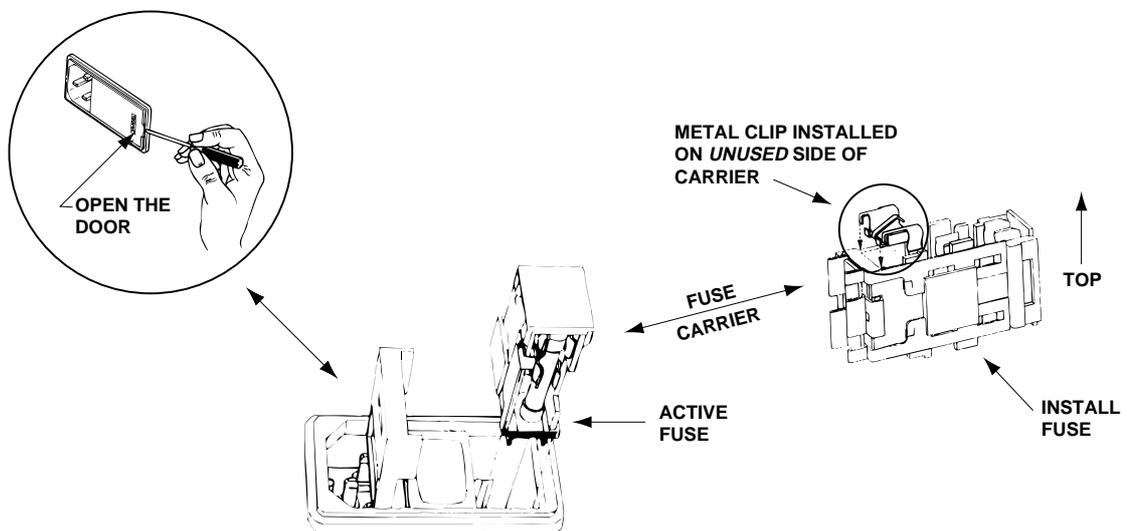
Figure 1–1. Check Main Fuse.

2. Check that the fuse in the active (lower) fuse carrier is of the correct rating. The fuse carrier must be inserted into the compartment with the orientation as shown. *Do not force the carrier into the compartment. Damage will result.*

3. Snap the fuse-compartment door closed.

1.6.2   Internal Fuse

The switching power supply contains a separate fuse. Most fault conditions are protected by the main fuse. The internal fuse rating is:

2 Amp, 250V, 20 mm, fast-blow (all operating voltages)

1.6.3   Internal Signal Ground-to-Chassis Link J13 [OPEN]

It is normal practice to connect signal ground to power-line ground for safety and other reasons. In most systems, minimum induced hum is obtained when this connection is made at only one piece of equipment in the audio chain.

To minimize the addition of circulating ground currents in a particular installation, the default setting for this jumper link is OPEN. Note that there is always a 1k Ω resistor across the link so that the audio ground is never totally isolated from the chassis ground.

To connect audio ground directly to chassis ground in the DP562, remove the cover and move jumper J13 to the closed position.

WARNING: The chassis is always connected to the ground pin of the power line cord. For safety reasons this ground wire must NEVER be disconnected.

Chapter 2
Installation

2.1   Mounting

The DP562 occupies a 1–U (1 ¾" x 19") rack mount and can be positioned in any orientation. Ensure adequate ventilation, and do not mount the unit directly above any heat-generating equipment. The unit operates within specifications up to a maximum ambient temperature of 50ºC (122ºF). Note that the ambient temperature inside a poorly ventilated rack may be considerably higher than the room temperature.

2.2   Audio Connections

See rear-panel drawing on page 1-5 .

- A single female input XLR is provided for digital sources (Dolby Digital [AC-3] or Dolby Surround-encoded PCM [Pulse Coded Modulation][*] data.).

- Three male output XLRs (Ch 1/2, Ch 3/4, Ch 5/6) are provided for PCM data (AES/EBU[**]).

- Six male XLR connectors that carry balanced floating analog outputs corresponds to Channels 1 through 6.

2.2.1   Digital Input

The DP562 accepts encoded Dolby Digital data, Dolby Surround-encoded PCM, or sampled PCM. All three types of input signals must be consistent with *AES/EBU or IEC 1937* (formerly *IEC 958 Annex B*) standards.

Connect the digital source to the DP562 **Digital Input** connector using a shielded cable with a male XLR connector. S/PDIF signals can be connected using a 75 Ω/110 Ωtransformer. The DP562 decodes sampling rates of 32 kHz, 44.1 kHz, and 48 kHz.

AES/EBU, S/PDIF (AC-3, PCM)

110 Ω, balanced, 3-pin XLR female rear-panel connector

Pin	Connection
1	Digital signal ground
2	Digital audio input +
3	Digital audio input -

---

[*] PCM (Pulse Coded Modulation) is a form of data transmission in which digital numbers represent amplitude samples of an analog signal.
[**] AES/EBU is a standard format in accordance with the Audio Engineering Society and European Broadcast Union.

2.2.2   AES/EBU Outputs

Use balanced and shielded audio cables with female XLR connectors to connect to the AES/EBU **Digital Audio Outputs** (Ch 1/2, Ch 3/4, Ch 5/6).

AES/EBU (PCM)
5Vp-p, 110 Ω, balanced, 3-pin XLR male rear-panel connector

Pin	Connection
1	Digital signal ground
2	Digital audio input +
3	Digital audio input -

2.2.3   Analog Audio Outputs

Connect the DP562 **Analog Outputs** (Ch 1-Ch 6) to balanced and shielded audio cables with female XLR connectors.

AES and IEC convention calls for XLR pin 2 to be "high/hot" and pin 3 to be "low/cold." In the interest of maintaining international standards, we recommend following AES and IEC standards.

In an installation where the load is unbalanced, avoid ground loops by using two-conductor shielded cables for balanced circuits. Ensure that the unbalanced load occurs only at the DP562 end by connecting pin 3 to ground.

NOTE*: Both audio pins of the XLRs (pins 2 and 3) must be connected; neither may be left open.*

AES/EBU
25 ohms output impedance, balanced floating, 3-pin XLR male rear-panel connector

Pin	Connection
1	Analog Signal Ground
2	Audio +
3	Audio -

2.3   Status / Remote

The Status/Remote 9-pin D-connector on the rear panel allows remote monitoring of status and remote control of the analog output level.

When monitoring status remotely, the front-panel controls continue to operate normally, while the analog output level adjustments are disabled and can only be adjusted by closing the remote switch. Please refer to the remote circuit example in *Figure 2-1*.

NOTE*: The output channel trims are adjustable when the remote level controller is enabled. The Master Level cannot be adjusted.*

Status/Remote
9-pin female D-connector on the rear panel

Pin	Connection	Comment
1	Fault	Processor/memory fault
2	AC-3/PCM	1=AC-3 bitstream; 0=PCM bitstream
3	AC-3 CRC Error	AC-3 CRC error encountered during decode
4	AC-3 CRC Error	Pins 3 and 4 closed for 100 ms when CRC error detected
5	Ground (and pot "low")	
6	Pot wiper	Controls remote level
7	Pot "hi"	
8	Remote LED	
9	Remote switch	

NOTE*: Pins 3 and 4 are internally shorted to detect CRC errors. CRC (Cyclical Redundancy Checksum) verifies whether the transmission of data is error free or not. When a CRC error is detected, it is added to the CRC log. The CRC log on the LCD display is viewable when the DP562 is in Status mode (see Section 4.9, Status Mode).*

The circuit in Figure 2-1 can be used for monitoring status and controlling the analog output level remotely.
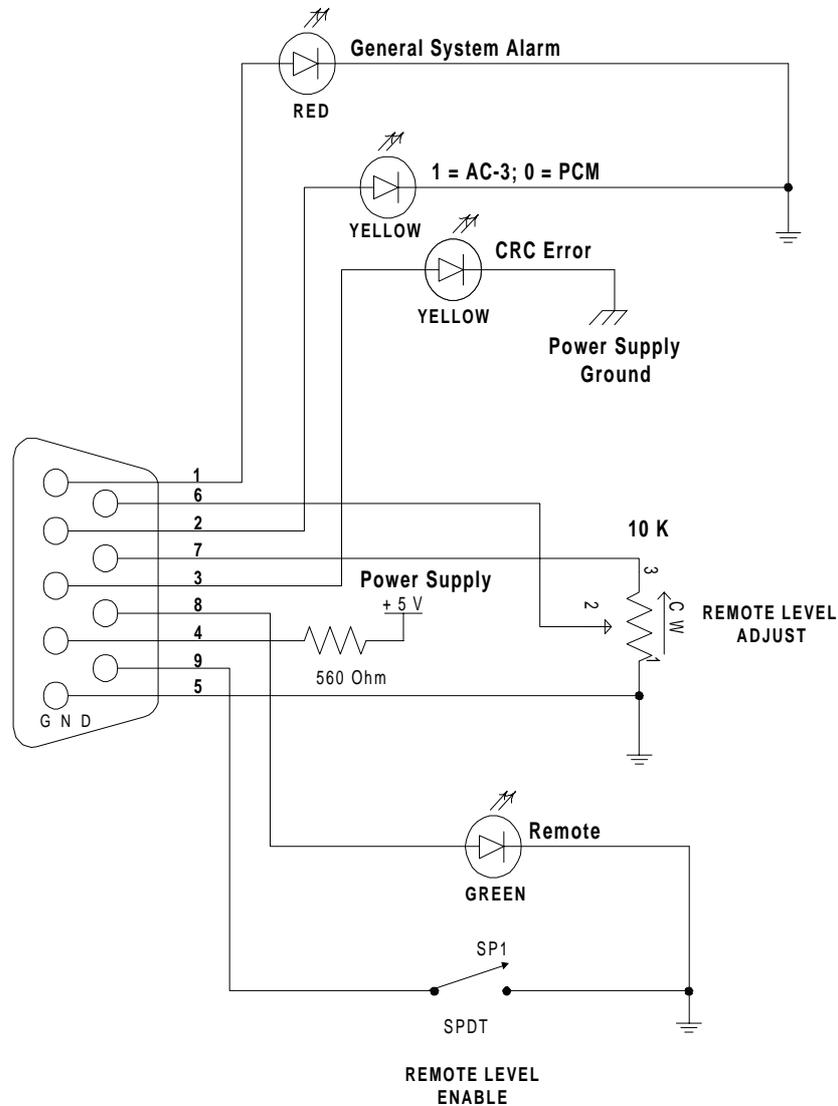
Figure 2-1. Remote Circuit Example

2.4   Serial Input / Output (I/O) Connections

The DP562 is categorized as a data communications equipment (DCE) device. The serial I/O connector of the DP562 can accept commands from a personal computer via an RS-232 serial interface protocol with a minimum two-wire connection (half-duplex). The serial I/O port supports a ten-bit asynchronous serial data format (one start bit, one stop bit, eight data bits, no parity, and 9600 baud). The serial I/O port is used not only as the serial interface but also for software upgrades.

When connected to a computer, the serial I/O connector can transmit and receive data to control the front-panel functions of the DP562. This allows control of the DP562 by the front-panel buttons or by a remote control. The front panel always reflects the current operating configuration of the DP562. The serial connector transmits all status changes back to the computer.

The Serial I/O, 9-pin D-connector is also used as a computer interface port to perform software upgrades for the DP562. For future software upgrades, connect the serial port of the computer to the serial 9-pin D-connector of the DP562. (See *Section 6*, *Future Software Upgrades*, for more information on upgrading the DP562.

Serial I/O, 9600 b/s
Female 9-pin D-connector on the rear panel.

Pin	Connection	Comment
1	DCD	Direct connection to DTR, DSR, and CTS
2	RX	Asynchronous data out
3	TX	Asynchronous data in
4	DTR	Direct connection to DCD, DSR, and CTS
5	GND	Signal ground
6	DSR	Direct connection to DCD, DTR, and CTS
7	NC	
8	CTS	Direct connection to DCD, DTR, and DSR
9	NC	

2.5   Headphone

The headphone connector and volume knob are located on the front panel.

Headphone, ¼" stereo connector
+ 11.5 dBu output into 50 Ωnominal

Pin	Connection
Tip	Left Channel
Ring	Right Channel
Sleeve	Common

2.6   Power

Connect the main power cable to the unit to turn on the DP562.

Powering up the DP562 begins an
initialization sequence. First, all of the
Output Activity LEDs illuminate. Next,
the Output Activity LEDs flash
12 times and the screens depicted in
Figure 2-2 appear on the front panel
display in sequence.

1)
```
▌◖  M O D E L   D P 5 6 2
MULTI CH DECODER
```

2)
```
▌◖  M O D E L   D P 5 6 2
Soft Ver. XXX
```

3)
```
▌◖  M O D E L   D P 5 6 2
I N I T I A L I Z I N G . . .
```
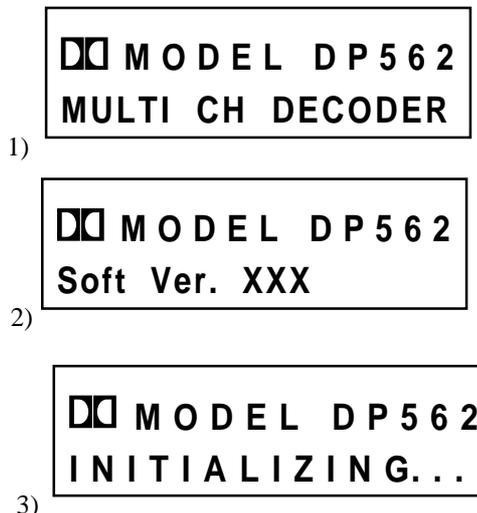
Figure 2-2.
Power-up display

*The "XXX" shown in the center figure represents the
software version.*

Next, the output activity LEDs again
and the screen displays the input status.

If a valid Dolby Digital bitstream is
present in the decoder, the message in
Figure 2-3 will display .

```
AC3 INPUT STATUS
A/BL  DDD  SS   NN  R
```

Figure 2-3.
Input status message with
a valid Dolby Digital bitstream.

***The letters** beneath the input status represent
parameters of the **Dolby Digital bitstream.** Refer to
Section 4.9, Status Mode, for a list and
explanation of all input status messages.*

If a source other than Dolby Digital is
present at the input, the message in
Figure 2-4 will display.

```
AES/EBU STATUS
No AES/EBU Input
```

Figure 2-4.
Input status message without a
Dolby Digital bitstream.

2.7   Default Values

The defaults listed below reflect the state of the DP562 when powered up for the first time. If you change these values, they can be recovered later by choosing "FACTORY RESET" in the **User Presets** of the **Setup** menu (see *Section 5.12*).

2.7.1   Front Panel Defaults

Defaults are indicated in ***bold italics***.

Decoding Mode	***Dolby Digital*** (with Dolby Digital bitstream present) */* ***Stereo*** (with PCM present)
Listening Mode	***Full*** (with Dolby Digital bitstream present) / ***Stereo*** (with PCM present)
Compression Mode	***Line*** (with Dolby Digital bitstream present) / ***None*** (with PCM present)
Test Noise	***Off***
User	***Off***
Setup	***Off***
Dim	***Off***

2.7.2   Defaults in Setup Menu

The following list shows the set of defaults in Factory Setup. (See *Section 5 Setup*.)

Master Level Adjust	**Level**	**0.00 dB**
Output Ch Trims	**Left**	**0.00 dB**
	Center	**0.00 dB**
	Right	**0.00 dB**
	R Surround	**0.00 dB**
	L Surround	**0.00 dB**
	Subwoofer	**0.00 dB**
Assign Channels	**|L|R|C|Sw|Ls|Rs|**	
LFE Monitor Mode	**Auto Select**	
Headphone Mode	**Mono**	
Bitstream Detect	**Reference**	
AES/EBU Ch Sel	**Auto Detect**	
Stream Select	**Stream 0**	
Srnd/Cntr Delay	**Center=0 ms**	
	AC3 Srnd=0 ms	
	PL Srnd=15 ms	
Bass Redirection	**None**	
Test Noise	**Pink**	
User Presets	**Recall=A**	
Contrast Adjust	**Factory set**	

Chapter 3
Calibration

3.1   Importance of Calibration

Calibrating your playback system to establish a balance between all five main channels and to ensure that all speakers play back at the same level is very important. For proper calibration, it is necessary to use a sound pressure level (SPL) meter when measuring a particular playback level. By correctly calibrating your mixing and playback monitor system so that dialogue levels are at universally referenced listening levels, you will know that your mix will be roughly at the same volume as other productions upon playback.

3.2   Playback Levels

Use an SPL meter and the built-in Pink Noise Generator to set your system's playback level to a particular reference level.

There are three options to adjust monitor system playback levels:

- Amplifier gain trim controls

- Mixer's group outputs (one for each of the L, C, R, Ls, Rs and Sw channels)

- DP562's Output Level Trim controls (only if using the analog outputs)

The best option is to use your amplifier gain controls to set proper playback levels. This option allows you to maintain optimum signal-to-noise performance from the DP562 and your mixer. If you decide to use either the console's group outputs or the DP562's Output Level Trim controls, then both may sacrifice signal-to-noise ratio. Use an SPL meter and the built-in Pink Noise Generator to set your system's playback level to a particular reference level.

Pink noise readings depend on the type of meter used to set level (and, strictly speaking, on the bandwidth of the pink noise signal). In film practice, pink noise level is read with a true VU meter, or a meter display with a VU characteristic. If reference level is specified as "0 VU", where "0 VU" corresponds to -20 dBFS for a digital recording medium, then pink noise should be set to "0 VU" on the console meter or meter display, and the SPL set accordingly.

NOTE: *Many, if not most, modern consoles have peak-reading meters or meter displays. Pink noise that reads at reference level using a true VU meter will read from 8–12 dB higher on a peak reading meter or meter display. If your console has switchable meter characteristics, be sure to select the "VU" mode when setting pink noise levels.*

For film work, pink noise at reference level should produce a sound pressure level (SPL) of 85 dBC for each of the front channels (left, center, and right). Each surround channel should produce a sound pressure level of 82 dBC (the lower surround level is specific to film-style mixing rooms).

For television work, pink noise at reference level is typically set to produce an SPL ranging from 79 dBC to 82 dBC for each of the main five channels. The lower reference level for television is due to the lower average listening levels used by the consumer (typically 70–75 dBC). Since the reference listening level used can dramatically effect the balance and intelligibility of the mix, it is important to consider both the level at which a program was mixed as well as the typical listening level in the home for the same program.

For music mixing, each speaker channel should be set to the same SPL (the same as television mixing). There is no standard practice for reference levels for music mixing. Some engineers prefer to mix louder than others do, but if the levels between channels are correct, the overall level is not as important. However, as stated previously, consideration of the typical consumer listening level is always good practice.

When mixing for television or music in small mixing rooms (i.e., remote recording trucks), the surround channel is generally set 2 dB lower than the front channels (for example, 80 dB in front and 78 dB in back). This takes into account the short distance to the surround speakers. Experience has shown that this setting makes the sound in the home environment very close to the sound heard by the mixer.

3.2.1   Sound Pressure Level Meter

To properly calibrate speaker levels, an SPL meter is necessary. A suitable and relatively inexpensive meter is readily available from Radio Shack® (Tandy Electronics outside of North America). Since the relative level between channels is more important than absolute level, the accuracy of this meter is sufficient for channel balancing. An inexpensive meter can be left in the control room for quick level checks.

3.2.2   Taking a First Measurement

1. Before you turn on the internal Pink Noise Generator, be sure that your playback system has been set to a moderate listening level. Adjust your amplifiers, self-powered speakers, or mixer only. Beware that if the playback level is very high, you may risk damaging your speakers or possibly your hearing.

2. Press **[TEST NOISE]** (you will learn more about this button in *Section 4*, *Operation*). This will cycle pink noise automatically between the Left, Center, Right, Right Surround, and Left Surround channels—remaining two seconds at each output before moving onto the next. If you do not hear pink noise in any of the five main channels as the noise cycles through that channel, check your system connections and settings.

3. Ensure that you are sitting in your normal, proper reference listening or mixing position. Set the SPL meter to "C" weighting and "slow" response.

4. Facing the front speakers, hold the SPL meter at chest level, with the microphone facing up at an angle of approximately 45 degrees to the center speaker. Keep the meter at arm's length to prevent measuring audio that may reflect from your body. You should be able to take SPL readings as you look down at the meter.

5. Keep the SPL meter in this position. Make sure that the meter is aimed at the center speaker as you take readings for the left and right speakers.

6. When taking the SPL readings for the left surround or right surround speakers, keep the meter at the same angle and position as you did for the front speakers. Turn your body 90° from the center speaker towards the wall closest to the surround speaker you are measuring. This will minimize "shadowing" or obscuring the meter with your body.

3.2.3   Bass Redirection

Before you calibrate the monitoring system, you should decide whether bass redirection of low frequencies is required. The DP562 decoder is capable of redirecting low-frequency information below a selected frequency (80, 100, or 120 Hz) to the channels capable of reproducing them. This is similar to the normal/wide mode for the center channel in a consumer Dolby Surround Pro Logic decoder.

For example, if you have five satellite speakers and a subwoofer, you could redirect the low frequencies from the five main channels to the subwoofer output. If you are using small center and surround speakers, you can redirect low frequencies from those channels to the left, right, or subwoofer outputs. If no subwoofer is available, the low–frequency effects signal (LFE) can be redirected to the left and right channel outputs of the DP562. For a list of all available Bass Redirection options, refer to *Section 5.10*, *Bass Redirection*.

To set up bass redirection:

1. Press [**SETUP**].

2. Scroll through the **Setup** menu, using the [↑] and [↓] buttons, to find **Bass Redirection**.

3. Select **Bass Redirection** by pressing [**ENTER**]**.**

4. Press [**ENTER**] again to display the current **Bass Redirection** option (default is **None**). You may use the [↑] button to return to previous menu level at this time.

5. Press [**ENTER**] again. (You will see a black flashing cursor.)

6. Use the [↑] and [↓] buttons to select the set of channels for bass redirection. The adjustment takes effect and returns you to the *Status Level* as soon as [**ENTER**] is pressed.

7. Next, you may wish to select the Bass Redirection Crossover frequency. Scroll down to **Bass Redirection X / OVER**.

8. Press [**ENTER**]. (You will again see a black flashing cursor.)

9. Press the [↑] and [↓] buttons to select the level of crossover you desire: 80 Hz, 100 Hz, or 120 Hz. Press [**ENTER**] to save the configuration.

10. Press [**SETUP**] on the front panel to exit the **Setup** menu and return to normal operation.

There are two settings for test noise in the DP562: "PINK" or "FILTERED." If no bass redirection is chosen ("NONE" - default), calibration of the monitoring system should be made using the "PINK" test noise option. This option uses a full-band pink noise signal.

If a Bass Redirection option other than "NONE" is selected, calibration of your monitoring system should be made using the "FILTERED" test noise option. This will compensate for the absence of low frequencies in your main monitoring speakers. These selections can be performed in the **Setup** menu. See *Section 5.10* for complete instructions.

3.3   Individual Channel Calibration

3.3.1   Main Channels

The following steps describe the procedure for systems that have individual amplifier gain controls, self-powered speaker gain controls, or adjustable console group outputs (not preferred). If using a console, it is advisable to set the master monitor volume control to a known marked position before proceeding with calibration. This allows an easy return to the calibrated listening level at any time.

1. Set the playback level for the Left channel. Press [**SETUP**] on the front panel of the DP562 to enable the **Setup** menu screen. Scroll down using the [↑] and [↓] buttons on the front panel to **Output Ch Trims** and press [**ENTER**] on the front panel. Scroll down again using the [↑] and [↓] buttons to **LEFT** and press [**ENTER**]. Press [**TEST NOISE**] on the front panel. Test noise ("PINK" or "FILTERED") will appear at the left speaker.

2. Check the SPL meter and take note of the level. Adjust that channel's amplifier gain control (or associated mixer group output or self-powered speaker gain control) up or down as necessary (to 85 dB or 79 dB). Unless you have an assistant, you may need to leave your sitting position to do this. If so, be sure to return to the same position, with the SPL meter in the same position.

3. Next, return to Step 1 and repeat the procedure for the remaining Center, Right, Right Surround, and Left Surround channels.

4. As a final check, exit the **Setup** menu and press **[TEST NOISE]**. Pink noise generation will return to automatic sequencing to all channels selected. Using the SPL meter (and your ears!) confirm that each speaker sounds equally loud at your reference position and that each speaker plays back at the exact reference level (85 dBC or 79 dBC). If there are any discrepancies, repeat these steps until everything is properly adjusted.

3.3.2   Subwoofer

The test noise generator uses band-limited pink noise for setting the level of the subwoofer. The noise signal is low-pass filtered at 120 Hz. To properly calibrate the subwoofer, a real time analyzer (RTA) is required. If an RTA is not available, you can approximate the settings with an SPL meter.

When using an RTA, proper alignment of the subwoofer requires setting the subwoofer level 10 dB higher than the center channel level, measured within the subwoofer bandwidth (typically 25–120 Hz).

For example, measure the center channel average value of each third-octave band between 25–120 Hz and set as the 0 dB datum on the RTA display. Then adjust the subwoofer level so that the average value of each third-octave band between 25–120 Hz is +10 dB higher.
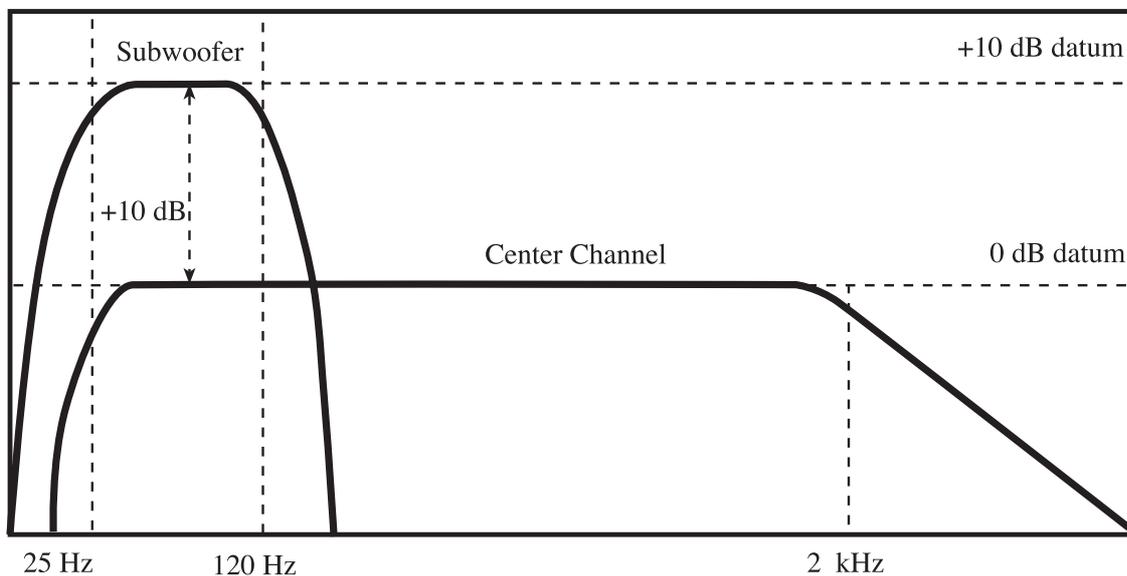
Figure 3-1. Real time analyzer (RTA) display.

If an RTA is not available, setting the subwoofer channel 4 to 6 dB higher (e.g. 89–91 dB for the subwoofer channel when the center channel measures 85 dB), will give an approximate calibration using an SPL meter. This level varies with the quality of your meter.

For future reference, if you have set up the level with an RTA, measure the level with your SPL meter and take note of the reading. You can then use this measured value for quick checks of the subwoofer calibration in the future.

To calibrate the subwoofer:

1. Press [**SETUP**] on the front panel to enable the **Setup** menu screen.

2. Scroll down using the [↓] button on the front panel to **Output Ch Trims.**

3. Press [**ENTER**].

4. Using the [↑] and [↓] buttons, scroll down to "**SW**" and press [**ENTER**].

5. Press [**TEST NOISE**] on the front panel to activate the low-pass filtered noise. Adjust the trim with the [↑] and [↓] buttons for the correct level in dB.

6. Press [**ENTER**] to save the adjustment.

7. To exit the **Setup** menu and to return to normal operation, press [**SETUP**]. For more information on the **Setup** menu, refer to *Section 5, Setup*.

3.3.3   Time Delays for Surround and Center Channels

In addition to setting the proper monitoring levels, be sure that the sound from each speaker arrives to your listening position at the correct time.

To achieve this, the delay times of the center and surround channels need to be adjusted. Channel delay is determined by calculating the distances from each speaker to the listening position, as shown in *Figure 3-2*.
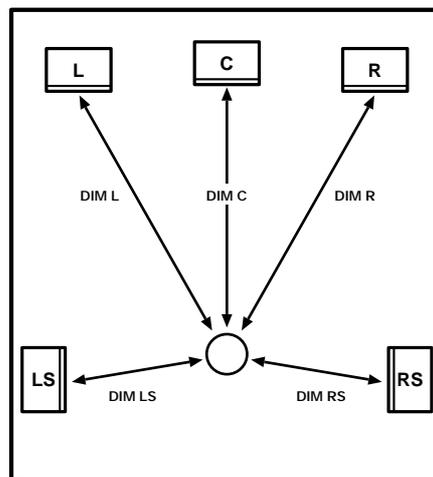
Figure 3-2. Speaker-to-listener dimensions.
"DIM" signifies distance

In the case of Dolby Surround Pro Logic, a sound common to both the front channel and the surround channel should arrive at the listening position a fraction of a second before the surround channels. This takes advantage of the Haas effect (also known as the precedence effect): When two similar sounds arrive at our ears at slightly different times, our brain tends to focus on the sound arriving first and ignore the second. Dolby Surround takes advantage of this effect to reduce the perceived crosstalk between the front and rear channels. A time arrival difference of 10–20 milliseconds is adequate to achieve this effect.

Because Dolby Digital is a discrete system, there is no crosstalk between channels. The delay times required for Dolby Digital are therefore different from Dolby Pro Logic, although related.

DP562 surround delays are set up separately for Dolby Pro Logic and Dolby Digital, while the setup for the center delay is the same for both modes.

Surround Delay for Dolby Pro Logic

To calculate the required delay in Dolby Pro Logic mode, you need to measure the distance from your referenced listening position to each of the five speakers:

- Left speaker (L) –
- Center speaker (C) –
- Right speaker (R) –
- Left surround speaker (Ls) –
- Right surround speaker (Rs) –

Make all of these measurements in feet. (If measuring in meters, then multiply the metric measurements by three to get the approximate distance in feet.)

The required delay settings for the room can now be calculated. Calculate the delays using the formula below:

Use the shorter distance to the Left Surround or Right Surround speaker for the value "S."

Distance from C – Distance from S + 15 = Surround delay for Dolby Surround Pro Logic

For example, if the listener's position is 15 feet from the center speaker and 10 feet from the surround speakers, the delay setting for the rear channels will be 20 milliseconds for Dolby Surround Pro Logic. See example below:

15 (*Distance from C*) – **10** (*Distance from S*) + 15 = **20** (*Surround delay for Pro Logic*)

The Dolby Pro Logic delay range can be adjusted from 10 to 75 milliseconds in one-millisecond steps. If your calculation is less than the minimum value available, then use the minimum value.

To set the Pro Logic Surround delay on the DP562:

1. Press [**SETUP**] on the front panel. This takes you to the **Setup** menu screen.

2. Scroll down the **Setup** menu using the [↓] button. Select **Srnd/Cntr Delay** and press [**ENTER**].

3. Using the [↑] and [↓] buttons, scroll through the **Srnd/Cntr Delay** menu until you find **PL Srnd**.

4. Press [**ENTER**], then use the [↑] and [↓] buttons to display the delay value in milliseconds. Press [**ENTER**] again to save the adjustment.

5. Press [**SETUP**] to exit the **Setup** menu and to return to normal operation. For more information on the **Setup** menu, refer to *Section 5*.

Surround Delay for Dolby Digital

Determining channel delays for Dolby Digital is similar in concept to Dolby Pro Logic, but with some important differences. Since Dolby Digital delivers discrete signals for each channel, there is no leakage or crosstalk between channels. There is no need to delay the surround channels to take advantage of the Haas effect (precedence effect). The surround and front channel signals should arrive at listening position at the same exact time (*coincident* arrivals). The DP562 Dolby Digital mode consequently uses approximately 15 milliseconds less delay than the Dolby Pro Logic mode for the same speaker and seating geometry. Surround channel delay can be adjusted at increments of one milliseond.

To calculate the surround delay for Dolby Digital, measure the distance from your listening position to each of the three speakers:

- Left speaker (L) or Right speaker (R) –
- Center speaker (C) –
- Nearest surround speaker (Ls or Rs) –

Make all of these measurements in feet. (If measuring in meters, then multiply the metric measurements by three to get the approximate distance in feet.)

The required delay settings for the room can now be calculated. Calculate the delays using the formula below:

Distance from C – Distance from S = Surround delay for Dolby Digital
"S" represents the distance from the Left Surround or Right Surround speaker.

For example, if the listener's position is 15 feet from the center speaker and 10 feet from the surround speakers, the delay setting for the surround channels would be five milliseconds for Dolby Digital:

15*(Distance from C)* **– 10** *(Distance from S)* **= 5** (*Surround delay for Dolby Digital)*

The Dolby Digital delay range can be adjusted from 0 to 50 milliseconds in one-millisecond steps:

To set the Dolby Digital Surround delay:

1. Press [**SETUP**] on the front panel. This takes you to the **Setup** menu.

2. Scroll down the **Setup** menu using the [↓] button. Select **Srnd/Cntr Delay** and press [**ENTER**].

3. Using the [↑] and [↓] buttons, scroll through the **Srnd/Cntr Delay** menu until you find **AC-3 Srnd**.

4. Press [**ENTER**], then use the [↑] and [↓] buttons to display delay value in milliseconds. Press [**ENTER**] again to save the adjustment.

5. Press [**SETUP**] to exit the **Setup** menu and return to normal operation. See *Section 5*, *Setup* for more information.

Time Delay for Center Channel

The concept of "coincident arrival" can also be applied to the adjustment of the center channel in Dolby Digital mode. The DP562 factory default for the center delay is 0 milliseconds. If the center speaker can be placed at the same distance from the listener as the left and right speakers (in an arc) to achieve equal path lengths, i.e., *Distance C = Distance L = Distance R*, then no delay is required. If the center speaker is placed closer to the listening position than the left and right speakers (or further back), then time delay can be added (or subtracted) to the center channel signal to bring it into acoustic alignment with the left and right speakers. This will electronically move the speaker to its proper position in the room. For example, if *Distance C* is two feet less than *Distance L*, center delay should be set to two milliseconds. If *Distance C* is two feet more than *Distance L*, center delay should be set to -2 milliseconds.

The center channel delay range is -3 to +5 milliseconds in one-millisecond steps. To make the C-time delay negative, the decoder is actually setting C delay to zero and adding delay of up to three milliseconds to the left, right, left surround, and right surround outputs. This keeps the surround and front arrival times aligned.

To set the Center delay:

1. Press [**SETUP**] on the front panel. This takes you to the **Setup** menu.

2. Scroll down the **Setup** menu using the [↓] button. Select **Srnd/Cntr Delay** and press [**ENTER**].

3. Using the [↑] and [↓] buttons, scroll through the **Srnd/Cntr Delay** menu until you find **Center**.

4. Press [**ENTER**], then use the [↑] and [↓] buttons to dispay the correct delay value in milliseconds. Press [**ENTER**] again to save the adjustment.

5. Press [**SETUP**] to exit the **Setup** menu and to return to normal operation.

Chapter 4
Operation

4.1 Navigation and Enter Buttons

The [↑] and [↓] buttons enable you to navigate through the **Status** menu and other menu screens shown in the liquid crystal display (LCD) of the DP562. Pressing down on either button allows you to scroll through a menu.

The [**ENTER**] button allows you to change parameters of the DP562 during menu navigation.

4.2 Decoding Modes

The DP562 supports four decoding modes:

- Dolby Digital

- Dolby Pro Logic

- Dolby Digital + Dolby Pro Logic

- Pass-through

Each mode and how it can be enabled using the front panel buttons is described below.

4.2.1 Dolby Digital Mode

When a Dolby Digital bitstream is present at the input of the DP562, Dolby Digital encoded data is decoded into PCM digital and multichannel analog outputs. Dolby Digital decoding can be enabled automatically or manually. To select the appropriate mode, "DISABLED" "REFERENCE" or "SILENT SWITCH," refer to **Bitstream Detect** in *Section 5.6 Setup*.

The default setting for **Bitstream Detect** is "REFERENCE." When a Dolby Digital bitstream is present, the DP562 automatically selects the Dolby Digital mode and the Dolby Digital LED steadily illuminates. If no Dolby Digital bitstream is present, the dedicated "Dolby Digital" LED flashes periodically and the DP562 passes-through any incoming PCM audio.

Automatic **Bitstream Detect** is helpful when your program material alternates between Dolby Digital, Dolby Pro Logic, and PCM input signals. In addition, if the DOLBY SRND MODE is indicated (flagged) as either "DOLBY SRND" or "Not Indicated" in a (2/0) bitstream, Dolby Pro Logic decoding engages automatically.

If you prefer to select the Dolby Digital mode manually, set **Bitstream Detect** to "DISABLED" in the **Setup** menu. Next, press [**DOLBY DIGITAL**] on the front panel. The "Dolby Digital" LED illuminates steadily, signifying that the Dolby Digital mode is enabled. Pressing the button again disables the Dolby Digital decode mode.

> ⚠️ *WARNING: When Bitstream Detect is set to "DISABLED" and the Dolby Digital mode is not selected, the unit will output un-decoded audio to the outputs at full scale if a Dolby Digital bitstream is present. If the playback level is very high, you risk damaging your speakers or possibly your hearing.*

NOTE*: When Bitstream Detect is set to "SILENT SWITCH" with a 48 kHz clock input, approximately 7 ms of buffering is implemented with AES/EBU, and approximately 20–30 ms with SPDIF. The minor delay is necessary to minimize audio artifacts while switching between Dolby Digital and PCM.*

4.2.2   Dolby Pro Logic Mode

Select Dolby Pro Logic decoding by pressing [**DOLBY PRO LOGIC**] when a Dolby Surround encoded PCM signal (Lt/Rt) is present at the input of the DP562.

The dedicated "Dolby Pro Logic" LED illuminates to indicate that the mode is enabled (ensure that the "Dolby Digital" LED is off). The DP562 decodes the incoming Dolby Surround encoded program to left, center, right, and surround outputs.

Pressing [**DOLBY PRO LOGIC**] again disables the Dolby Pro Logic mode.

4.2.3   Dolby Digital + Dolby Pro Logic Mode

To *manually* decode in the Dolby Digital + Dolby Pro Logic mode, set **Bitstream Detect** to "DISABLED" in the **Setup** menu and press [**DOLBY DIGITAL**] and [**DOLBY PRO LOGIC**]. Both "Dolby Digital" and "Dolby Pro Logic" LEDs illuminate.

To automatically decode in the Dolby Digital + Dolby Pro Logic mode when Dolby Digital bitstream is present, set **Bitstream Detect** to "REFERENCE" or "SILENT SWITCH" and press [**DOLBY PRO LOGIC**]. The "Dolby Digital" LED will illuminate only when a Dolby Digital input is detected, while the "Dolby Pro Logic" LED will illuminate steadily.

The Dolby Digital + Dolby Pro Logic mode is automatically selected if Dolby Surround is indicated (flagged) in a Dolby Digital 2 channel (2/0) bitstream.

When the Dolby Digital + Dolby Pro Logic Mode is enabled, the bitstream is first decoded in the Dolby Digital mode in a Lt/Rt PCM bitstream. This bitstream is then decoded in the Pro Logic mode and directed to the analog and digital outputs. For example, if 3/2 material is present in the bitstream, the Dolby Digital decoder downmixes to 2/0 (Lt/Rt) before it passes through the Dolby Pro Logic decoder to the outputs.

NOTE*: When the Listening mode is set to "Stereo," Lt/Rt is present on the left and right analog and digital output connectors.*

*If **Bitstream Detect** is set to "REFERENCE or "SILENT SWITCH" the Dolby Digital LED will flash periodically when no valid Dolby Digital bitstream is present at the input.*

4.2.4   Pass-through Mode

In the Pass-through mode, the DP562 functions as a bypass device with DAC (Digital-to-Analog Converter). Only *Stereo* and *Mono* listening modes are supported.

To enable the Pass-through mode:

> Press [**DOLBY DIGITAL**] and [**DOLBY PRO LOGIC**] to turn off both modes. Their respective LEDs will turn off.
>
> (Enabling the Pass-through mode, as mentioned above, only applies when the bitstream detect is set to "DISABLED".)

NOTE*: If **Bitstream Detect** is set to "REFERENCE" or "SILENT SWITCH," the "Dolby Digital" LED will flash periodically.*

4.3   Listening Modes (Downmix)

Three decoding modes—Dolby Digital, Dolby Pro Logic, and Dolby Digital + Dolby Pro Logic—support five listening modes. Simply press the button labeled with the listening mode you desire.

- **Full**: All channels (dependent upon input channel mode and decoding mode selected) are reproduced in their appropriate location. (factory default)

- **3 Stereo**: Left surround signal is sent to left front speaker; right surround signal is sent to right front speaker

- **Phantom**: Center channel signal is sent to the front left and right speakers

- **Stereo**: Left surround signal is sent to left front speaker, right surround signal is sent to right front speaker, center channel signal is sent to the left and right front speakers. LFE channel is not included.*

- **Mono**: All channels signals are sent to the center speaker. LFE channel is not included.*

  * *When decoding Dolby Digital bitstreams, if "LFE MONITOR MODE" in the setup section is set to "AUTO" (factory default), the Low Frequency Effects (LFE) channel will be muted when "Stereo" or "Mono" listening modes are selected. In addition, LFE channel information is never distributed or downmixed to any other speakers when Pro Logic decoding is selected. LFE channel information may only be distributed to the front left and right speakers when appropriate bass management is selected in the setup menu.*

NOTE*: The Pass-through mode supports Stereo (factory default) and Mono listening modes, and two-channel outputs.*

The combination of the defined listening mode, the input bitstream audio-coding mode, and the active decoding mode, determine the output (downmix) mode.

4.4   Compression Modes

With Dolby Digital or Dolby Digital + Dolby Pro Logic decoding, the following compression modes are valid:

- **None**: A professional mode

  No compression and no dialogue normalization.

- **Custom**: A professional mode

  Adjustment of low- and high-level scaling of dynamic range compression parameters is allowed.

  Enabling and disabling of dialogue normalization are allowed.

- **Line**: A consumer emulation mode

  Dialogue normalization always on.

  Adjustment of low-level scaling of dynamic range compression parameter is allowed.

  Adjustment of high-level scaling of dynamic range compression parameter is allowed except during downmixing (factory default).

- **RF**: A consumer emulation mode

  Dialogue normalization always on.

  Dynamic range compression, and heavy compression always on.

  Adjustment of low- and high-level scaling of dynamic range compression parameters is not allowed.

See *Appendix C* for a detailed description of dialogue normalization, dynamic range compression, Line, RF, and Custom compression modes.

NOTE*: For **None** and **Custom** compression mode, an 11dB attenuation is applied during downmixing to protect against saturation conditions.*

You are responsible for providing the gain to bring the signals back to normal levels if utilizing digital outputs. The 11dB attenuation is recovered in analog outputs.

4.4.1   Custom Compression Setup

Custom Compression allows you to enable or disable dialogue normalization, and adjust hi-level cut and low-level boost compression parameters.
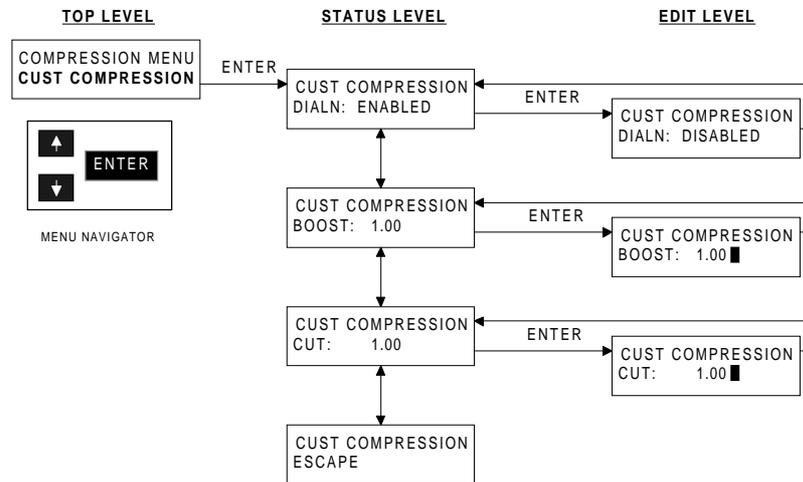
To adjust the parameters in the Custom menu:

1.   Press [**CUSTOM**] for three seconds until the **Custom** menu appears on the LCD screen.

     You can escape from this menu and return to the **Status** screen by either (1) waiting for approximately five seconds, or (2) pressing [**TEST NOISE**] or [**SETUP**].

2.   Press [**ENTER**] to get into the *Status Level*[*] to view the current parameter settings. Use the [↑] and [↓] buttons to scroll through the menu to select which parameter you need to modify**: Dialog Norm**, **Adjust Lo Level**, **Adjust Hi Level**, or to return to a previous menu level.

3.   Upon selecting which adjustment you need to modify, press [**ENTER**] to initiate the *Edit Level*. (The *Edit Level* is defined by a black solid flashing cursor.)

4.   Once in the *Edit Level*, if you have selected:

     •   **Dialog Norm**: Press [**ENTER**] to toggle between "ENABLED" or "DISABLED." The adjustment takes effect as soon as the button is pressed.

     •   **Adjust Lo Level** or **Adjust Hi Level**: Use the [↑] and [↓] buttons to adjust the level of "BOOST" or "CUT" from "1.00" (no level) to "0.00" (maximum) in 0.1 increments. The adjustment takes effect as soon as [**ENTER**] is pressed.

NOTE*: If the menu is enabled but not in use for approximately five seconds, the display will revert to the Status screen.*

---

[*] For an explanation of *Status Level* and *Edit Level*, refer to ***Section 5*** of this manual.

4.4.2   Line Compression Setup

Line Compression allows scaling of low-level boost and high-level cut parameters in *Full* listening mode. When downmixing, low-level boost parameters can be adjusted. High-level cut parameters are automatically fixed by the DP562 to avoid saturation. The Line compression mode always provides dynamic compression and dialogue normalization.
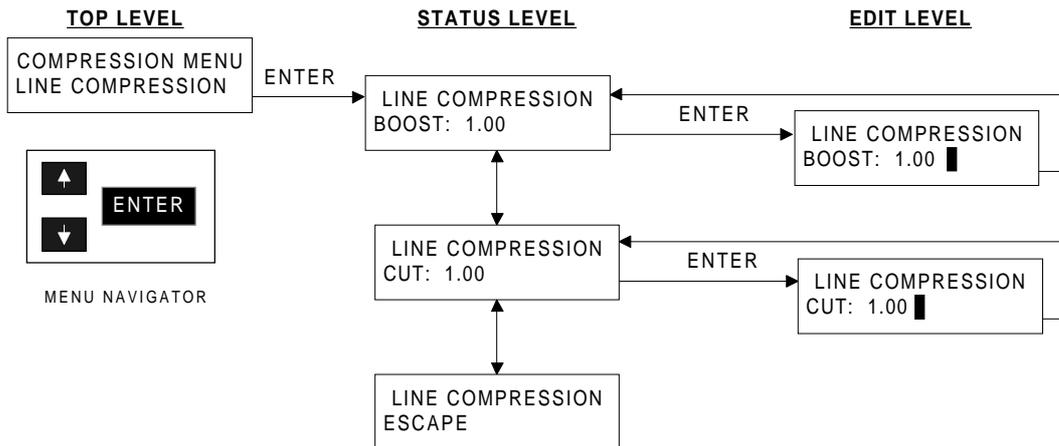
To adjust the parameters in the Line menu:

1. Press [**LINE**] for three seconds until the **Line** menu appears on the LCD screen.

    You can escape from this menu and return to the **Status** screen by either: (1) waiting for approximately five seconds, or (2) pressing [**TEST NOISE**] or [**SETUP**].

2. Press [**ENTER**] to get into the *Status Level*[*] to view the current parameter settings. Use the [↑] and [↓] buttons to scroll through the menu and select the level you need to modify: "BOOST" or "CUT,"or to return to previous menu level.

3. Upon selecting which adjustment you need to make, press [**ENTER**] to initiate the *Edit Level*. (The *Edit Level* is defined by a black solid flashing cursor.)

4. Once in the *Edit Level*, use the [↑] and [↓] buttons to adjust the amount of **Boost** or **Cut**. (The default setting for both is "1.00"). The adjustment takes effect as soon as [**ENTER**] is pressed.

NOTE*: If the menu is enabled but not used for approximately ten seconds, the display will revert to the Status screen.*

---

[*] For an explanation of *Status Level* and *Edit Level*, refer to **Section 5** of this manual.

4.5   Test Noise

The DP562 provides a built-in test noise sequencer that you turn on by pressing [**TEST NOISE**] on the front panel. Enabling [**TEST NOISE**] automatically sends a sequence of band-limited pink noise (400–1500 Hz) or full-bandwidth pink noise sequentially to each activated channel at two-second intervals. The subwoofer channel is always band-limited to 20–120 Hz. The DP562 retains the current decoding mode when you enable **Test Noise**.

The selected listening mode determines the number of channels activated. For example, if the listening mode is *Full*, the generator will sequence through all six channels. If the listening mode is *Stereo*, the generator sequences between the left and right channels only.

You can also generate test noise for a selected channel. Press the [**SETUP**] button on the front panel of the DP562 to enable the **Setup** menu screen. Scroll down using the [↑] and [↓] buttons on the front panel to **Output Ch Trims** and press the [**ENTER**] button on the front panel. Now select **Test Noise**. Scrolling up or down the output channel trims using the [↑] and [↓] buttons, test noise ("PINK" or "FILTERED") will generate from the selected speaker.

4.6   User

The **User** menu allows you to assign one of the **Setup** menu parameters to the [**USER**] button. The [**USER**] button then acts as a shortcut to one of the **Setup** menu functions listed in the *Edit Level* below.
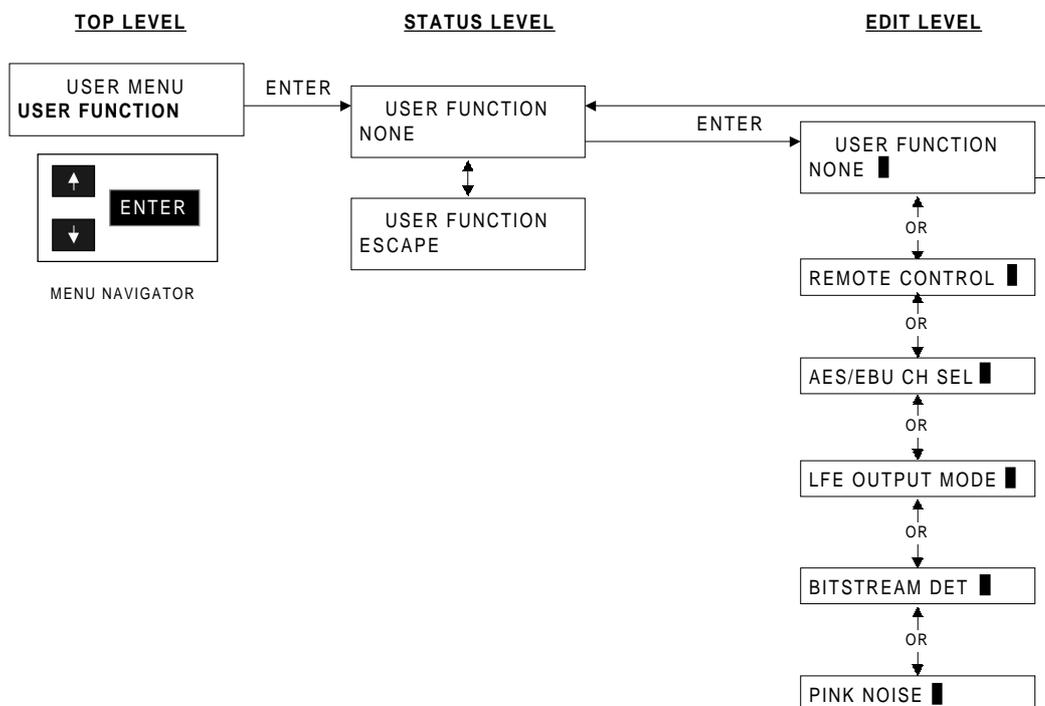
NOTE*: If the menu is enabled and not in use for approximately ten seconds, the display will revert to the Status screen.*

To create a shortcut in the User menu:

1. Press [**USER**] for three seconds until the menu appears on the LCD screen.

    You can escape from this menu and return to the **Status** screen either: (1) waiting for approximately five seconds, or (2) pressing [**TEST NOISE**] or [**SETUP**].

2. Press [**ENTER**] to get into the *Status Level**[*]* and view the current setting. Use the [↑] and [↓] buttons to scroll through the status or to return to the previous menu level.

3. Upon selecting the current parameter, press [**ENTER**] to initiate the *Edit Level*. (The *Edit Level* is defined by a black solid flashing cursor.)

4. Once in the *Edit Level*, use the [↑] and [↓] buttons to select an option. The adjustment takes effect as soon as [**ENTER**] is pressed.

5. Use the [↑] and [↓] buttons to scroll to "ESCAPE." Press [**ENTER**] to return to the **User** menu.

4.7   Output Activity LEDs

Six LEDs, located to the right of the 14 front-panel buttons, indicate the digital output (unity gain output from the DSP) activity level for each channel: Left, Center, Right, Left Surround, Right Surround, and Subwoofer.

LED activity:

> *Off*: Input not present or not selected for playback in listening mode
>
> *Yellow*: Input present and selected for playback in listening mode
>
> *Green*: Signal present with a level above –60 dBFS
>
> *Red*: Signal present with a level above –0.1 dBFS (clipped)

NOTES: *Ls and Rs LEDs will illuminate simultaneously to indicate the presence of mono surround channel in Pro-Logic decoded signal.*

The SW LED will indicate the potential for audio if Bass Redirection is appropriately applied even if the LFE input channel has not been selected.

4.8   Dim

Pressing [**DIM**] attenuates all analog outputs by 20 dB. Headphone and digital outputs will not be affected.

---

[*] For an explanation of *Status Level* and *Edit Level*, refer to **Section 5** of this manual.

4.9   Status Mode

As mentioned in **Section 2**, once the system has completed its power-up sequence, the DP562 is placed into **Status** mode. The **Status** menu, which is shown in the front-panel LCD, displays any AES/EBU digital bitstream information according to the *ATSC Digital Audio Compression Standard*. (To view the most current ATSC standards please refer to our web site, *www.dolby.com* under the "Technical Information" link.)

4.9.1   Status Screen for A Valid Dolby Digital Bitstream (with or without Dolby Pro Logic Enabled)

When valid Dolby Digital information is present in the AES/EBU bitstream, the initial status screen in *Figure 3-1* is displayed on the LCD. In addition, the Status mode displays several other bitstream parameters that convey information about how the program was prepared by a Dolby Digital encoder. The following diagrams depict the status screens of the DP562. The [↑] and [↓] buttons scroll through the parameters.

A/B represents the bitstream audio coding mode. "A" represents the number of front channels, and "B" represents the number of surround channels. "L" indicates whether the LFE channel is present.

AC3 INPUT STATUS
A/BL   DDD   SS   NN R

Channel Mode
Data Rate
Sample Rate
Dolby Digital Parameters

R represents the currently selected Dolby Digital bitstream (IEC 1937).

DDD stands for the data rate in kb/s.

SS denotes the sample rate in kHz.

NN indicates the dialog normalization level in dB (in -dB relative full scale).

Figure 4 - 1. Initial status screen for a valid Dolby Digital bitstream.

The following table lists all eight bitstream audio-coding modes (A/B):

Audio Coding Mode	Encoded Channels
1+1	L (Ch 1), R (Ch 2)
1/0	C
2/0	L, R
3/0	L, C, R
2/1	L, R, Mono Surround
3/1	L, C, R, Mono Surround
2/2	L, R, Ls, Rs
3/2	L, C, R, Ls, Rs

| BITSTREAM MODE |
| Complete Main |

or

| Main M & E |

or

| Assc Visual Imp |

or

| Assc Hear Imp |

or

| Assc Dialogue |

or

| Assc Commentary |

or

| Assc Emergency |

or

| Main Sv Karaoke |

or

| Assc VoiceOver |

Bitstream mode indicates the type of data information "service" in the Dolby Digital bitstream.

| CENTER MIX LEVEL |
| -3.0 dB |

or

| -4.5 dB |

or

| -6.0 dB |

| Not Active |

Center Mix Level indicates the normal downmix level of the center channel with respect to the left and right channels when three front channels are in use.

Surround Mix Level indicates the normal downmix level of the surround channels if surround channels are in use.

```
SRND MIX LEVEL
   -3.0 dB
```

or

```
   -6.0 dB
```

or

```
    None
```

```
  Not Active
```

Dolby Surround mode indicates whether or not the program has been encoded in Dolby Surround when operating in the two-channel mode.

```
DOLBY SRND MODE
 Not Indicated
```

or

```
 Not Dolby Srnd
```

or

```
   Dolby Srnd
```

```
  Not Active
```

Bit Stream Identification (**N**) represents a numerical version of the Dolby Digital standard as defined by the *ATSC*.

```
 BITSTREAM ID
      N
```

Language Code represents the language of the audio service.

```
LANGUAGE CODE
      N
```

Mixing Level indicates the absolute acoustic sound pressure level of an individual channel during the final audio mixing session. The peak mixing level range is from 80 to 111 dB SPL.

```
 MIXING LEVEL
  NNN dB SPL
```

```
 Not Present
```

```
┌──────────────────────────────┐
│        ROOM TYPE             │
│      Not Indicated           │
└──────────────────────────────┘
```

or

```
┌──────────────────────────────┐
│          Large               │
└──────────────────────────────┘
```

or

```
┌──────────────────────────────┐
│          Small               │
└──────────────────────────────┘
```

```
┌──────────────────────────────┐
│       Not Present            │
└──────────────────────────────┘
```

Room Type indicates the type and calibration of the mixing room used for the final audio mixing session. The DP562 does not use this parameter for decoding; it is simply information for the user.

```
┌──────────────────────────────┐
│       COPYRIGHTED            │
│           YES                │
└──────────────────────────────┘
```

or

```
┌──────────────────────────────┐
│           NO                 │
└──────────────────────────────┘
```

Copyrighted indicates that the content of the stream is copyright protected.

```
┌──────────────────────────────┐
│      ORIGINAL STREAM         │
│           YES                │
└──────────────────────────────┘
```

or

```
┌──────────────────────────────┐
│           NO                 │
└──────────────────────────────┘
```

Original Bit Stream identifies whether the stream is an original or a copy.

```
┌──────────────────────────────┐
│        TIME CODE             │
│      00:00:00:00             │
└──────────────────────────────┘
```

Time Code represents the SMPTE time stamp.

```
┌──────────────────────────────┐
│       CRC ERRORS             │
│        AC3   N               │
└──────────────────────────────┘
```

If a **CRC Error** is found in the AC-3 bitstream. "**N"** represents the number of CRC errors found. "AC-3" logs Dolby Digital CRC Errors.

NOTE *To reset the CRC Error log, press* [ENTER].

4.9.2   Status Screen for PCM Dolby Pro Logic
or Invalid Dolby Digital Bitstream

When information other than Dolby Digital is in the AES/EBU bitstream, one of several parameters shown below may be displayed on the second line of the liquid crystal display.

<div align="center">

AES/EBU STATUS
No AES/EBU Input

or

</div>

Bitstream sample rate.   **48kHz SampRate**

<div align="center">or</div>

Bitstream sample rate.   **44kHz SampRate**

<div align="center">or</div>

Bitstream sample rate.   **32kHz SampRate**

<div align="center">or</div>

When Dolby Digital is manually selected:

Invalid Dolby Digital sample rate.   **No AC-3 Input**

<div align="center">or</div>

Invalid Dolby Digital sample rate.   **Invalid S Rate**

<div align="center">or</div>

Invalid Dolby Digital data rate.   **Invalid D Rate**

<div align="center">or</div>

Bitstream selected in Setup menu is absent.   **Stream N Absent**

<div align="center">or</div>

Invalid Dolby Digital bitstream.   **Invalid Bitstream**

4.10   Runtime Fault Messages

If any of the following faults occurs during initialization or normal system use, the unit will attempt to reboot after approximately 20 seconds or you can also manually reset the DP562. To reset the DP562:

1.   Press the **Setup** menu. If you are not able enter the **Setup** menu, turn off the power by unplugging the unit, and then turn on the unit again.

2.   Scroll through the menu to **User Presets.**

3.   Select "Factory Reset."

If the problem continues to persist, note down the error number and contact Dolby Laboratories for technical support.

▯▯ **MODEL DP562** **System Error 0**

Initialization fault.

or

System Error 1

HC11 fault.

or

System Error 2

Dolby Digital Decoder fault.

or

System Error 3

Dolby Pro Logic Decoder fault.

or

System Error 4

SPI Communications fault.

or

System Error 5

General System fault.

Chapter 5
Setup

The **Setup** menu allows you to configure and modify the DP562 into different modes of operation. The diagram below depicts the **Setup** menu.

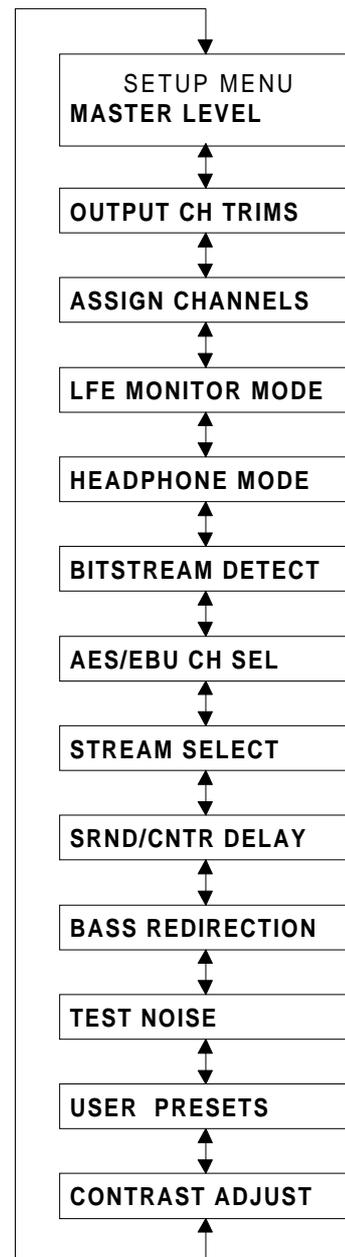### *To operate the Setup menu:*

- Press [**SETUP**] on the front panel. The dedicated LED illuminates red to indicate that the DP562 is in **Setup** mode. Press the [↑] and [↓] buttons on the front panel to scroll through the menu.

- Press [**ENTER**] at any parameter to enter the *Status Level*.

- Press [**SETUP**] at any time to exit the **Setup** menu and return to **Status** mode.

Each menu item has three levels: *Top Level*, *Status Level*, and *Edit Level*.

The *Top Level* represents the **Setup** menu**,** which lists the various audio adjustments available. Press the [↑] and [↓] buttons to **navigate** through this level. Press [**ENTER**] to initiate the *Status Level*.
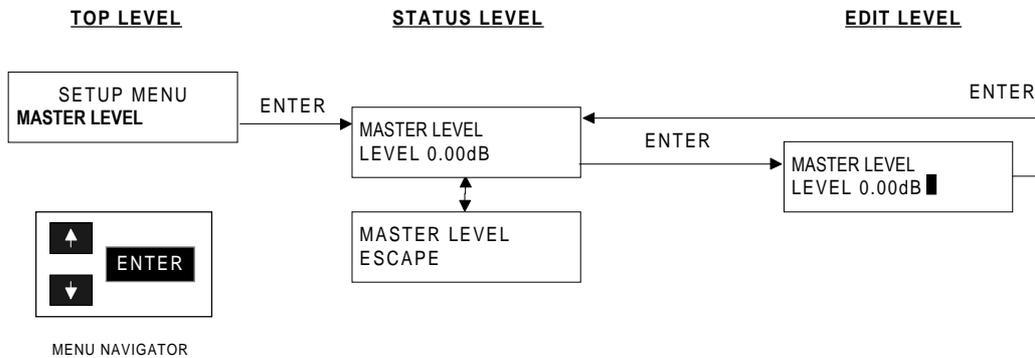
The *Status Level* allows you to **view** a current parameter setting and **select** the parameter you would like to adjust. Press the [↑] and [↓] buttons to **navigate** through this level, too. (Note that the *Status Level* provides an option to return to the previous menu as you loop through the menu at this stage.) Press [**ENTER**] to initiate the *Edit Level*.

The *Edit Level* allows you to **perform adjustments**. A flashing black cursor in the LCD display indicates that you are in the *Edit Level*. Press the [↑] and [↓] buttons to **adjust** a selected parameter at this level. Press [**ENTER**] for the adjustment to take effect which will also return you to the *Status Level*.

```
        ┌──────────────────────┐
        ↓
┌────────────────────────┐
│   SETUP MENU           │
│   MASTER LEVEL         │
└────────────────────────┘
        ↕
┌────────────────────────┐
│   OUTPUT CH TRIMS      │
└────────────────────────┘
        ↕
┌────────────────────────┐
│   ASSIGN CHANNELS      │
└────────────────────────┘
        ↕
┌────────────────────────┐
│   LFE MONITOR MODE     │
└────────────────────────┘
        ↕
┌────────────────────────┐
│   HEADPHONE MODE       │
└────────────────────────┘
        ↕
┌────────────────────────┐
│   BITSTREAM DETECT     │
└────────────────────────┘
        ↕
┌────────────────────────┐
│   AES/EBU CH SEL       │
└────────────────────────┘
        ↕
┌────────────────────────┐
│   STREAM SELECT        │
└────────────────────────┘
        ↕
┌────────────────────────┐
│   SRND/CNTR DELAY      │
└────────────────────────┘
        ↕
┌────────────────────────┐
│   BASS REDIRECTION     │
└────────────────────────┘
        ↕
┌────────────────────────┐
│   TEST NOISE           │
└────────────────────────┘
        ↕
┌────────────────────────┐
│   USER  PRESETS        │
└────────────────────────┘
        ↕
┌────────────────────────┐
│   CONTRAST ADJUST      │
└────────────────────────┘
```

5.1   Master Level Adjust

Master Level allows you to adjust the overall level of the six balanced analog outputs from +10 dB to -95 dB in 0.5 dB increments (0 dB equals +24 dBU full scale).

TOP LEVEL   **STATUS LEVEL**   **EDIT LEVEL**

ENTER

SETUP MENU
MASTER LEVEL   ENTER

MASTER LEVEL
LEVEL 0.00dB   ENTER

MASTER LEVEL
LEVEL 0.00dB█

MASTER LEVEL
ESCAPE

↑
ENTER
↓

MENU NAVIGATOR

To modify Master Level:

NOTE*: If you are not in the **Setup** mode, press [**SETUP**].*

1.   Find **Master Level** by scrolling through the **Setup** menu by pressing the [↑] and [↓] buttons. Select **Master Level** by pressing [**ENTER**]**.**

NOTE*: Use the [↑] and [↓] buttons if you wish to return to the previous menu.*

2.   Press [**ENTER**] again to initiate the *Edit Level*. (The *Edit Level* is defined by a black flashing cursor.)

3.   At the *Edit Level*, use the [↑] and [↓] buttons to adjust the overall output level. The parameter takes effect as soon as the [↑] and [↓] buttons are pressed.

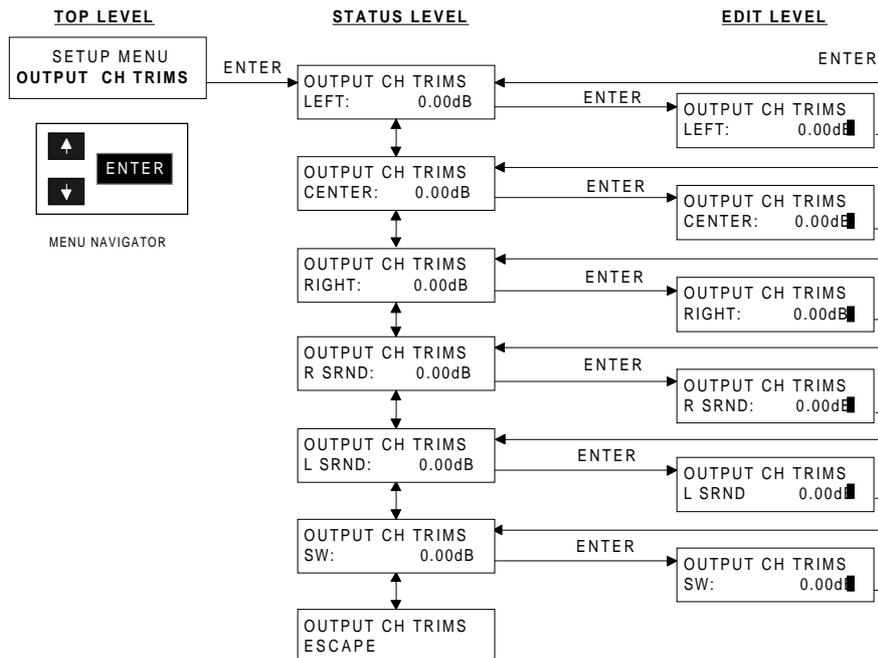**NOTE***: Holding down the [↑] or [↓] button for three seconds in the Edit Level enables speed scrolling.*

4.   After adjusting the output, press [**ENTER**] to return to the *Status Level*, at which time the cursor will stop flashing.

To exit Master Level:

•   Scroll to the "ESCAPE" menu selection and press [**ENTER**] to return to the main **Setup** menu.

•   Press [**SETUP**] to return to the **Status** mode.

5.2   Output Ch Trims

Output Channel Trims allows you to adjust individual audio channels in the analog outputs (only) to compensate for different amplifier gains, speaker efficiencies, and room acoustics. The trim parameter tracks the audio channel specified in the **Assign Channels** screen of the **Setup** menu. Each channel can be adjusted to a range from +6 dB to -18 dB in 0.125 dB steps (0 dB equals +24 dBU full scale). If you enable **Test Noise** while in the **Output Channel Trims** menu, test noise will remain stationary on the audio channel of interest (See *Section 3*, *Calibration*).

TOP LEVEL

```
SETUP MENU
OUTPUT  CH TRIMS
```

ENTER →

STATUS LEVEL

```
OUTPUT CH TRIMS
LEFT:        0.00dB
```

```
↑
ENTER
↓
```

MENU NAVIGATOR

EDIT LEVEL

ENTER

```
OUTPUT CH TRIMS
LEFT:        0.00dB
```

ENTER

```
OUTPUT CH TRIMS
CENTER:    0.00dB
```

ENTER →
```
OUTPUT CH TRIMS
CENTER:    0.00dB
```

```
OUTPUT CH TRIMS
RIGHT:      0.00dB
```

ENTER →
```
OUTPUT CH TRIMS
RIGHT:      0.00dB
```

```
OUTPUT CH TRIMS
R SRND:      0.00dB
```

ENTER →
```
OUTPUT CH TRIMS
R SRND:      0.00dB
```

```
OUTPUT CH TRIMS
L SRND:      0.00dB
```

ENTER →
```
OUTPUT CH TRIMS
L SRND      0.00dB
```

```
OUTPUT CH TRIMS
SW:          0.00dB
```

ENTER →
```
OUTPUT CH TRIMS
SW:          0.00dB
```

```
OUTPUT CH TRIMS
ESCAPE
```

To adjust Output Channel Trims:

NOTE*: If you are not in the **Setup** mode, press* [**SETUP**].

1.   Scroll through the **Setup** menu**,** by pressing the [↑] and [↓] buttons, to find **Output Ch Trims**.

2.   Select **Output Ch Trims** by pressing [**ENTER**]**.** This enters you into the *Status Level*.

3.   Use the [↑] and [↓] buttons in the **Output Ch Trims** menu to find the audio channel to be adjusted or to loop back to the previous menu.

4.   Press [**ENTER**] after you have selected the channel. This initiates the *Edit Level*. (The *Edit Level* is defined by a black solid flashing cursor.)

5.   At the *Edit Level*, use the [↑] and [↓] buttons to adjust the output trim of that particular channel. The parameter takes effect as soon as the [↑] and [↓] buttons is pressed.

NOTE*: Holding down the* [↑] *or* [↓] *buttons for three seconds in the Edit Level enables speed scrolling.*

6.   Press [**ENTER**] to return to the *Status Level*.

To exit Output Ch Trims:

- Scroll to the "ESCAPE" menu selection and press [**ENTER**] to return to the main **Setup** menu.

- Press [**SETUP**] to return to the **Status** mode.
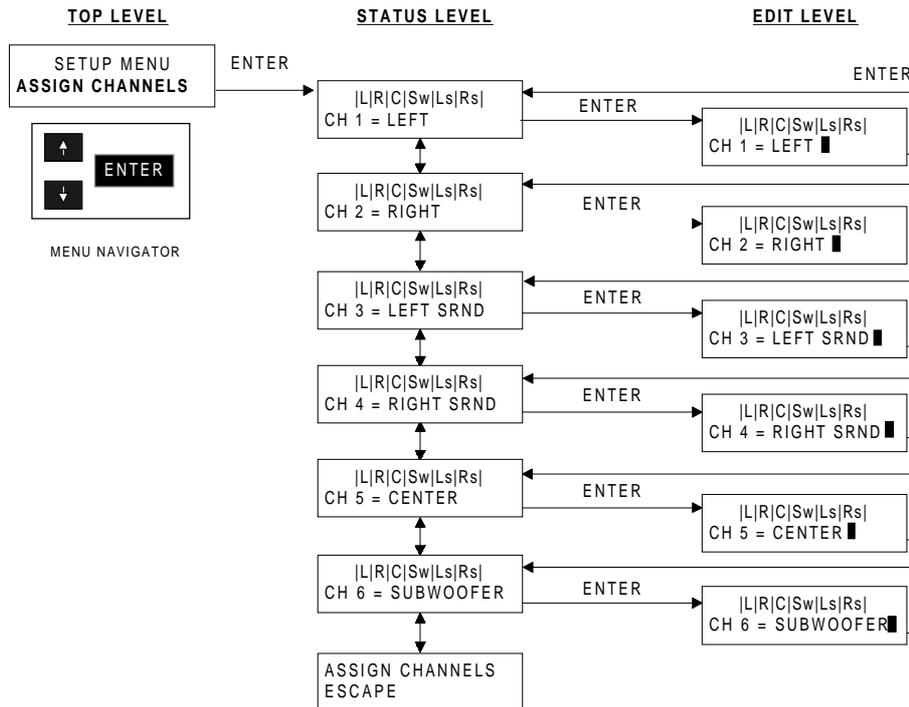
5.3   Assign Channels

The **Assign Channels** feature allows you to assign one audio channel to any one of the output connectors (*Ch 1–6* in both analog and digital output connectors). If you are facing the rear panel, the left-most position of the row of **analog** output connectors represents *Ch 1*. The left-most position of the row of **digital** output connectors represents the first two channels, *Ch 1* and *2* (*Figure 5-1*.)

Analog Outputs Ch 1 - 6   **Digital Outputs Ch 1/2, 3/4, 5/6**
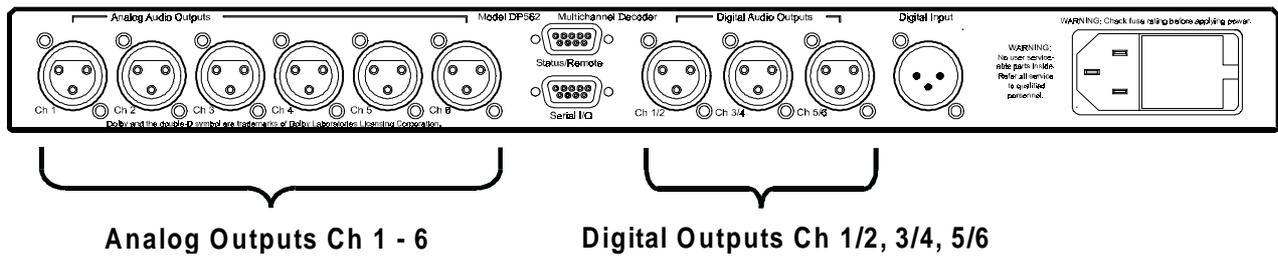
Figure 5 – 1. Rear panel.

When you reassign a channel, that channel exchanges its channel *number* with the desired location. For example, using the following diagram, if you selected "CH 1 = LEFT" and reassigned it to the location of "CH 4 = RIGHT SRND," the **Status Level** screens will read "CH 1 = RIGHT SRND" and "CH 4 = LEFT."

The default positions for **Assign Channels** are depicted below:

TOP LEVEL   **STATUS LEVEL**   **EDIT LEVEL**

To assign audio channels to an output connector:

NOTE*: If you are not in the **Setup** mode, press the* [**SETUP**] *button*.

1. Scroll through the **Setup** menu, using the [↑] and [↓] buttons, to find **Assign Channels**.

2. Select **Assign Channels** by pressing [**ENTER**]**.** You have now entered the *Status Level*.

3. Use the [↑] and [↓] buttons to scroll through the **Assign Channels** menu and select a physical output connector to be changed. You can also loop back to the previous menu by continuing to scroll through the menu.

4. After you have selected the output connector you want to adjust, press [**ENTER**] to get to the *Edit Level*. (The *Edit Level* is defined by a black flashing cursor.)

5. At the *Edit Level*, use the [↑] and [↓] buttons to assign the audio channel to a new output connector.

6. Press [**ENTER**] to allow the adjustments to take effect and to return to the *Status Level*.

To exit Assign Channels:

- Scroll to the "ESCAPE" menu selection and press [**ENTER**] to return to the main **Setup** menu.

- Press the [**SETUP**] button to return to the **Status** mode.

5.4   LFE Monitor Mode

Low Frequency Effects Monitor Mode allows you to select whether you want the incoming LFE channel to pass-through to the subwoofer output:

• Select "ON" to allow the incoming LFE channel to pass through to the subwoofer output.

• Select "OFF" to disallow the incoming LFE channel to pass-through.

• Select "AUTO SELECT" to allow the decoder to automatically determine the LFE function through both the audio-coding mode set in the original bitstream and the selected decoding mode.

In "AUTO SELECT," LFE monitoring is enabled only during the Dolby Digital mode, and when *Full*, *3-Stereo,* or *Phantom* listening mode is selected. LFE monitoring mode is disabled in all other decoding and listening modes. This mode is designed to best represent how a consumer decoder will work.
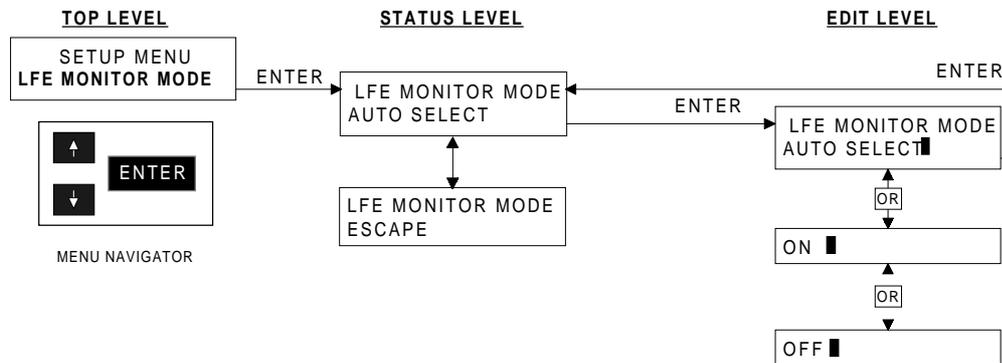
To adjust the LFE Monitor Mode:

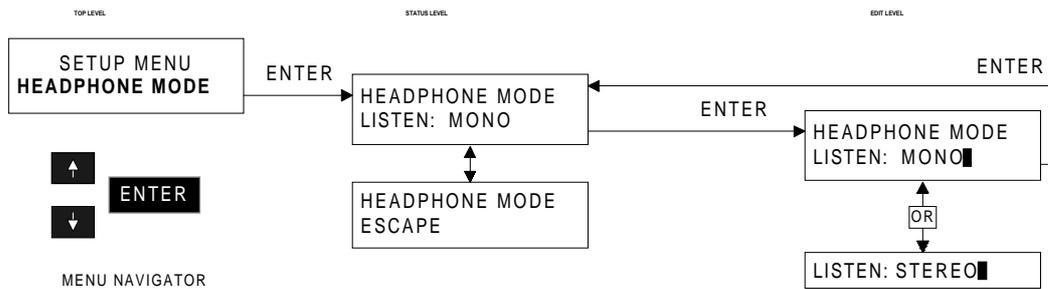**NOTE***: If you are not in the **Setup** mode, press* [**SETUP**].

1.   Press the [↑] and [↓] buttons to find **LFE Monitor Mode** by scrolling through the **Setup** menu.

2.   Select **LFE Monitor Mode** by pressing [**ENTER**]**.** This initiates the *Status Level*.

3.   Press [**ENTER**] again to get into to the *Edit Level*. (The *Edit Level* is defined by a black flashing cursor.) "AUTO SELECT" is the default.

4.   Once in the *Edit Level*, use the [↑] and [↓] buttons to choose "AUTO SELECT," "ON," or "OFF."

5.   The parameter takes effect as soon as [**ENTER**] is pressed.

To exit LFE Monitor Mode:

• Scroll to the "ESCAPE" menu selection and press [**ENTER**] to return to the main **Setup** menu.

• Press [**SETUP**] to return to the **Status** mode.

5.5   Headphone Mode

Headphone Mode adjusts the headphone listening mode for either Stereo or Mono. Due to some hardware limitations, correct imaging in *Stereo mode* can only be achieved if the output channel assignments in the "ASSIGN CHANNEL" section of the setup menu are set to L/R/Ls/Rs/C/Sw. The default mode is "Mono."

To set the Headphone Mode:

NOTE*: If you are not in the **Setup** mode, press* [**SETUP**].

1. Find **Headphone Mode** by using the [↑] and [↓] buttons to scroll through the **Setup** menu.

2. Select **Headphone Mode** by pressing [**ENTER**]**.** You have entered the *Status Level*.

3. Select **Listen: Stereo** (default) by pressing [**ENTER**]. This initiates the *Edit Level*. (The *Edit Level* is defined by a black flashing cursor.)

4. Once in the *Edit Level*, use the [↑] and [↓] buttons to adjust the **Headphone Mode** to "STEREO" or "MONO."

5. Press [**ENTER**] after you have selected the mode you want. The mode takes effect and returns you to the *Status Level*, as soon as [**ENTER**] is pressed. Use the [↑] and [↓] buttons to scroll to other status information or to return to the previous menu level.

To exit Headphone Mode:

• Scroll to the "ESCAPE" menu selection and press [**ENTER**] to return to the main **Setup** menu.

• Press [**SETUP**] to return to the **Status** mode.

5.6   Bitstream Detect

Bitstream Detect enables or disables automatic Dolby Digital (AC-3) bitstream detection. When set to "REFERENCE" or "SILENT SWITCH," if a valid IEC 1937 (formerly IEC 958 Annex B) input stream is present, the decoder automatically detects and decodes a Dolby Digital bitstream. If the IEC 1937 stream is not present, the decoder assumes the input bitstream to be PCM. The Dolby Digital LED will flash at a periodic rate to indicate that the DP562 is in one of the automatic bitstream detection modes and that no valid Dolby Digital stream is present. If [**DOLBY PRO LOGIC**] is selected while in automatic bitstream detection, and the input is a valid Lt/Rt AES/EBU digital audio, then Pro Logic is decoded.

Bitstream Detect has three modes:

- Select "DISABLED" to manually select either Dolby Digital or PCM decoding modes.

- Select "REFERENCE" to enable automatic bitstream detection without adding any additional decoding delay. When using this mode, audio clicks may occur when switching between Dolby Digital and PCM.

- Select **"SILENT SWITCH"** to enable automatic bitstream detection but minimize audio artifacts while switching between Dolby Digital and PCM. When Bitstream detect is set to **"SILENT SWITCH,"** extra buffering is implemented in both Dolby Digital and PCM. This buffering is necessary to allow more seamless transitions from PCM to Dolby Digital (or Dolby Digital and PCM).
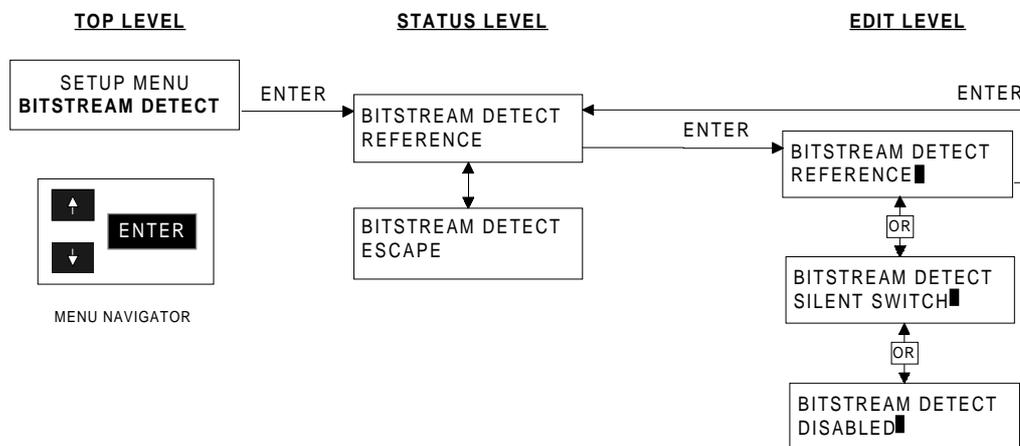
To enable or disable Bitstream Detect:

NOTE: *If you are not in the **Setup** mode, press* [**SETUP**].

1. Find **Bitstream Detect** by using the [↑] and [↓] buttons to scroll through the **Setup** menu.

2. Press [**ENTER**] to select **Bitstream Detect**. Use the [↑] and [↓] buttons to view the current parameter or to return to the previous menu level.

3. Press [**ENTER**] again to toggle between "DISABLED," "REFERENCE," and "SILENT SWITCH," and return to the *Status Level*.

> *WARNING: When **Bitstream Detect** is set to "DISABLED" and the Dolby Digital mode is not selected, the unit will output un-decoded audio to the outputs at full scale if a Dolby Digital bitstream is present. At high playback levels, you may run the risk of damaging your speakers or possibly your hearing.*
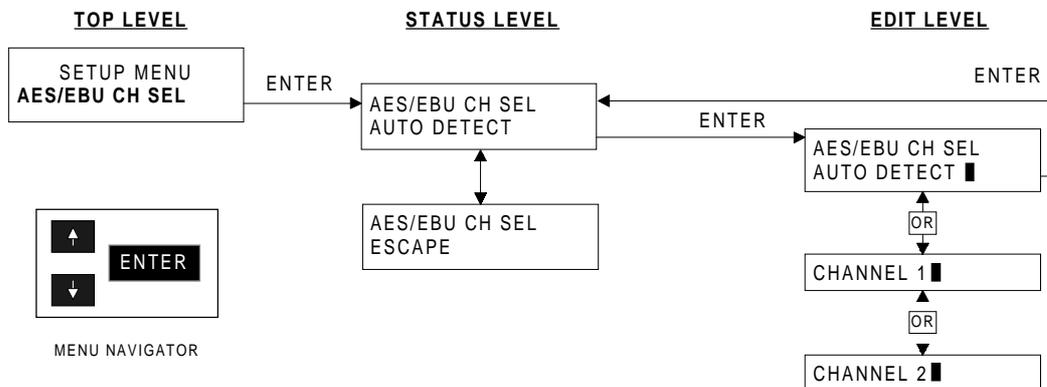
To exit Bitstream Detect:

*   Scroll to the "ESCAPE" menu selection and press [**ENTER**] to return to the main **Setup** menu.

*   Press [**SETUP**] to return to the **Status** mode.

5.7   AES/EBU Ch Select

AES/EBU Channel Select allows the DP562 to select the AES/EBU input bitstream channel. The AES/EBU input bitstream can be formatted in either 32-bit or 16-bit mode. If the input bitstream is formatted in 16-bit mode, two channels (PCM/Dolby Digital) can be carried within a single bitstream. For example, the input bitstream can have a Dolby Digital bitstream on one 16-bit channel and PCM on the other; Dolby Digital can be in both channels; or PCM can be in both channels.

AES/EBU Ch Select allows you to select one of three options:

- Select "CHANNEL 1" or "CHANNEL 2" to decode a bitstream formatted in 16-bit mode.

- Select "AUTO DETECT" (factory default) to decode a bitstream formatted in 32-bit mode. If the input bitstream is formatted in 16-bit mode when "AUTO DETECT" is selected, decoding of Channel 1 will take precedence over decoding of Channel 2.

To adjust AES/EBU Channel Select:

NOTE*: If you are not in the **Setup** mode, press* [**SETUP**].

1. Scroll through the Setup menu, using the [↑] and [↓] buttons, to find **AES/EBU Ch Sel**.

2. Select **AES/EBU Ch Sel** by pressing [**ENTER**].

3. Press [**ENTER**] again to enable the *Edit Level*. (The Edit Level is defined by a black flashing cursor.)

4. Once in the *Edit Level*, use the [↑] and [↓] buttons to choose "AUTO DETECT," "CHANNEL 1," or "CHANNEL 2." Press [**ENTER**] to save your selection. The parameter takes effect and returns you to the *Status Level*.

To exit AES/EBU Ch Sel:

- Scroll to the "ESCAPE" menu selection and press [**ENTER**] to return to the main **Setup** menu.

- Press [**SETUP**] to return to the **Status** mode.

5.8   Stream Select

Stream Select chooses one of eight streams (0–7) within the AES/EBU bitstream.[*]
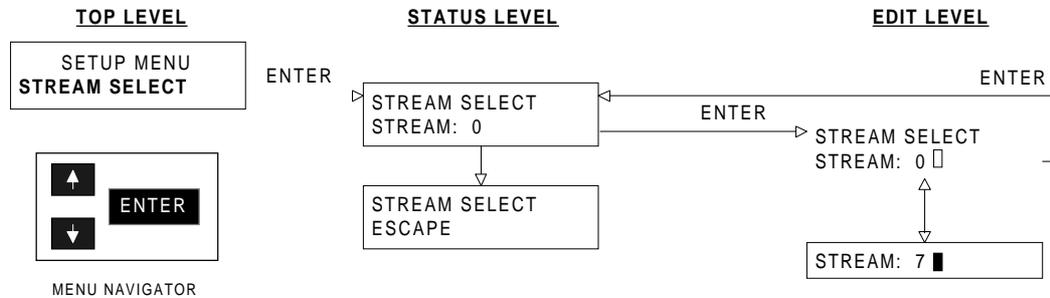
To adjust Stream Select:

NOTE*: If you are not in the **Setup** mode, press* [**SETUP**]*.*

1.   In the **Setup** menu, use the [↑] and [↓] buttons to find **Stream Select**.

2.   Select **Stream Select** by pressing [**ENTER**].

3.   At the *Status Level*, press [**ENTER**] to initiate the *Edit Level*.

4.   At the *Edit Level*, use the [↑] and [↓] buttons to choose a stream (0–7) within the AES/EBU bitstream. Press [**ENTER**] to save your selection. The parameter takes effect and returns you to the *Status Level*.

To exit Stream Select:

•   Scroll to the "ESCAPE" menu selection and press [**ENTER**] to return to the main **Setup** menu.

•   Press [**SETUP**] to return to the **Status** mode.

---

[*] The stream number is specified in the *burst_info* variable of the IEC 1937 Preamble. Please refer to *Annex B of the ATSC Standard*.

5.9   SRND/CNTR Delay

SRND/CNTR Delay allows the adjustment of Center Channel Delay (from -3 to +5 ms), Dolby Digital Surround (from 0 to 50 ms), and Pro Logic Surround (from 10 to 75 ms) For information on calibrating the DP562, see *Section 3*, *Calibration*.
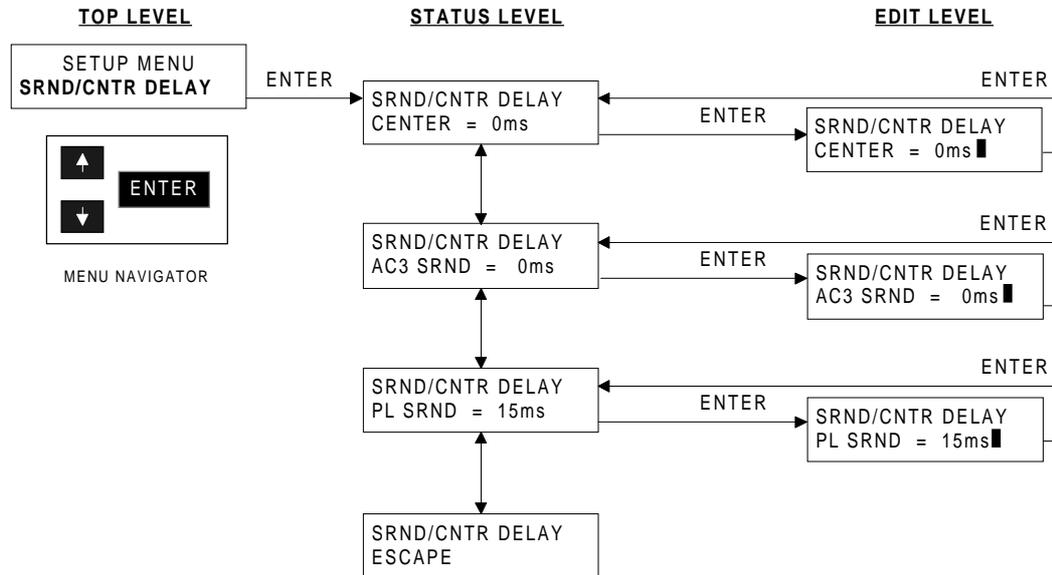
To adjust the surround or center delay:

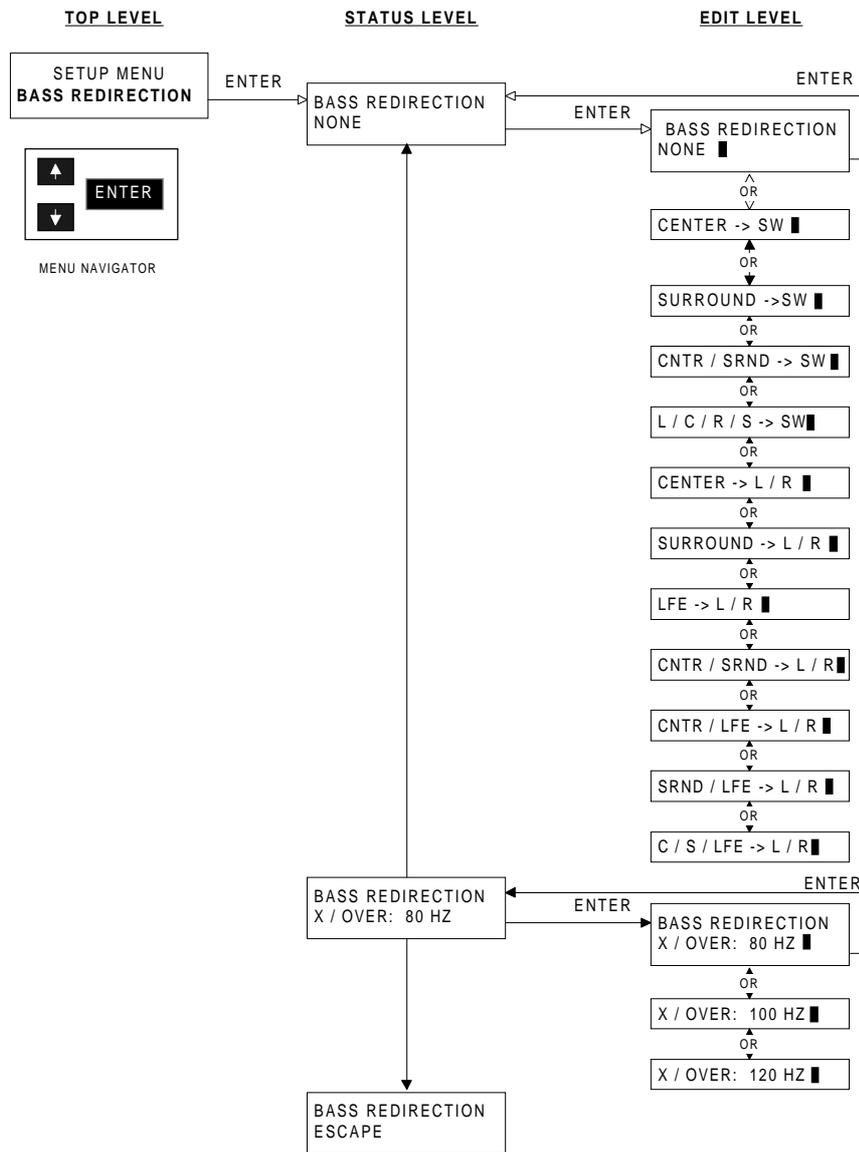**NOTE***: If you are not in the **Setup** mode, press* [**SETUP**].

1.   Using the [↑] and [↓] buttons, scroll through the **Setup** menu to find **SRND/CNTR Delay**.

2.   Press [**ENTER**] to select **SRND/CNTR Delay**.

3.   At the *Status Level*, use the [↑] and [↓] buttons to choose which delay you would like to modify (or you may return to the previous menu level at this time).

4.   Press [**ENTER**] after making your selection.

5.   In the *Edit Level*, use the [↑] and [↓] buttons to adjust the delay. Press [**ENTER**] to save your selection. The parameter takes effect and returns you to the *Status Level*.

To exit SRND/CNTR Delay:

• Scroll to the "ESCAPE" menu selection and press [**ENTER**] to return to the main **Setup** menu.

• Press [**SETUP**] to return to the **Status** mode.

5.10 Bass Redirection

Bass Redirection redirects bass from one set of audio channel(s) to another.[*] The number and frequency specifications of loudspeakers present in the listening environment usually determine the selection.

TOP LEVEL   **STATUS LEVEL**   **EDIT LEVEL**

SETUP MENU
BASS REDIRECTION → ENTER → BASS REDIRECTION
NONE   → ENTER → BASS REDIRECTION NONE ▮ ← ENTER

MENU NAVIGATOR
[↑] ENTER [↓]

OR

CENTER -> SW ▮

OR

SURROUND ->SW ▮

OR

CNTR / SRND -> SW ▮

OR

L / C / R / S -> SW▮

OR

CENTER -> L / R ▮

OR

SURROUND -> L / R ▮

OR

LFE -> L / R ▮

OR

CNTR / SRND -> L / R▮

OR

CNTR / LFE -> L / R ▮

OR

SRND / LFE -> L / R ▮

OR

C / S / LFE -> L / R▮

BASS REDIRECTION
X / OVER: 80 HZ ← ENTER → BASS REDIRECTION X / OVER: 80 HZ ▮ ← ENTER

OR

X / OVER: 100 HZ ▮

OR

X / OVER: 120 HZ ▮

BASS REDIRECTION
ESCAPE

(To adjust Bass Redirection, please see the following page.)

NOTE*: Using Bass Redirection adds an additional output delay of approximately 15 ms.*

[*] The Bass Redirection algorithm implements digital fourth-order Linkwitz-Riley filters. (Additional filters are used for decimation and interpolation of sample rates.)

To adjust Bass Redirection:

NOTE*: To properly adjust Bass Redirection in the **Setup Menu**, first modify the "Bass Redirection" option (i.e., "CENTER -> L / R" or "SURROUND -> L / R," etc.), then select the **Bass Redirection Crossover** (X / OVER) frequency.*

---

1. Scroll through the **Setup** menu**,** using the [↑] and [↓] buttons, to find **Bass Redirection**.

2. Select **Bass Redirection** by pressing [**ENTER**]**.**

3. Press [**ENTER**] again to enable the *Status Level* which displays the current **Bass Redirection** option (default is **None**). You may use the [↑] and [↓] buttons to choose the **Bass Redirection X / OVER** or to return to the previous menu level at this time.

4. Press [**ENTER**] again, to initiate the *Edit Level*. (The *Edit Level* is defined by a black flashing cursor.)

5. Use the [↑] and [↓] buttons to select the set of channels for bass redirection. The adjustment takes effect and the DP562 returns to the *Status Level* as soon as [**ENTER**] is pressed.

6. Next, you must select the Bass Redirection Crossover. At the Status Level, scroll down to **Bass Redirection X / OVER**.

7. Press [**ENTER**] to initiate the *Edit Level*. (The *Edit Level* is defined by a black flashing cursor.)

8. Use the [↑] and [↓] buttons to select the level of crossover you desire: "80 Hz," "100 Hz," or "120 Hz." Press [**ENTER**] to save your selection. The parameter takes effect and returns you to the *Status Level*.

To exit Bass Redirection:

- Scroll to the "ESCAPE" menu selection and press [**ENTER**] to return to the main **Setup** menu.

- Press [**SETUP**] to return to the **Status** mode.

5.11   Test Noise

Test Noise enables the selection of noise type generated by a noise sequencer built into the DP562. When **Test Noise** is "FILTERED," bandpass-filtered pink noise from 400 Hz to 1500 Hz is generated to allow you to balance channels by ear from all speakers. When **Test Noise** is "PINK," full-band pink noise is generated. The subwoofer outputs are band-limited to 120 Hz.
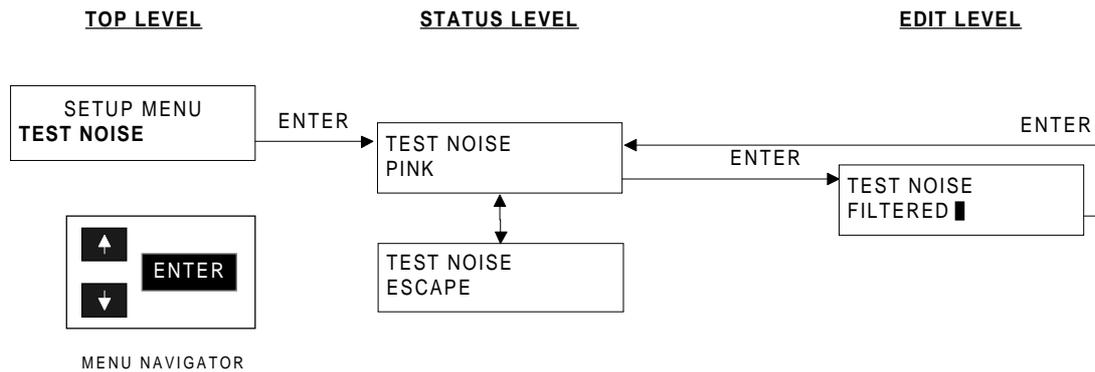
To enable or disable "FILTERED" Noise:

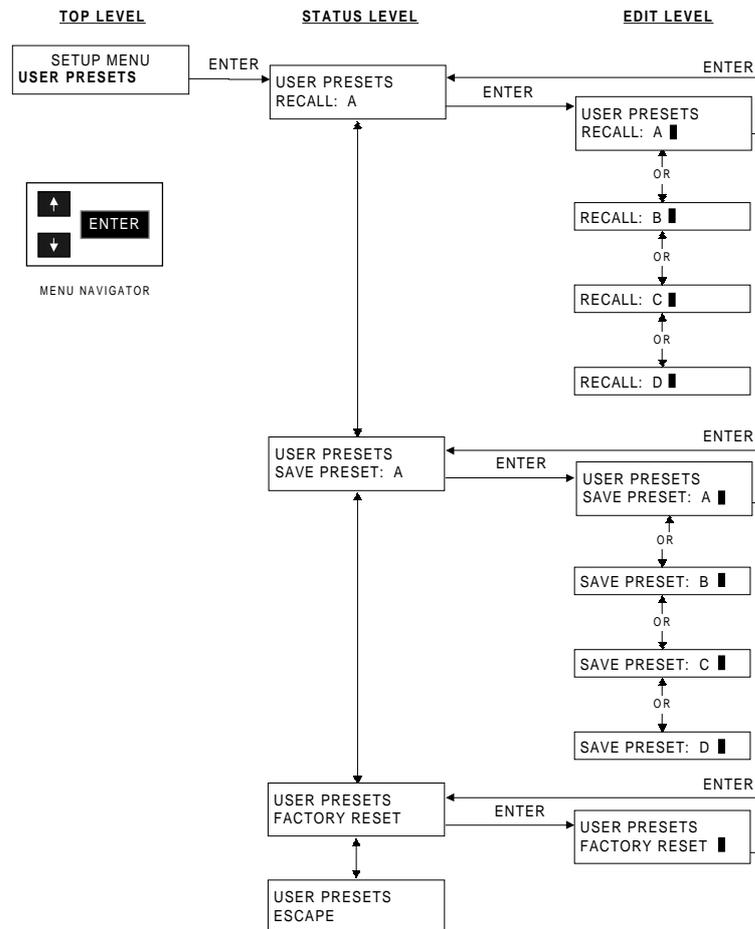**NOTE***: If you are not in the **Setup** mode, press the* [**SETUP**] *button.*

1.   Using the [↑] and [↓] buttons**,** find **Test Noise** by scrolling through the **Setup** menu.

2.   Select **Test Noise** by pressing [**ENTER**]**.**

3.   Press [**ENTER**] again to toggle between "PINK" and "FILTERED." (You may also return to the previous menu by using the [↑] and [↓] buttons.) Press [**ENTER**] to save your selection. The parameter takes effect and returns you to the *Status Level*.

To exit Test Noise:

•   Scroll to the "ESCAPE" menu selection and press [**ENTER**] to return to the main **Setup** menu.

•   Press [**SETUP**] to return to the **Status** mode.

5.12   User Presets

User Presets saves preset menu parameters for future recall. For example, you can save setup parameters for different room types or A/B listening. When a preset is recalled, the hardware is immediately configured based on what was saved in that location. Selecting "FACTORY RESET" adjusts the preset parameters back to the factory defaults.

To set up User Presets:

NOTE*: If you are not in the **Setup** mode, press* [**SETUP**].

1.   Find **User Presets** by scrolling through the **Setup** menu with the [↑] and [↓] buttons.

2.   Press [**ENTER**] to select **User Presets**.

3.   At the *Status Level* use the [↑] and [↓] buttons if you wish to "RECALL," "SAVE," or "FACTORY RESET" the current DP562 Setup. You may also return to the previous menu level at this time.

4.   Press [**ENTER**] after you have chosen the setup you would like to recall or adjust.

5.   At the *Edit Level*, use the [↑] and [↓] buttons to modify the parameter. Press [**ENTER**] to save your selection. The parameter takes effect and returns you to the *Status Level*.

To exit User Presets:

- Scroll to the "ESCAPE" menu selection and press [**ENTER**] to return to the main **Setup** menu.

- Press [**SETUP**] to return to the **Status** mode.

5.13   Contrast Adjust

Contrast Adjust changes the viewing quality of the liquid crystal display.
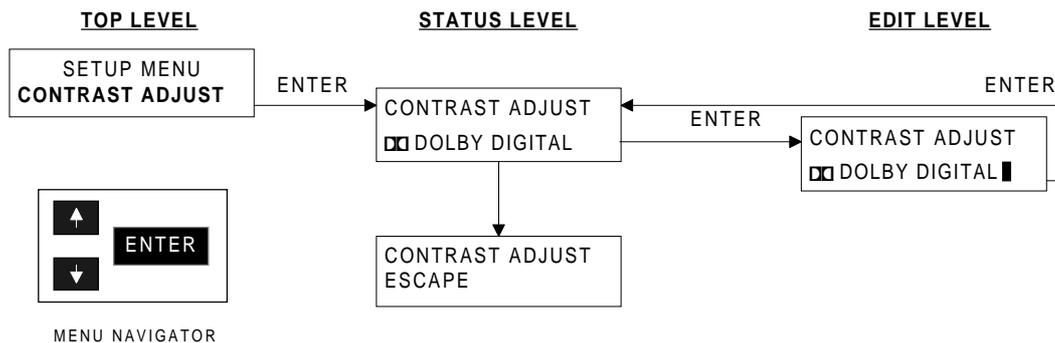
To adjust contrast:

NOTE*: If you are not in the **Setup** mode, press [**SETUP**].*

1. Scroll through the **Setup** menu using the [↑] and [↓] buttons to find **Contrast Adjust**.

2. Select **Contrast Adjust** by pressing [**ENTER**]**.** This enters you into the *Status Level*. Use the [↑] and [↓] buttons to scroll to other status information or to return to the previous menu level.

3. Press [**ENTER**] again to initiate the *Edit Level*. (The *Edit Level* is defined by a black flashing cursor.)

4. At the *Edit Level*, use the [↑] and [↓] buttons to adjust the contrast. Press [**ENTER**] to save your selection. The parameter takes effect and returns you to the *Status Level*.

To exit Contrast Adjust:

- Scroll to the "ESCAPE" menu selection and press [**ENTER**] to return to the main **Setup** menu.

- Press [**SETUP**] to return to the **Status** mode.

Chapter 6
Future Software Upgrades

The DP562 can be upgraded to the latest Dolby Digital (AC-3) decoding algorithm and user-interface features via the rear-panel serial I/O port connected to a personal computer.

6.1   Equipment Needed

- Personal computer with an RS-232 port, capable of communication at 9600 b/s.
- A 9-pin D-connector and cable to connect the computer and DP562.
- An upgrade software disk provided by Dolby Laboratories when a new release becomes available.
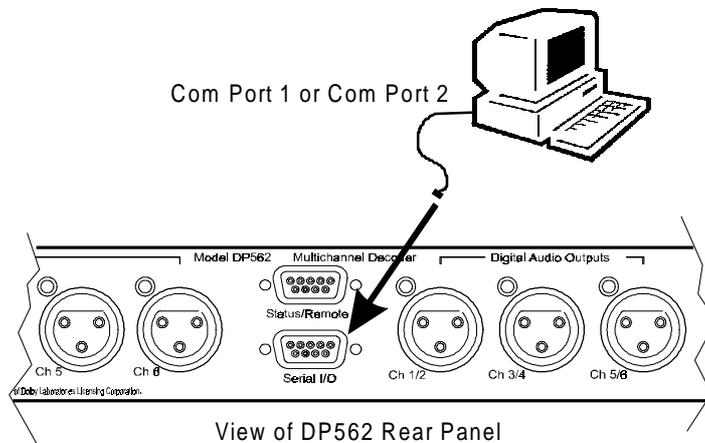
6.2   Software Upgrade

1.   Connect the computer to the DP562 serial I/O port.

> ⚠ **CAUTION**: *Do not connect the computer to the Status/Remote port.*

Com Port 1 or Com Port 2

View of DP562 Rear Panel

2.   Press [**SETUP**] to enter the **Setup** menu.

3.   Scroll through the **Setup** menu and select **User Presets.**

4.   Under **User Presets** choose "FACTORY RESET." While the unit resets, press down and hold the [**ENTER**] button until the following message appears:

> ⊓⊔ **MODEL DP562**
> **READY TO LOAD**

5.   Follow the instructions provided with the software upgrade that will appear on the computer. When the computer indicates that the software installation is successfully completed, the computer will automatically reboot the DP562. To verify that the proper software is loaded, compare the revision level indicated on the software distribution disk to the display during a power-on reset.

6.   Next, a series of messages appear on the DP562 display. The messages depend upon the software module being loaded and may appear in any order.

```
┌────────────────────────────┐
│   ▌▌ MODEL DP562           │
│   LOADING HC11             │
└────────────────────────────┘
```

or

```
┌────────────────────────────┐
│ LOADING BOOT               │
└────────────────────────────┘
```

or

```
┌────────────────────────────┐
│ LOADING DSP1               │
└────────────────────────────┘
```

or

```
┌────────────────────────────┐
│ LOADING DSP2               │
└────────────────────────────┘
```

6.3   Software Upgrade Fault Messages

If one of the following faults occur while performing a serial upgrade, reset the DP562 and repeat the software upgrade process. Reset the DP562 by following these steps:

1.   Press the **Setup** menu.

2.   Scroll through the menu to **User Presets**.

3.   Select "FACTORY RESET." If you are not able to enter the **Setup** menu, unplug then re-plug mains power to the DP562. This resets the DP562.

If the problem continues, record the error number displayed and contact Dolby Laboratories technical support.

Checksum Error Level 1

┌─────────────────────────┐
│   ▯▮ MODEL DP562   │
│   DOWNLOAD ERROR 0   │
└─────────────────────────┘

or

Program Error Level 1

┌─────────────────────────┐
│   DOWNLOAD ERROR 1   │
└─────────────────────────┘

or

Checksum Error Level 2

┌─────────────────────────┐
│   DOWNLOAD ERROR 2   │
└─────────────────────────┘

or

Program Error Level 2

┌─────────────────────────┐
│   DOWNLOAD ERROR 3   │
└─────────────────────────┘

or

Low Level SCI Fault

┌─────────────────────────┐
│   DOWNLOAD ERROR 4   │
└─────────────────────────┘

Appendix A
About Dolby Pro Logic and Dolby Digital

A.1 Dolby Pro Logic

Multichannel sound was first heard widely with the widescreen movie formats developed in the early 1950s. They featured four to six channels recorded on stripes of magnetic oxide applied to each release print, a costly and time-consuming process.

In the mid-1970s, Dolby Laboratories introduced a new, more practical multichannel sound format for 35-mm prints. Instead of magnetic striping, it was based on the photographic, or optical, soundtrack that had been used to put mono sound on film since the 1930s.

To enable compatible playback in mono theaters, Dolby developed a new multichannel soundtrack to fit in the same space on release prints as a conventional mono track. This was made possible by using two optical tracks matrix-encoded with four channels of information: left, center, and right screen channels, plus a surround channel. The four channels were reconstructed in the cinema by means of a matrix decoder combined with sophisticated steering techniques.

Because the original four-channel encoding remained intact when encoded films were transferred to stereo home video media, it was a logical next step to provide multichannel playback in the home. At first, simple Dolby Surround decoders, as they were called, made it possible to reproduce the surround channel along with the left and right channels. Because of the limitations of this approach, including the lack of a center channel to locate dialogue firmly on the screen, more sophisticated Dolby Pro Logic decoding was soon introduced to the consumer market. It used the advanced steering circuitry developed originally for the cinema to provide full four-channel playback (*Figure A-1*).
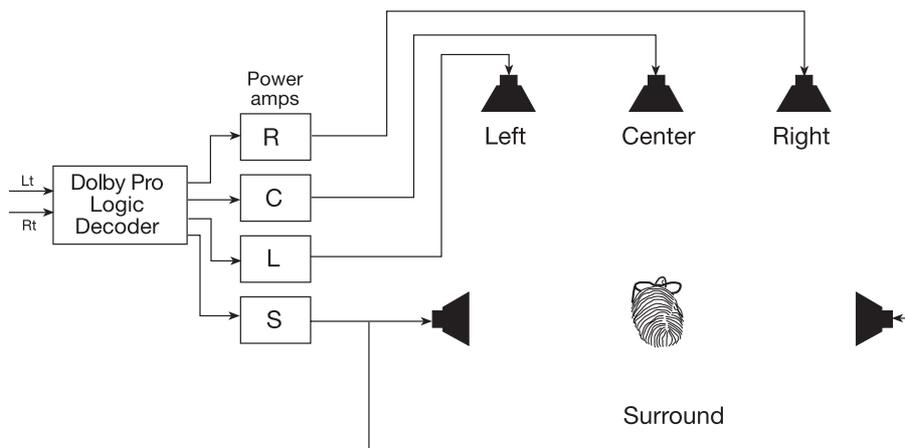
Figure A-1 Dolby Pro Logic playback configuration.

The presence of Dolby Pro Logic decoding in millions of home theater systems has fostered a high demand for Dolby Surround encoded programming, from TV shows to video games. Any conventional stereo or mono program can carry surround-encoded Left total (Lt) and Right total (Rt) tracks, which can be played back in conventional stereo without loss of information, or through a Dolby Pro Logic decoder to reconstruct the original left, center, right, and surround channels.

The DP562's Pro Logic decoder (*Figure A-2*) incorporates an adjustable surround-channel delay (15–75 ms, with a 20 ms default) that takes advantage of the Haas effect.[*] This reduces the audibility of front-channel information that might leak into the surround speakers. Limiting surround-channel bandwidth to 7 kHz minimizes leakage of high-frequency surround information into the front channels due to azimuth error.
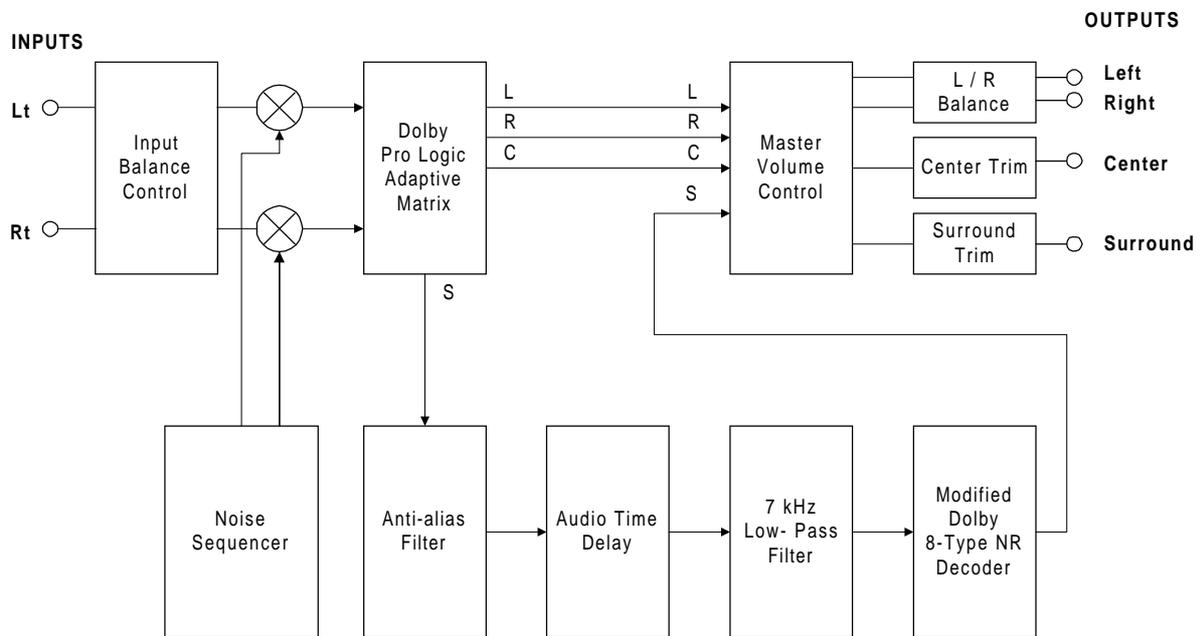
**Figure A-2** Dolby Pro Logic decoder block diagram.

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[*] The Haas Effect states that if two similar sounds arrive at our ears at slightly different times, the brain tends to focus on the first arriving sound and ignore the second.

A.2   Dolby Digital

In 1992, Dolby Laboratories introduced a new film sound format featuring a *digital* optical soundtrack placed between the sprocket holes on 35-mm release prints. The standard matrix-encoded analog optical soundtrack was left intact so that the prints could play in any cinema.

In response to film industry requirements, the new digital soundtrack provided six discrete channels: three full-range channels for left, center, and right screen speakers; two full-range surround channels; and a bass-only low-frequency effects channel (*Figure A-3*). (Now generally called the "5.1-channel" format, this configuration has since become the standard for digital surround sound in the home as well as in the cinema.)

To fit all this information in the space allotted to the digital track on movie prints, Dolby Laboratories developed a new form of multichannel perceptual digital audio coding. Being the third coding system developed by Dolby, it was originally named Dolby AC–3, but is now widely known simply as Dolby Digital.
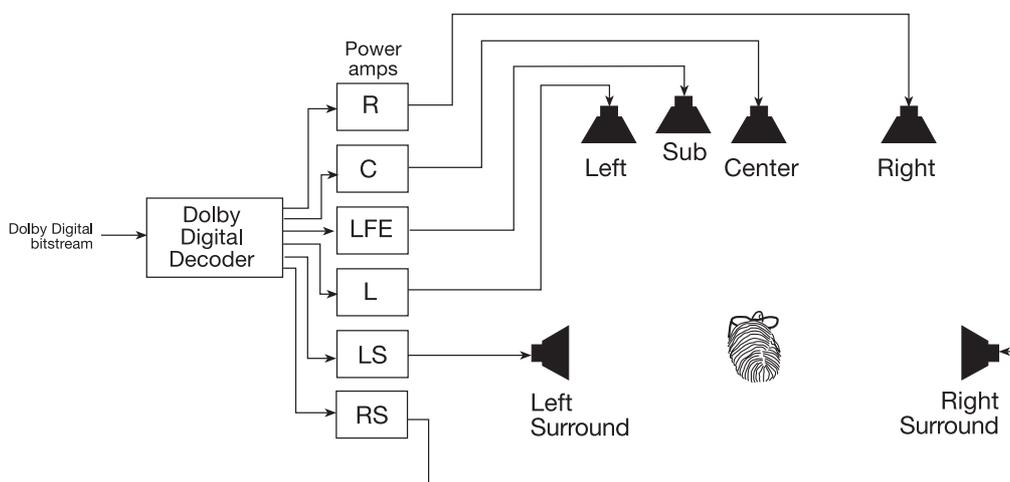
Figure A-3 Dolby Digital 5.1-channel playback configuration.

As happened with Dolby's analog surround technologies, film sound provided a springboard for the application of Dolby Digital to other formats. It was launched in two-channel DBS (Digital Broadcast Satellite) applications in 1994 and reached consumers through 5.1-channel laser discs in 1995. Dolby Digital is the standard multichannel audio format for the ATSC (Advanced Television System Committee) digital television standard and is one of the standard multichannel audio formats for DVB (Digital Video Broadcasting). Dolby Digital is also the standard multichannel audio format for DVDs (Digital Versatile Discs) in NTSC (National Television Standards Committee) countries as well as one of the standard multichannel audio formats for PAL and SECAM DVDs. Other uses include digital cable systems, DAB, and multimedia.

Dolby Digital is unusually versatile, allowing such parameters as bit-rate and number of channels to be tailored to particular applications. It also provides features to ensure listener satisfaction under a wide range of circumstances:

- Data identifying each program's original production format—mono, stereo, matrixed or discrete surround—can be sent to eliminate confusion at playback or reception.

- Additional code can be added to program material when it is originally mixed so that subjectively constant, dialogue-keyed loudness is maintained as the listener switches between program sources. No alteration of program dynamics is involved other than playback volume.

- Decoders can be designed to provide optimum mixdowns from multichannel programming, such as a matrix-encoded two-track mix for analog Dolby Surround decoding, a conventional stereo mix, or even a mono mix.

- When programs with wide dynamic range, such as movie soundtracks, are played at low volume, the system can apply appropriate compression to preserve low-level content. The degree of compression can be made to vary according to need.

- The listener can program the home system's Dolby Digital decoder to route non-directional low bass only to those channels in the system that have wide-range speakers or subwoofers.

Designed to take full advantage of human auditory masking, Dolby Digital divides the audio spectrum of each channel into narrow frequency bands of different sizes optimized with respect to the frequency selectivity of human hearing. This makes it possible to sharply filter coding noise so that it is forced to stay very close in frequency to the audio signal being coded. By reducing or eliminating coding noise wherever there are no audio signals to mask it, the sound quality of the original signal can be subjectively preserved.

Bits are distributed among the filter bands as needed by the particular frequency spectrum or dynamic nature of the program. A built-in model of auditory masking allows the coder to alter its frequency selectivity (as well as time resolution). By allowing the coder this kind of flexibility, a sufficient number of bits can be used to describe the audio signal in each band, therefore ensuring noise is fully masked.

Dolby Digital also decides how the bits are distributed among the various channels from a common bit pool. This technique allows channels with greater frequency content to demand more data than sparsely occupied channels, for example, strong sounds in one channel provide masking for noise in other channels.

Dolby Digital's sophisticated masking model and shared bitpool arrangement are essential factors in its extraordinary spectrum efficiency. Furthermore, where other coding systems have to use considerable (and precious) data to carry instructions for their decoders, Dolby Digital can use proportionally more of the transmitted data to represent audio, which means better sound quality.

Dolby Digital can process 20-bit digital audio signals over a frequency range from 20 Hz to 20 kHz ±0.5 dB (-3 dB at 3 Hz and 20.3 kHz). The low-frequency effects channel covers 20 to 120 Hz ±0.5 dB (-3 dB at 3 and 121 Hz). Sampling rates of 32, 44.1, and 48 kHz are supported, while data rates range from as low as 56 kb/s for a single mono channel to as high as 640 kb/s for multiple channels.

Appendix B
Latency Values

The latency of the DP562 is defined as the difference in time between the beginning of a valid Dolby Digital frame entering the DP562 and the output of the first PCM sample represented by that frame. Note that at any sample rate, a Dolby Digital frame represents 1536 audio samples. The DP562 accepts data over both professional and consumer industry standardized interfaces. The professional interface is specified by SMPTE[1]. The consumer interface is specified by IEC 61937[2]. Each interface specification includes a specification for the latency of the decoder with respect to a reference point in the data stream. The DP562 has a small additional delay over that specified in the SMPTE or IEC standards, due to some extra processing that is performed.

Section B.1 below specifies the DP562 excess delay with respect to the professional SMPTE specification. Section B.2 below specifies the excess delay with respect to the consumer IEC specification. Section B.3 describes the latency (absolute delay with respect to the beginning of the Dolby Digital frame) as it appears on either type of interface.

B.1 SMPTE Specification (professional AES3)

The SMPTE specification describes placement of the Dolby Digital frame on the AES3 interface such that a decoder with the specified latency will produce audio that is timed correctly. The specified decoder latency is exactly one Dolby Digital frame period, with respect to the reference point. The reference point is the first bit of the Dolby Digital frame occurring on the interface. The SMPTE specification allows equipment that outputs Dolby Digital streams on the AES3 interface to know when, with respect to time code or associated video, to output each Dolby Digital frame of data in order to allow a standard decoder to reproduce audio at the proper presentation time. The specification also allows equipment that receives the Dolby Digital data to know when the audio in each frame should be reproduced.

Another way to maintain synchronization is by using time stamps. The SMPTE specification provides for a time stamp to be associated with an audio frame of data. The time stamp can indicate the time code value that applies to an identified sample within a Dolby Digital frame. The time stamp can also indicate an absolute time offset of the stream so that a Dolby Digital encoder, for example, can identify its own latency to another piece of equipment, for example a MPEG multiplexer.

---

[1] Proposed SMPTE Standard for Television – Format for Non-PCM Audio and Data in AES3 – ATSC A/52 (AC-3) Data Type, Second Draft, W25.008, Aug. 9, 1999. [Note: Check with SMPTE for availability of this standard.]

[2] Committee Draft, "Interface for non-linear PCM encoded audio bitstreams applying IEC 60958", 15 March 1998, IEC 100C/196/CDV. [Note: Check with IEC for availability of this standard.]

Table B1 shows the SMPTE Standard specified decoder latency with respect to the first word of the Dolby Digital frame (which is the reference point).

Table B-1 SMPTE specified Latency

Sample Rate	Decoder latency with respect to reference point
48 kHz	32 ms
44.1 kHz	34.83 ms
32 kHz	48 ms

B.1.1 DP562 Delay with respect to SMPTE specification

Table B2 shows the DP562 excess delay with respect to the latency specified in the SMPTE Standard.

Table B-2 DP562 additional delay with respect to SMPTE AES3 specified latency

Sample Rate	*Disabled* and *Reference Detect* Modes	*Silent Switch Mode*
48 kHz	0 ms	6.6 ms
44.1 kHz	0.87 ms	8.0 ms
32 kHz	0.4 ms	10.3 ms

B.2 IEC 61937 Specification (consumer SPDIF)

Like the SMPTE professional specification, the IEC consumer specification also describes placement of the Dolby Digital frame on the SPDIF interface such that a decoder with the specified latency will produce correctly timed audio. The specified decoder latency is exactly one Dolby Digital block period, with respect to the reference point. Note that since there are 6 blocks in a frame, the latency specification is for 1/6 of a frame time, or the time span of 256 audio samples. In the IEC specification, the reference point is defined as the point 2/3 of the way through the frame of Dolby Digital data. Since the frame size depends on the data rate, the reference point follows the beginning of the frame by a variable amount based on the data rate.

Table B3 shows the latency specified by IEC 61937, with respect to the beginning of the Dolby Digital frame on the SPDIF interface.

Table B-3 IEC 61937 specified latency, with respect to beginning of data burst.

Data Rate	Sample Rate = 48 kHz	Sample Rate = 44.1 kHz	Sample Rate = 32 kHz
56 kbps	6.11 ms	6.65 ms	9.17 ms
64 kbps	6.22 ms	6.77 ms	9.33 ms
80 kbps	6.44 ms	7.01 ms	9.67 ms
96 kbps	6.67 ms	7.25 ms	10.00 ms
112 kbps	6.89 ms	7.49 ms	10.33 ms
128 kbps	7.11 ms	7.74 ms	10.67 ms
160 kbps	7.56 ms	8.22 ms	11.33 ms
192 kbps	8.00 ms	8.70 ms	12.00 ms
224 kbps	8.44 ms	9.19 ms	12.67 ms
256 kbps	8.89 ms	9.67 ms	13.33 ms
320 kbps	9.78 ms	10.64 ms	14.67 ms
384 kbps	10.67 ms	11.61 ms	16.00 ms
448 kbps	11.56 ms	12.57 ms	17.33 ms
512 kbps	12.44 ms	13.54 ms	18.67 ms
576 kbps	13.33 ms	14.51 ms	20.00 ms
640 kbps	14.22 ms	15.48 ms	21.33 ms

B.2.1   DP562 Delay with respect to IEC 61937 specification for *disabled* or *reference detect* modes

Table B4 gives the additional delay of the DP562 with respect to IEC 61937, when the DP562 is in *Disabled* or *Reference Detect* modes. The values shown indicate that the audio output of the DP562 is some milliseconds late compared to an ideal decoder that exactly meets the IEC 61937 specification.

Table B-4 DP562 additional delay with respect to the IEC 61937 specified latency for *Disabled* or *Reference Detect* modes.

Data Rate	Sample Rate = 48 kHz	Sample Rate = 44.1 kHz	Sample Rate = 32 kHz
56 kbps	1.19 ms	2.35 ms	2.93 ms
64 kbps	1.18 ms	2.33 ms	2.97 ms
80 kbps	1.26 ms	2.39 ms	3.13 ms
96 kbps	1.23 ms	2.35 ms	3.30 ms
112 kbps	1.21 ms	2.41 ms	3.47 ms
128 kbps	1.19 ms	2.37 ms	3.63 ms
160 kbps	1.14 ms	2.38 ms	3.97 ms
192 kbps	1.20 ms	2.50 ms	4.30 ms
224 kbps	1.16 ms	2.51 ms	4.63 ms
256 kbps	1.21 ms	2.53 ms	4.97 ms
320 kbps	1.22 ms	2.66 ms	5.63 ms
384 kbps	1.23 ms	2.70 ms	6.30 ms
448 kbps	1.14 ms	2.83 ms	6.97 ms
512 kbps	1.16 ms	2.86 ms	7.63 ms
576 kbps	1.17 ms	2.99 ms	8.30 ms
640 kbps	1.18 ms	3.13 ms	8.97 ms

B.2.2. DP562 Delay with respect to IEC 61937 specification *for Silent Switch* mode

Table B5 gives the additional delay of the DP562 with respect to IEC 61937, when the DP562 is in *Silent Switch* mode. The delay is increased due to the buffering needed to minimize audible glitches from appearing at the output when the input data is switched between Dolby Digital and PCM.

Table B-5 DP562 additional delay with respect to the IEC 61937 specified latency for *Silent Switch* mode.

Data Rate	Sample Rate = 48 kHz	Sample Rate = 44.1 kHz	Sample Rate = 32 kHz
56 kbps	32.49 ms	36.15 ms	49.13 ms
64 kbps	32.38 ms	36.03 ms	48.97 ms
80 kbps	32.16 ms	35.79 ms	48.63 ms
96 kbps	31.93 ms	35.55 ms	48.30 ms
112 kbps	31.71 ms	35.31 ms	47.97 ms
128 kbps	31.49 ms	35.07 ms	47.63 ms
160 kbps	31.04 ms	34.58 ms	46.97 ms
192 kbps	30.60 ms	34.10 ms	46.30 ms
224 kbps	30.16 ms	33.61 ms	45.63 ms
256 kbps	29.71 ms	33.13 ms	44.97 ms
320 kbps	28.82 ms	32.16 ms	43.63 ms
384 kbps	27.93 ms	31.20 ms	42.30 ms
448 kbps	27.04 ms	30.23 ms	40.97 ms
512 kbps	26.16 ms	29.26 ms	39.63 ms
576 kbps	25.27 ms	28.29 ms	38.30 ms
640 kbps	24.38 ms	27.33 ms	36.97 ms

B.3 DP562 Latency (Absolute delay) specification

This section specifies the latency (absolute delay) of the DP562 with respect to the beginning of the data burst on either the AES3 or SPDIF interfaces.

B.3.1 DP562 latency for AES3 professional interface

Table B6 specifies the DP562 latency with respect to the beginning of the Dolby Digital data packet on the AES3 professional interface. In *Disabled* and *Reference Detect* modes, the latency is nominally one sample frame. In *Silent Switch* Mode, there is approximately 7–10 ms additional decoding latency in order to minimize audio artifacts during bitstream switching.

Table B-6 DP562 latency for AES3 interface (and for SPDIF interface in *Silent Switch* mode).

Sample Rate	Disabled and Reference Detect Modes	Silent Switch Mode
48 kHz	32 ms	38.6 ms
44.1 kHz	35.7 ms	42.8 ms
32 kHz	48.4 ms	58.3 ms

B.3.2 DP562 latency for SPDIF interface

Table B7 specifies the DP562 latency with respect to the beginning of the Dolby Digital data packet on the SPDIF consumer interface, when the DP562 is in *Disabled* or *Reference Detect* modes. SPDIF latency values are identical to professional latency values in *Silent Switch* mode. For those values, refer to the table in B6 above.

Table B7 DP562 latency for SPDIF interface in *Disabled* and *Reference Detect* modes.

Data Rate	Sample Rate = 48 kHz	Sample Rate = 44.1 kHz	Sample Rate = 32 kHz
56 kbps	7.3 ms	9 ms	12.1 ms
64 kbps	7.4 ms	9.1 ms	12.3 ms
80 kbps	7.7 ms	9.4 ms	12.8 ms
96 kbps	7.9 ms	9.6 ms	13.3 ms
112 kbps	8.1 ms	9.9 ms	13.8 ms
128 kbps	8.3 ms	10.1 ms	14.3 ms
160 kbps	8.7 ms	10.6 ms	15.3 ms
192 kbps	9.2 ms	11.2 ms	16.3 ms
224 kbps	9.6 ms	11.7 ms	17.3 ms
256 kbps	10.1 ms	12.2 ms	18.3 ms
320 kbps	11.0 ms	13.3 ms	20.3 ms
384 kbps	11.9 ms	14.3 ms	22.3 ms
448 kbps	12.7 ms	15.4 ms	24.3 ms
512 kbps	13.6 ms	16.4 ms	26.3 ms
576 kbps	14.5 ms	17.5 ms	28.3 ms
640 kbps	15.4 ms	18.6 ms	30.3 ms

Appendix C
Dynamic Range Compression, Dialogue Normalization, and Downmixing for Dolby Digital

C.1 Introduction / Audio Features

Dolby Digital (AC-3) tailors audio reproduction to specific needs through it's many built-in system features. When encoding with Dolby Digital, audio programs are encoded with the same degree of spatiality and dynamic range deemed necessary by the content producers. However, when these programs are reproduced with downmixing enabled or at reduced listening levels, it is important to preserve as much of the program content and the producer's original intent as possible. Downmixing and dynamic range compression aid in this task while the dialogue normalization feature helps unify the subjective listening level of all programs.

Two variables within the Dolby Digital bitstream, *dynrng* and *compr,* provide automatic gain control in the decoder (based upon information developed during the Dolby Digital encoding process). Dynamic range compression information, represented by the variable *dynrng*, is useful for situations such as late night viewing of programs with a wide dynamic range or for program material that needs significant compression for improved coverage and intelligibility. *Dynrng* also provides the peak limiting required during downmixing to avoid peak level overload. *Compr*, the variable for heavy compression, provides peak limiting only and is used as an extra measure of protection against peak level overload when the audio signal must follow an RF remodulation path for playback or for late night listening when the least amount of dynamic range (lower signal peaks to dialogue) is desired. Encoding dynamic range information into a Dolby Digital bitstream allows you to customize compression levels in the listening environment to suit your needs as well as automatically providing for reduced dynamic range where appropriate or required.

The variable *dialnorm* provides information about average dialogue level (or program level) so that dialogue normalization can be performed at the decode stage. When dialogue normalization is applied properly, all programs are reproduced at the same subjective loudness, i.e., the average dialogue level remains constant from program to program. Large changes in level between different programs and even between program material and commercials can be avoided.

The process of downmixing also allows Dolby Digital to cater to a large audience while still providing for the individual needs of a user. A 5.1-channel program can be downmixed at the decoder (such as a DVD player or DTV set-top box) to create optimized two-channel or even optimized mono reproductions. The original 5.1-channel material is un-compromised and available to a consumers with a Dolby Digital multi-channel decoder, while consumers with a Dolby Pro Logic, stereo, or mono playback systems can enjoy the program in a format appropriate for their playback system.

C.2   Operational Modes

The following operational modes have been defined for implementing dynamic range compression and dialogue normalization within Dolby Digital.

C.2.1.   Line Mode

The Line Mode is useful for products generating a baseband "line out" signal. The amount of compression that is applied may be scaled down or even defeated if downmixing is not in operation by the use of a scale factor between 0.0 and 1.0, where 0.0 is no compression and 1.0 is full compression.

Features:

- Dialogue normalization always enabled
- Dialogue reproduced at a constant level (-31 dBFS)
- *Dynrng*-style compression used
- Compression scaling allowed (low-level boost adjustment only)
- High-level cut compression scaling allowed, if not downmixing

Dynrng values protect against peak level overload when downmixing. Therefore, no scaling of the high-level cut is allowed.

C.2.2   RF Mode

The RF mode is useful for products generating a downmixed signal for subsequent RF channel remodulation. This mode results in the greatest amount of peak limiting. RF Mode may also be used in consumer products as a late night listening mode with less dynamic range than Line Mode (lower signal peaks to dialogue).

Features:

- Dialogue normalization always enabled
- Dialogue reproduced at a constant level (-20 dBFS) in RF remodulated outputs
- *Compr*-style compression used
- Compression scaling not allowed
- +11 dB gain shift imposed in RF remodulated outputs

Compr values protect against peak level overload. No scaling of compression is allowed in this mode.

Because full compression is always applied and peak level compression is always at least -11 dB in this mode, the entire signal can be boosted by +11 dB without fear of overload. Note that because of the +11 dB boost, in an RF remodulated output dialogue would be reproduced at -20 dBFS. By setting a maximum of 20 dB of headroom above dialogue, overmodulation of television receivers is protected. The DP562 however does not apply the +11 dB boost to allow easy comparisons of "Line" and "RF" mode compression in either the analog or digital outputs.

C.2.3   Custom Mode

The Custom Mode is a professional listening mode that gives more control over level adjustments. The amount of compression that is applied may be scaled down or defeated at any time without restriction. In addition, dialogue normalization can be enabled or disabled. In the DP562 decoder, this mode can be useful (for example when listening in the "Full" listening mode) for comparing full dynamic range (by leaving dialnorm turned on and scaling the cut and boost to 0.0 or off) and reduced dynamic range (late night listening using the "Line Mode"). Caution must be used however. Depending upon how Custom Mode is setup, it may not represent either the level balance or dynamic range heard in a consumer decoder.

Features:

- Dialogue normalization can be defeated
- If dialogue normalization is enabled, dialogue is reproduced at a constant level (-31 dBFS)
- *Dynrng*-style compression used
- Compression scaling allowed (low-level boost/high-level cut adjustment allowed)
- –11 dB gain shift imposed when downmixing to prevent peak level overload

Custom Mode allows you to adjust both low-level boost and high-level cut scaling parameters.

A restriction when downmixing in Custom Mode is that an -11 dB gain shift is imposed to insure that there is no overload, since compression can be defeated. This provides "fail-safe" protection when downmixing because *dynrng* may not be present to prevent overload. (the –11dB gain shift is recovered in the analog outputs but not in the digital outputs)

C.2.4   No Compression ("None")

In "None" Mode, both compression and dialogue normalization are disabled. None Mode is a professional listening mode used to hear the full level and dynamics in the program material being encoded. It is never allowed in a consumer product and may not represent either the level balance or dynamic range heard in a consumer decoder. Encoding judgments based on this compression mode should not be made.

Features:
- Dialogue normalization disabled
- *Dynrg*-style compression disabled
- –11 dB gain shift imposed when downmixing to prevent peak level overload (The –11 dB gain shift is recovered in the analog outputs but not in the digital outputs.)

C.2.5   Summary of Operational Modes

The following table is a summary of the various operational modes discussed above:

| Mode	Dialogue Normalization On-chip	High Cut Scaling Allowed	Low Boost Scaling Allowed	Dialogue Reproduced At
None	No	No	No	Adjusted in analog hardware
Custom	Selectable	Yes	Yes	-31 dBFS (-42 dBFS if downmixing) if dialogue normalization enabled; adjusted in hardware if disabled.
Line	Yes	Yes, if NOT downmixing	Yes	-31 dBFS
RF	Yes	No	No	-20 dBFS in RF remodulated outputs

C.3   Dialogue Normalization

The audio signals from Dolby Digital transmissions include known dialogue reference levels, referred to as Dialogue Normalization. The dialogue normalization feature essentially corrects for variations in subjective program loudness, and may vary gain from just a few dB to as much as 12 dB typically, although more than 20 dB is possible.

The purpose of dialogue normalization is actually twofold: (1) to ensure that different programs are reproduced at the same subjective loudness; (2) to ensure that the dialogue levels remain constant no matter what operational mode is being used.

C.3.1   Dialogue Normalization Across Programs

Consumers receive audio/video signals from a variety of sources. Traditionally based on analog delivery, audio signals are typically normalized to their peak values to maximize dynamic range through the transmission and recording media. Digital formats are changing the way programs are normalized because it is no longer difficult to maintain signal quality. CDs, laserdiscs, and even VHS Hi-Fi tapes are able to maintain high dynamic range with ease. All are used to deliver programs with peak signal levels some 10 to 20 dB above their normal reference levels. When these programs are compared to the higher density of conventional television broadcasts, they sound as though they are recorded at a lower volume. In fact, the unlimited program may have signal peaks, which far exceed those of the subjectively louder program.

Figure C-1 compares typical program levels from various sources. The heavy marker represents the average dialogue modulation level, and the shaded area represents the maximum signal headroom available above that point.
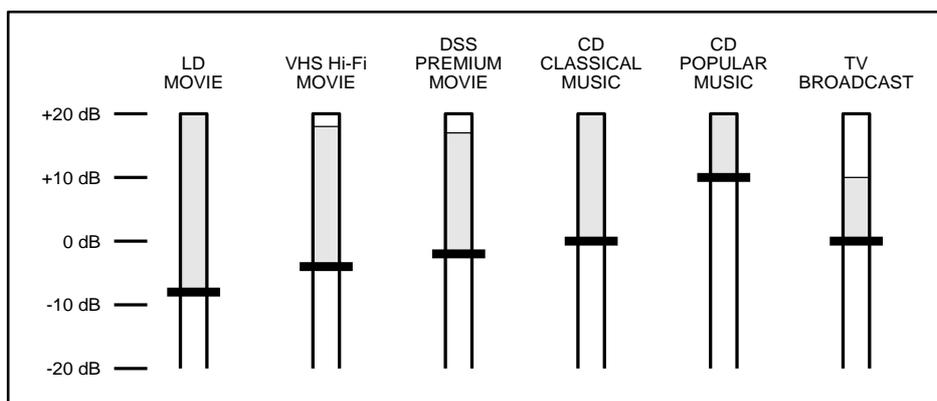
Figure C-1. Audio levels from various consumer signal sources.

Figure C-2 shows all the sources normalized for equal listening volume. The gains have been adjusted by some 10–12 dB, essentially to attenuate the "densest" sources in order to match the subjective loudness of the sources that are not compressed. This allows each program to be decoded at the same subjective loudness, so you no longer need to make volume adjustments each time new programs are received.
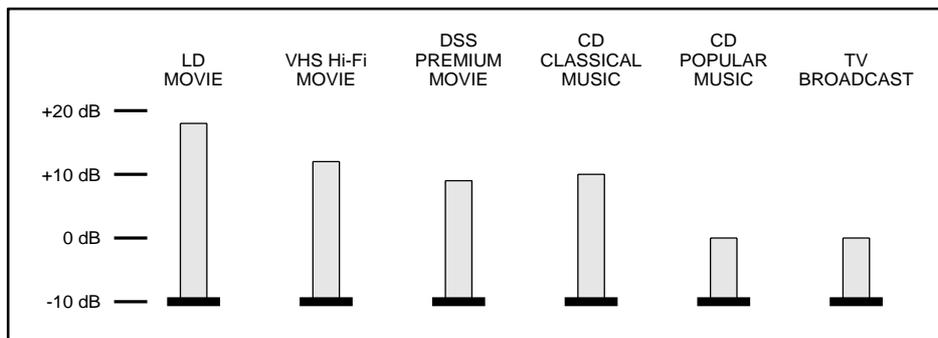
Figure C-2. Audio levels normalized for equal loudness.

C.4   Downmixing Modes

Whenever the number of decoder outputs is less than the number of channels in the Dolby Digital program, downmixing is required. The decoder may change the downmixing process based on the number of speakers in the system or on the number of audio outputs needed, e.g. with headphone listening.

C.4.1   Speaker Output Configurations

The following table describes the behavior of the Dolby Digital decoder when downmixing. Rather than considering the following modes as forced downmix modes, they should be seen as corresponding to speaker output configurations.

		Input						
		3/2	**3/1**	**3/0**	**2/2**	**2/1**	**2/0**	**1/0**
Listening Mode	**Full**	n/a	*2*	*3*	*4*	*2,4*	*3,4*	*3,5*
	3 Stereo	*1*	*1*	n/a	*1,4*	*1,4*	*4*	*5*
	Phantom	*1*	*1,2*	*1,3*	n/a	*2*	*3*	*3,6*
	Stereo	*1*	*1*	*1*	*1*	*1*	n/a	*6*
	Mono	*1*	*1*	*1*	*1*	*1*	*1*	n/a

Key for input configurations:

1 - Imposes –11 dB gain shift if using Custom or None Compression Modes
    (the –11 dB gain shift is recovered in the analog outputs only)
2 - Splits the mono surround into two and attenuates each by 3 dB
3 - No surround channel outputs
4 - No center channel outputs
5 - No left and right channel outputs
6 - Routes center to left and right

Appendix D
Glossary

AES/EBU Format. Audio Engineering Society/European Broadcasting Union format.

Algorithm. A procedure for solving a mathematical problem (finding the greatest common divisor) in a finite number of steps that frequently involves repetition of an operation; a step-by-step procedure for solving a problem or accomplishing some end especially by a computer.

Compr. The variable used for heavy compression.

CRC (Cyclical Redundancy Checksum). It can verify code to see if the transmission is error free.

Custom mode. A professional compression mode similar to Line Mode that allows Dialnorm to be turned on or off and adjust the high level cut and low level boost compression.

Dialnorm. From dialogue normalization. It provides information about average program listening level so that dialogue normalization can be performed at the decode stage.

Dialogue normalization. The ability to allow all programs to be reproduced at the same subjective loudness with the same average dialogue level.

Dolby Digital (AC-3). Dolby Digital (AC-3) is a perceptual audio coding algorithm that takes advantage of auditory masking and both intra- and inter-channel redundancy to enable the efficient storage and transmission of high-quality digital audio.

Originally conceived as a multichannel coding system, Dolby Digital was first introduced in 1992 to provide digital surround sound in the cinema. Due to its combination of high audio quality, spectrum efficiency, and flexibility, it has since become the multichannel digital audio standard for laser discs, DVDs (digital versatile discs) in NTSC (National Television Standards Committee) countries (optional elsewhere), and DVD-ROMs. In both multichannel and two-channel formats, it is also specified for ATSC digital television broadcasting.

Dolby Digital can process at least 20-bit dynamic range digital audio signals over a frequency range from 20 Hz to 20kHz ±0.5 dB (-3 dB at 3Hz and 20.3kHz). Sampling rates of 32, 44.1, and 48 kHz are supported, with data rates ranging from as low as 32 kb/s for a single mono channel to as high as 640 kb/s for multiple channels. Typical applications include 384 kb/s for 5.1-channel Dolby Digital consumer formats, and 192 kb/s for two-channel distribution.

Dolby Pro Logic. Dolby Pro Logic is Dolby's second generation licensed home surround system. A major advantage of Dolby Pro Logic is the use of an active center channel with its own speaker. Conventional stereo systems create a phantom center channel, which is effective for viewers seated directly in front of the television screen. However, for viewers seated elsewhere, the dialogue can appear to come from off center.

Dolby Pro Logic and the use of an appropriately placed center channel loudspeaker allow the dialogue to be heard straight from the screen. The main left and right stereo speakers are widely spaced for a good spread on music and effects. Dolby Pro Logic decoders also optimally decode surround information which is typically fed to a pair of surround speakers slightly behind and to the left and right of the listener.

Downmixing. The combination (or mix down) of the content of n original channels to produce m channels, where $m < n$.

DVD. Digital versatile disc.

Dynrg. The variable used for dynamic range compression information.

Dynamic range. The range between the loudest and softest sounds that a system can reproduce.

Haas Effect. A psychoacoustic phenomenon whereby the earlier of two identical sounds arriving at the ear will dominate the other.

Input Modes. The following table lists all eight modes and defines which input channel is used for encoding based on the selected mode.

Audio Coding Mode	Encoded Channels
1+1	L (Ch 1), R (Ch 2)
1/0	C
2/0	L, R
3/0	L, C, R
2/1	L, R, Mono Surround
3/1	L, C, R, Mono Surround
2/2	L, R, Ls, Rs
3/2	L, C, R, Ls, Rs

LFE. Low Frequency Effects.

Line mode. A consumer emulation mode where dialogue normalization is always on and adjustment of the low-level scaling of dynamic range compression parameter is allowed. Adjustment of high-level scaling of dynamic range compression parameter is allowed only when downmixing does not occur.

Listening modes.
Allows monitoring of various downmix modes to emulate consumer applications. This covers both the number of outputs channels from a decoder as well as the number of speakers in the listening environment.

Lt/Rt. Left total and Right total channels of a Dolby Surround encoded stereo signal.

Lo/Ro. Left only and Right only channels of a regular stereo signal.

NTSC. National Television Standards Committee

Pulse Coded Modulation (PCM). The standard form of uncompressed audio data transmission in which amplitude samples of an analog signal are represented by digital numbers.

Phantom. Information heard from a perceived point between two speakers rather than from the speakers directly.

Pink noise. A noise signal whose spectral power distribution is such that the powers per octave, per decade, or in any other equal-percentage section is the same anywhere across the spectrum. For example, pink noise has the same power in the octave between 50 Hz and 100 Hz as in the octave between 10 kHz and 20 kHz.

RF mode. A consumer emulation mode with dialogue normalization, dynamic range compression, and heavy compression always on.

S/PDIF. Sony/Philips Digital Interface standard, also known as Serial/Parallel Digital Interface.