Dolby®

Model DP561

Dolby Digital (AC-3)
Real-Time Encoding System

Users' Manual

Issue 4 Part No. 91396

Users' Manual

For

Model DP561

Dolby Digital (AC-3)

Real-Time Encoding System

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SECTION 7 Software

SECTION 8 Remote Control

1.1 Introduction

The Dolby Model DP561 is a Dolby Digital (AC-3) encoder designed to provide the high performance and configuration flexibility needed for emerging applications of the Dolby Digital standard in new digital audio-video delivery systems and digital storage media. The encoder is housed in an industrial rack mount package and is made up of a PC-compatible motherboard, Dolby Cat. No. 532 I/O board, linear time code board, and DSP board. The multiple DSPs perform Dolby AC-3 encode processing, and are controlled by the host CPU. A menu-driven user interface provides access to setting all user-adjustable AC-3 parameters.

The Model DP561 may be controlled from an external Windows-based PC, using a standard RS-232 serial port link. This feature provides access to all encoder modes and functions, and supports real-time display of encoder status.

The Model DP561 supports channel configurations ranging from single-channel (mono) through the 5.1-channel format used in discrete multi-channel surround applications as defined by SMPTE and ITU-R.

The encoder accepts up to three pairs of digital audio inputs in either AES/EBU or S/PDIF (IEC 958) format, and produces output bit streams which currently range in data rate from 56 to 640 kbps. The output is in the form of a single AES/EBU or S/PDIF data stream, formatted in accordance with Dolby Laboratories' proposal for using this digital audio transmission standard as a means to transport non-linear coded audio data[1].

The Model DP561 incorporates the provision to start or stop the encoding process at defined intervals based on supplying time code in EBU/SMPTE Longitudinal Time Code (LTC) format. This feature is designed to accommodate the production of a contiguous Dolby AC-3 encoded data file from multi-channel audio source material residing on multiple media.

The architecture employed in the DP561 encoder allows for convenient software updates as the encoder algorithm is improved from time to time. When significant changes are made to the encoding process, Dolby Laboratories will supply users with an upgrade diskette.

The purpose of this manual is to describe operational aspects of the Model DP561 encoder, including the hardware architecture, input/output connections, and user-

[1]The format is described in documentation entitled "Annex B, AC-3 Data Stream in IEC 958 Interface"; please refer to this document for detailed specifications of the data formatting. The document describes a consumer mode and a professional mode for formatting the AC-3 data; both formats are supported by the Model DP561.

adjustable encoding parameters. Guidelines and recommendations for AC-3 bit stream parameter values are given for a few representative examples. It is not the intent of this manual to describe in detail the AC-3 algorithm or bit stream syntax; these topics are given thorough treatment in the United States Advanced Television Systems Committee (ATSC) standards documentation for Advanced Television Systems (ATS, or HDTV). The document number is A/52, Digital Audio Compression Standard (AC-3). A copy is included as part of the documentation package for the DP561.

1.2 Features

- Supports full range of AC-3 channel configurations
- Operates at data rates from 56-640 kbps (dependent on channel configuration)
- Time code controllable
- Remote control via RS-232 serial link from an external PC running Microsoft Windows 95® or Microsoft Windows NT® 4.0
- Software upgradable
- Optional DAT-Link+ for connection to SCSI devices for storage of AC-3 data
- Optional "Dolby Digital Recorder" program, which allows recording of Dolby Digital bitstreams to disk using a digital audio sound card under Microsoft Windows

1.3 Dolby 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, DVD video discs in NTSC countries (optional elsewhere), and DVD-ROMs. In both multichannel and two-channel formats it is also specified for ASTC digital television broadcasting.

Please contact Dolby Laboratories for available technical papers on Dolby AC-3.

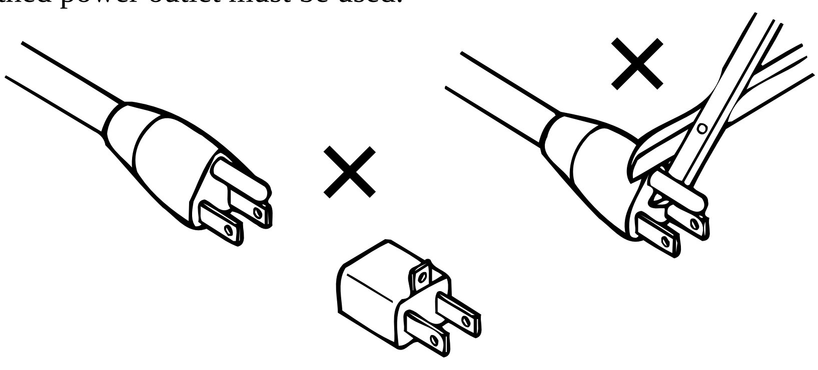
1.4 Regulatory Notices

UL

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

Exposed portions of the power supply that are electrically "hot" may exist when the cover is removed. 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 teminal of the power plug is connected directly to the chassis of the unit. For continued protection against electric shock, a three-pin correctly wired and earthed power outlet must be used.



Do not use a ground-lifting adapter and never cut the ground pin on the three-prong plug.

UK

Connections For the United Kingdom:

WARNING: THIS APPARATUS MUST BE EARTHED
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.

IMPORTANT SAFETY NOTICE

GB

This unit complies with the safety standard IEC65. To ensure safe operation and to guard against potential shock hazard or risk of fire, the following **must** be observed:

- If the unit has a **voltage selector**, ensure that it is set to the correct mains voltage for your **supply**. If there is no voltage selector, ensure that your supply is in the correct range for the input requirement of the unit
- Ensure **fuses** fitted are the **correct rating and type** as marked on the unit.
- The unit **must be earthed** by connecting to a correctly wired and **earthed** power outlet.
- The **power cord** supplied with this unit must be wired as follows:

Live—Brown Neutral—Blue Earth—Green/Yellow

IMPORTANT – NOTE DE SECURITE

F

Ce materiel est conforme à la norme IEC65. 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.

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

Phase—Brun Neutre—Bleu Terre—Vert/Jaune

WICHTIGER SICHERHEITSHINWEIS

D

Dieses Gerät entspricht der Sicherheitsnorm IEC65. Für das sichere Funktionieren des Gerätes und zur Unfallverhütung (elektrischer Schlag, Feuer) sind die folgenden Regeln unbedingt einzuhalten:

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

Phase—braun Nulleiter—blau Erde—grün/gelb

NORME DI SICUREZZA – IMPORTANTE

I

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:

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

Filo tensione—Marrone Neutro—Blu Massa—Verde/Giallo

AVISO IMPORTANTE DE SEGURIDAD

E

Esta unidad cumple con la norma de seguridad IEC65. Para asegurarse un funcionamiento seguro y prevenir cualquier posible peligro de descarga o riesgo de incendio, se han de observar las siguientes precauciones:

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

Vivo—Marrón Neutro—Azul Tierra—Verde/Amarillo

VIKTIGA SÄKERHETSÅTGÄRDER!

S

Denna enhet uppfyller säkerhetsstandard IEC65. För att garantera säkerheten och gardera mot eventuell elchock eller brandrisk, måste följande observeras:

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

Fas—Brun Neutral—Blå Jord—Grön/Gul

BELANGRIJK VEILIGHEIDS-VOORSCHRIFT:

NL

Deze unit voldoet aan de IEC65 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:

- Controleer of de spanningscaroussel op het juiste Voltage staat.
- Gebruik alleen zekeringen van de aangegeven typen en waarden.
- Aansluiting van de unit alleen aan een geaarde wandcontactdoos.
- De netkabel die met de unit wordt geleverd, moet als volgt worden aangesloten:

Fase—Bruin Nul—Blauw Aarde—Groen/Geel

2.1 Overview

Note The exact hardware configuration may change as hardware availability and improvements dictate. Your unit may vary slightly from the description below. The Dolby AC-3 encoder specifications, however, remain unchanged.

The physical encoder consists of a number of printed circuit cards plugged into an ISA (Industry Standard Architecture) bus computer. The audio processing executes on multiple Texas Instruments TMS320C4x DSP chips. Signal input and output is achieved with a custom digital audio interface card. The encoder's operational modes are controlled via a program executing on the ISA bus host CPU; user I/O is achieved with a keyboard and user-supplied monitor, or via remote control from an external PC using an RS-232 serial link.

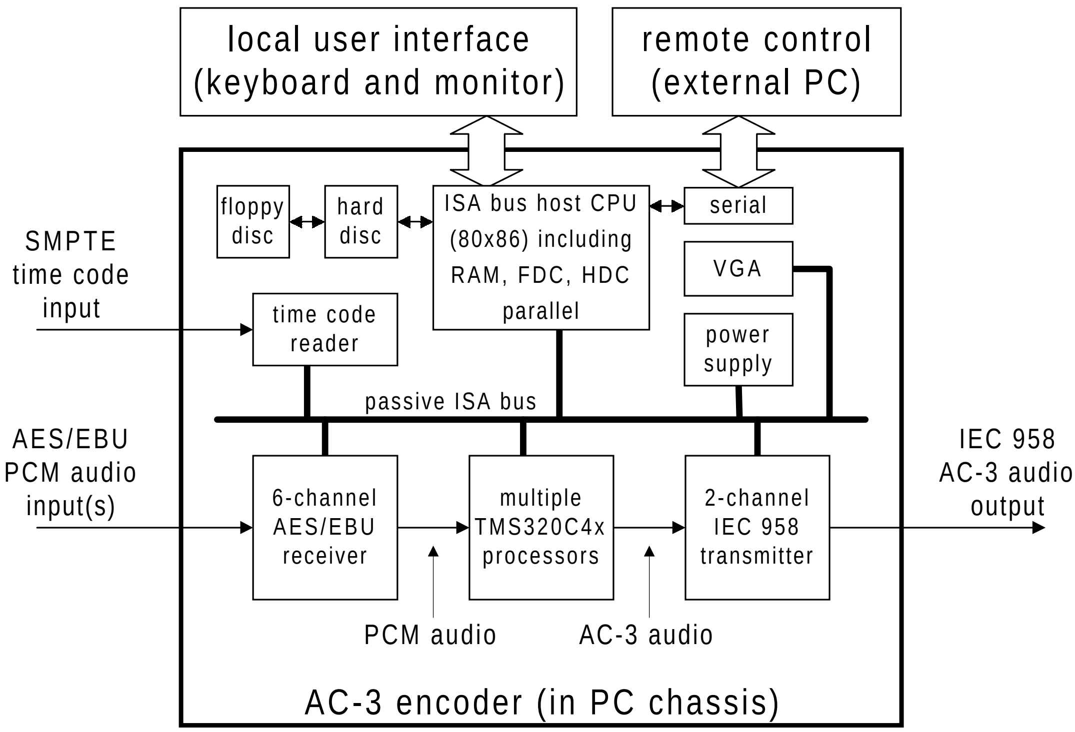


Figure 2.1 Overall Block Diagram

The overall block diagram of the AC-3 encoder is shown in Figure 2.1. The encoder resides in an industrial rack mount chassis which contains a CPU motherboard with ISA bus, DSP card(s), a digital I/O card, a time code reader card, a power supply, a 3.5" floppy disc drive, a hard disc drive, and a VGA display card. The various physical components are described below.

2.2 Host Computer

The host computer includes the CPU itself, disc controllers, keyboard, serial, and parallel ports, and RAM. On the model DP561, these functions are located on a ISA bus plug-in PC card. On the DP561B, the functions are located on the motherboard. The host CPU is responsible for loading the DSP processing cards with the Dolby AC-3 executable code, and also for interpreting user commands which are input via the keyboard or from an external PC via serial link connection. These commands are then relayed through dual-port RAM to the DSP cards. It should be noted that no part of the Dolby AC-3 audio processing algorithm is performed on the host CPU, and that none of the audio signals are present on the ISA bus; these components of the system are used only for booting the DSP chips and for system control signals.

2.3 DSP Cards

The digital signal processing is performed on multiple DSPs installed on one or more DSP cards, interconnected to form a processing network using the DSPs' high speed parallel communication ports. The architecture allows for expanding or reducing the network to an arbitrary number of processors to best match computing requirements for a given algorithm.

The DSPs perform all AC-3 encode processing, and are controlled by the host CPU. Configuring the encoder's operational modes is described in Section 6.

2.4 Digital Audio Input

The Model DP561 encoder accepts multi-channel digital audio inputs via three AES/EBU inputs. These inputs feed a custom digital input/output card on the ISA bus, which in turn feeds the DSP network through a high speed parallel port connection.

Each AES/EBU digital audio pair is identified with generic channel pair numbers one through three which can be configured to a variety of channel assignments via software control as described in Section 6.

2.5 Dolby AC-3 Output Data

The Dolby AC-3 audio data is transmitted out of the DP561 via a single-ended connector residing on the same custom digital i/o card which accepts the input signals. The connector type is BNC.

The AC-3 data stream output is formatted into an IEC 958 signal in accordance with Dolby Laboratories' proposal for using this digital audio transmission standard as a means to transport non-linear coded audio data. The technique involves packetizing each Dolby AC-3 data frame along with header information, (an additional layer wrapped around the Dolby AC-3 frame's own sync word and header information), and sending the packets with a period equal to that of the coded data's period. The data channel is "padded" with zero data between each packet burst.

The format is described in detail in the accompanying ATSC documentation entitled "Annex B, AC-3 Data Stream in IEC 958 Interface;" please refer to this for detailed specifications of the data formatting. The document describes a consumer mode and a professional mode for formatting the Dolby AC-3 data; both formats are supported by the DP561.

2.6 SMPTE Time Code

In addition to the audio inputs, the Model DP561 can accept a SMPTE time code signal by means of a reader card installed on the ISA bus.

The time code signal may be used to accurately control the start or stop time of the encoder. Also, time code values may optionally be included in the output data stream using the same packetizing technique as that of the AC-3 audio data frames. Time code packets are distinguished from audio data packets by means of an identification word within the packet header which has codes for different types of data. An external Dolby AC-3 audio decoder will recognize only the AC-3 data frames and will ignore the time code packets, which are of no consequence to an elementary stream decoder. This method of combining the time code and Dolby AC-3 data outputs can be used by external equipment to create coded data files with timing reference for mastering applications.

Refer to the Appendix A for information about recording AC-3 data into a hard disc file in preparation for DVD mastering.

3.1 System Synchronization

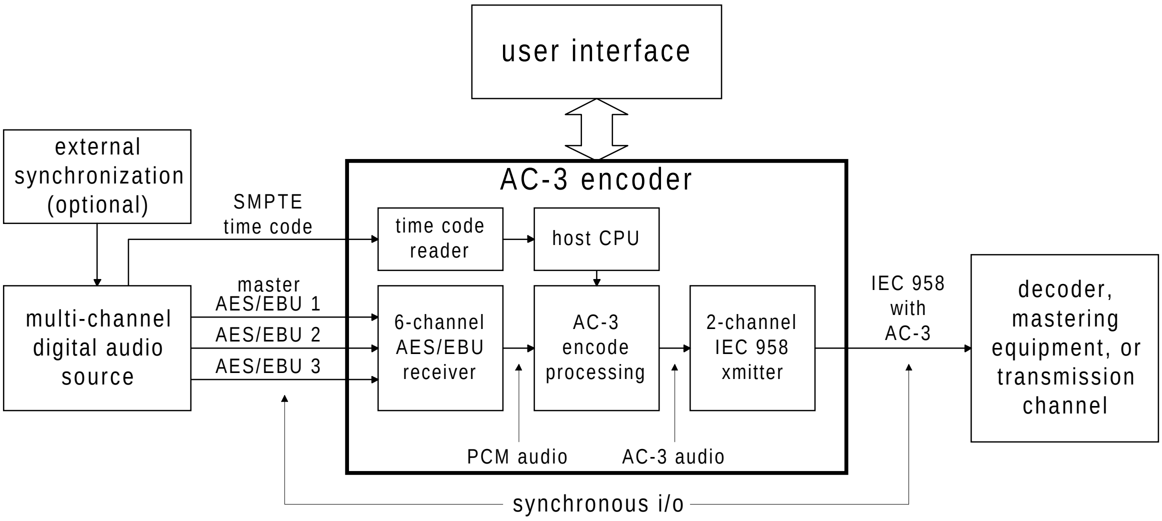


Figure 3.1 System Synchronization

A diagram which depicts system synchronization is shown in Figure 3.1. Each AES/EBU input signal may carry two channels of PCM audio. All clocks in the Dolby AC-3 encoder are locked to the AES/EBU input signal labeled number one. This is the master clock reference for the entire encoder system. Signal must be fed into this input regardless of the number of input channels for any given coding mode. For single channel or two channel encoding, the XLR input labeled number one must be used. For encoding more than two channels, the additional XLR inputs labeled numbers two and three should be used in conjunction with the master input. In this case, all AES/EBU input signals from the digital audio source(s) must be synchronous with each other.

At this time, the DP561 encoder will support input digital audio sampling frequencies of 48 kHz and 44.1 kHz. The encoder will be automatically sense the input sampling frequency and set all internal encoder parameters based on this value. The IEC 958 output signal carrying AC-3 data is synchronous with the input signal(s), and operates at the same sampling frequency.

The encoder may be operated with or without time code. If time code is used, the user must assure that the time code signal being fed to the encoder is synchronous with the input audio data. See Sections 5.3 and 6.2 for details.

4.1 Unpacking

Before unpacking the DP561, inspect the outer carton for shipping damage. If there has been any penetration to the carton, be sure to inspect the unit for any physical damage in those areas.

Several accessories have been included in the packaging. Please compare them with the following list to ensure that there are no missing items:

Rack screws and washers
Power cord
Audio Input Cable Assembly (25-pin D-connector to three XLR female)
Audio Output Cable Assembly (BNC-to-BNC)
Matching transformer, 110 ohm XLR-to-75 ohm BNC (Part No. 54059)
Attenuator, 6 dB , 75 ohm, 1W, BNC (Part No. 19006)
Keyboard
This manual
Digital Audio Compression Standard (AC-3), ATSC Doc. A/52
Dolby AC-3 encoder program disk
DAT-Link+ UNIX driver disk (when this accessory is ordered with the DP561)
Keys
Warranty card (Part No. 91292)
Product Registration card (Part No. 91228)
Various PC software and manuals
Various PC hardware manuals and other documentation
Model DP561 Remote Control installation disk for Windows 95/NT
RS-232 serial link cable
Null modem adapter

Notes

A user-supplied VGA monitor is required.

This manual contains the information necessary to install and operate the Dolby Digital (AC-3) multi-channel encoder. Other manuals, software, and documentation have been supplied for completeness and reference purposes.

4.2 Inspection

Carefully remove the unit from its carton. Remove the plastic wrapping and place on a flat surface. Also remove the keyboard from its packing material. Inspect both for damage.

Rough handling during shipment may result in loose cards inside the DP561. **Therefore, it is very important to inspect the interior of the unit and make sure all ISA bus cards are fully and securely mounted onto the bus.** To gain access to the interior, proceed as follows:

WARNING Be sure that the unit is NOT powered up.

Using a #2 Phillips head screwdriver, loosen the three half-turn retainers along the front apron securing the top cover to the DP561B chassis (model DP561 uses two screws on the rear apron). Raise the rear of the top cover upwards, and pull the cover back and up. Reverse this procedure when re-assembling the unit.

If there are no signs of physical damage, and all cards and internal connectors appear to be in place, proceed to "Voltage Selection" below.

4.2.1 Claims for Shipping Damage

If, in your inspection procedure, you should find physical damage, please notify the carrier immediately. All claims for damage must be filed by the recipient. Dolby Laboratories will be happy to assist where possible.

4.3 Voltage Selection

The DP561 utilizes a power supply with a dual voltage selector (110/230 volts, nominal). Using a small flat-blade screwdriver, move the rear panel voltage selector to the appropriate mains voltage in your area.

Note There are no user-serviceable fuses in the DP561. Contact Dolby Laboratories for service if the unit appears to be inoperative due to an internal power supply problem.

Figures 5.1, 5.2, and 5.3 show the rear panel.

4.4 Jumpers and Switch Settings

The Model DP561does *not* require the user to set any jumpers. All user-configurable settings are made with the keyboard while observing the on-screen user interface.

WARNING The DP561 has numerous jumpers and switches, both inside the unit and on the rear panel. These should *not* be disturbed, as results may be unpredictable. Dolby does not support the use of the DP561's PC platform for applications that are not described in this manual.

5.1 Mounting

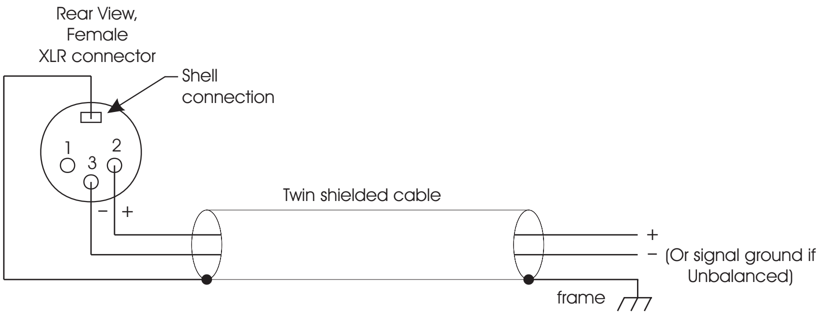
The unit is designed for 19 inch rack mounting and occupies 4U (7") of space. Pick a suitable location to ensure that connections to the keyboard and user-supplied monitor can be made with the available cord lengths.

With its built-in fans, the unit will operate within specifications up to a maximum ambient temperature of 50° C (122° F). Keep in mind that the ambient temperature inside a poorly-ventilated rack may be considerably higher than that in the room.

DP561B - Refer to Figure 5.1 for rear panel connector locations. See Figure 5.2 for connections to the earlier version of the DP561B.

DP561 - Because of its weight, devise a method of supporting the unit at its rear by means of accessories that may be available for your specific rack. Mounting slides are another suitable alternative. If you have a RETMA standard rack, slides can be ordered from the company Thorson West at 650-964-9300. Refer to Figure 5.3 for connector locations, all of which are on the rear panel. PC slots are numbered 1-10, starting at the left when viewing the unit from the rear. PC plug-in cards may not appear in the exact locations shown. If you have a unit with a serial number 500014 or lower, refer to Figure 5.4.

5.2 Audio Connections



For optimum immunity to RF interference, cable shields must be connected to the metallic shell of D-sub connectors or any XLR connectors (XLR connector pin 1 should not be used).

5.2.1 Digital Audio Input

The DP561 accepts multi-channel digital audio via multiple AES/EBU inputs. These inputs feed a custom digital I/O card on the ISA bus, which in turn feeds the DSP network through a high speed parallel port connection. The external audio connections are made using a custom cable supplied with the unit. It has three female XLR type connectors on one end and a 25-pin D-sub connector on the other.

Attach the D-sub connector to the mating connector on the back of the PC chassis (shown in Figure 5.1, 5.2, 5.3, or 5.4 as "DIGITAL IN"). Each XLR input connector carries an AES/EBU digital audio pair. For single channel or two channel encoding, the XLR input labeled number one must be used. For encoding more than two channels, the additional XLR inputs labeled numbers two and three should be used in conjunction with the master input. In this case, all AES/EBU input signals from the digital audio source(s) must be synchronous with each other.

Note: The XLR inputs are labeled with generic channel pair numbers one through three which can be configured to a variety of channel assignments via software control as described in Section 6 Operation.

5.2.2 Digital Audio Output

The digital output of the DP561 is syntactically compliant with AES3-1992 (AES/EBU) and with IEC 958. The physical interface is a female BNC connector which supplies 2 V p-p from a 75 Ohm source.

With appropriate adapters, this output can be made to completely comply with both AES3-1992 (XLR, 3 V p-p, 110 Ohms) and with the new SMPTE 276M standard (BNC, 1 V p-p, 75 Ohms). Use a matching transformer (such as Canare BCJ-XP-TRA) to obtain AES3-1992. Use a 75 Ohm 6 dB BNC-to-BNC pad (such as JFW 75F-006-BNC) to obtain SMPTE 276M.

Attach the supplied BNC cable to the AC-3 output connector on the back of the PC chassis (shown in the following figures). Connect the other end to the succeeding piece of equipment.

5.3 SMPTE Time Code Connections

In addition to the audio inputs, the DP561 accepts SMPTE time code, which can be used to accurately control the start and stop times for the encoder. In addition, the time code may be embedded in the output data stream with its associated audio data. This feature is useful for mastering applications requiring audio/video synchronization in which the audio and video elements may reside on physically different media, and/or may be encoded at different times or places.

Make the time code connection to the encoder via the upper female RCA jack on the back of the unit (shown in the following figures). The lower female RCA jack is a loop-through connection of the time code input signal.

5.4 PC Connections

Make keyboard and monitor connections per standard PC installations, using the connectors provided on the rear of the DP561B, as depicted in Figures 5.1 (see figures 5.2, 5.3,or 5.4 for earlier versions).

Notes

The video card supports a user-supplied VGA monitor.

The DP561 does not make use of the COM2 or LPT1 ports.

5.5 Power

Read the safety information in Section 1. When you are confident that you have observed its provisions, connect AC power to the unit using the supplied power cord. If your monitor is provided with a cord that permits connection to the switched AC output receptacle, make that connection also. Otherwise, connect the monitor's AC cord to a power source separately.

5.6 Remote Interface Connections

The Model DP561 may be remotely controlled by an external PC running Windows 95 or Windows NT 4.0 using an RS-232 serial port link. To enable this function, securely attach the null modem adapter to the COM1 serial port connector at the rear of the DP561 (shown in the following figures). Connect this port to either COM1 or COM2 of the external PC using the supplied serial cable (or any serial cable with all 9 pins wired straight through).

Note: Dolby Laboratories does not supply or recommend specific brands or models of external PC equipment. The system requirements for the external PC are:

- Windows 95® or Windows NT® 4.0 Operating System
- COM1 or COM2 port free for connection to the Model DP561
- The COM port must be able to make a reliable connection at 38400 bps

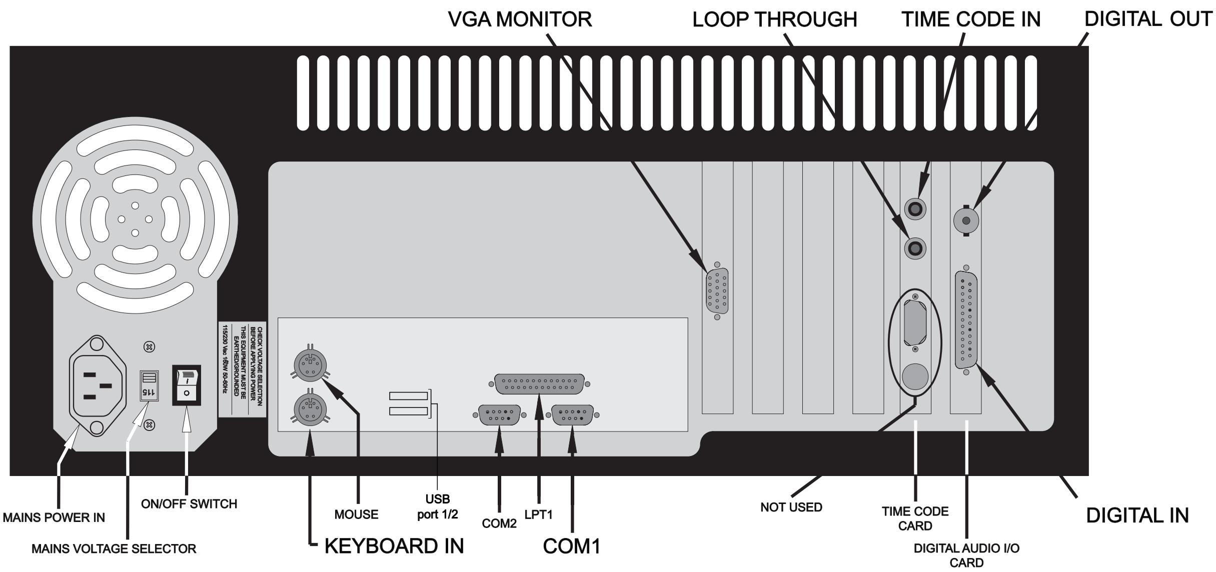


Figure 5.1 Model DP561B Rear Panel Connectors - Current Model

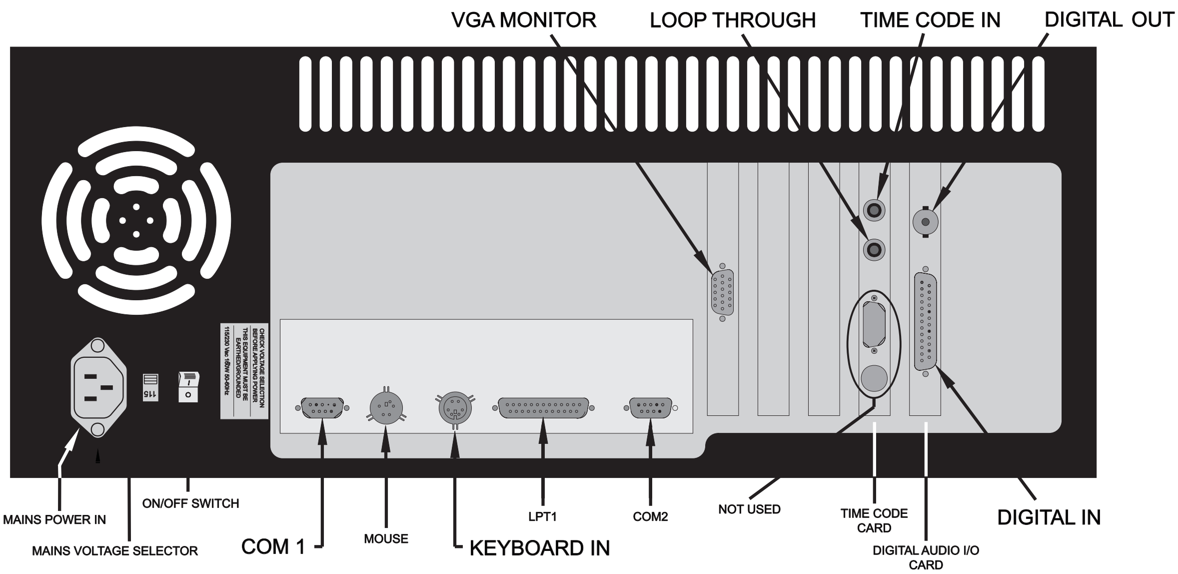


Figure 5.2 Model DP561B Rear Panel Connectors - Earlier Model

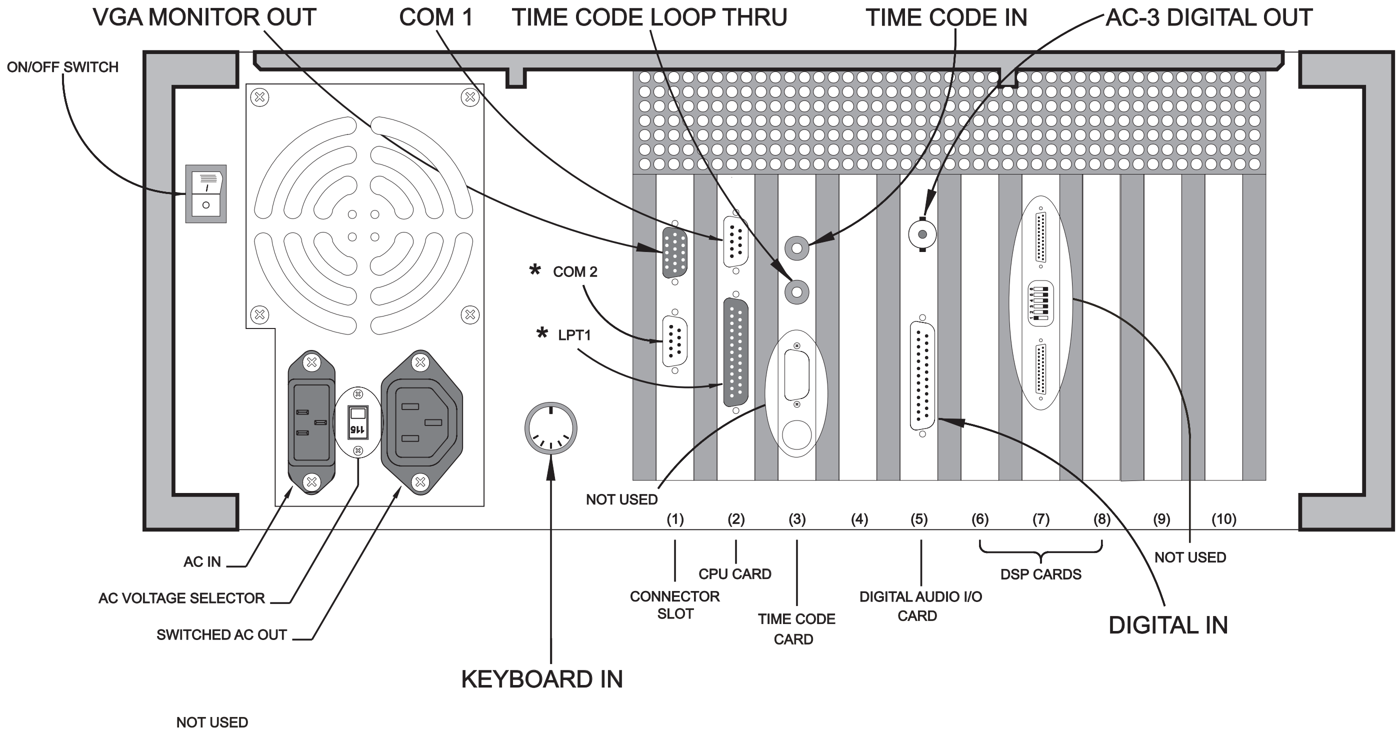


Figure 5.3 Model DP561 Rear Panel Connectors

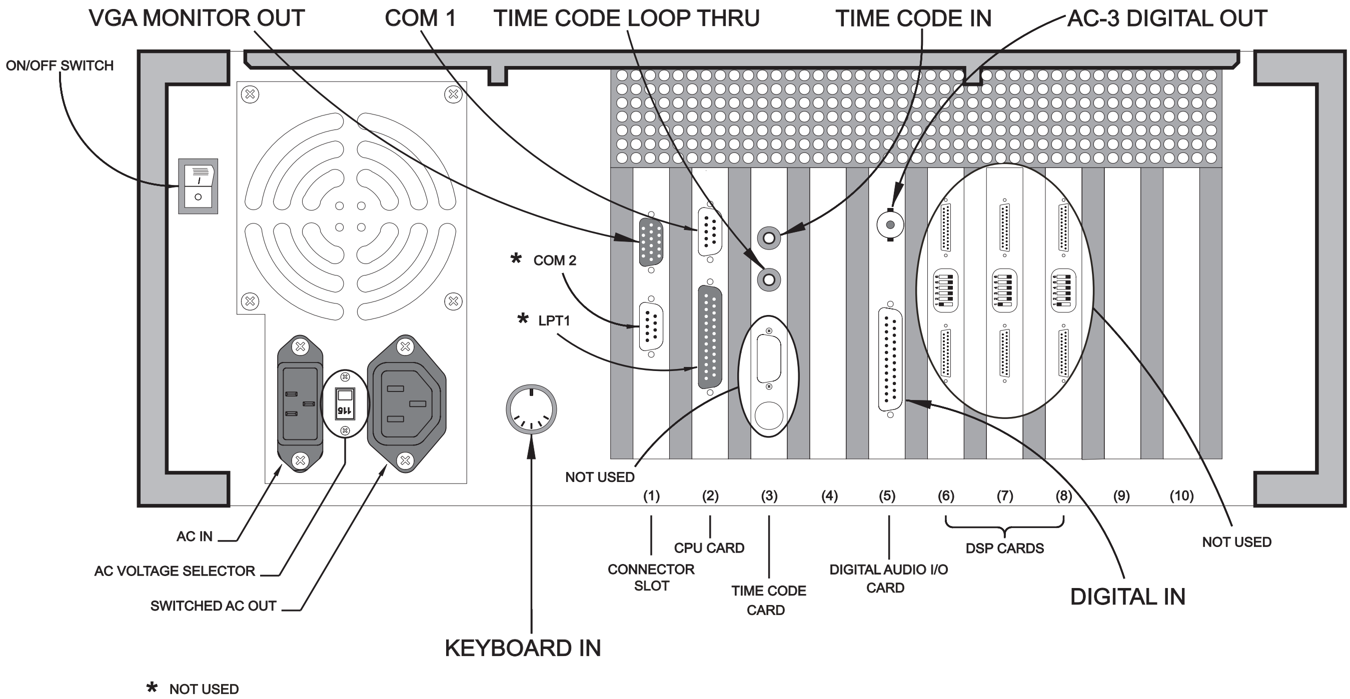


Figure 5.4 Model DP561 Rear Panel Connectors
(serial numbers lower than 500014)

6.1 Dolby AC-3 Parameter Control

After making the proper input and output connections, power may be applied to the DP561. The autoexec.bat file will automatically load the DSP cards with Dolby AC-3 code, and will subsequently invoke the DP561 control program in local control mode. This program is named "ac3ctrl".

In local control mode, the DP561 will only respond to commands entered from the DP561 keyboard. For information on using the remote control features of the DP561, please see Section 6.4.

As delivered from the factory, the DP561 will operate with preset default settings for all parameters, and the PC's screen will display the menu and status screen shown in Figure 6.1. The menu items are grouped into four categories: Encoder Status, Audio Service Configuration, Input/Output Control, and Bit Stream Information.

Dolby AC-3 Encoder: Local Control

Current Time: 00:00:00:00 30 fps
x - Start at: 00:00:00:00:0
y - Stop at: 00:00:00:00:0
r - Resume Enter – Manual stop

Audio Service Configuration

1 - Data Rate: 448 kbps
2 - Bit Stream Mode: complete main
3 - Audio Coding Mode: 3/2
4 - Low Frequency Effects Channel: on

Input/Output Control

5 - Input Channel Assignment: mode 1
6 - Output Format: pro, 32bit + TC
7 - DC Filter (all channels): on
8 - LFE Lowpass Filter (120 Hz): on
9 - Surround Input 3 dB Atten.: off
0 - Surround Channel phase-shift: on
o - Digital Input/Output Settings

Encoder Status

Dolby AC-3 Encoder Version: 6.3.0
Encoder State: encoding
Sampling Frequency: 48.0 kHz
Current Audio Bandwidth: 20.3 kHz
Digital Input Status: no errors

Bit Stream Information

a - Center Downmix Level: 0.707
b - Surround Downmix Level: 0.707
c - Dolby Surround Mode: none
d - Dialog Normalization: -27 dB
e - Language Code: none
f - Audio Production Info: no
g - Mixing Level: none
h - Room Type: none
i - Copyright Bit: copyright protected
j - Original Bit Stream: original
k - Compression: Film, Standard

Q - Quit C - Remote Control S - Save Settings R - Recall Settings

Figure 6.1 AC-3 Control Program Main Screen (local control mode)

In addition to these items, there are fields in the upper left corner of the screen which display time code related information. Each user-adjustable parameter has a corresponding character listed next to it. The user may change a parameter by typing the corresponding character. The control program is case sensitive, so the character must be typed in the same case as shown on screen. The meaning of each menu item is described below.

6.1.1 Audio Service Configuration

Data Rate

Data rates that are supported by the Dolby AC-3 real-time encoder depend on the selected Audio Coding Mode parameter. In general, audio coding modes that include fewer channels in the bit stream have lower data rate limits. Table 6.1 indicates current data rate ranges as a function of the Audio Coding Mode. The default value is "448 kbps."

Audio Coding Mode	Data Rate Range
1/0	56 - 640 kbps
2/0 or 1+1	96 - 640 kbps
3/0 or 2/1	128 - 640 kbps
3/1 or 2/2	192 - 640 kbps
3/2	224 - 640 kbps

Table 6.1 Data Rate Ranges

Bit Stream Mode

This parameter indicates the type of service that the bit stream conveys. The service types are listed in Table 6.2. The default setting is "main audio service: complete main."

Bit Stream Mode	Type of Service
1 (default)	main audio service: complete main (CM)
2	main audio service: music and effects (ME)
3	associated service: visually impaired (VI)
4	associated service: hearing impaired (HI)
5	associated service: dialog (D)
6	associated service: commentary (C)
7	associated service: emergency (E)
8	associated service: voice over (VO) (if audio coding mode = 1/0) main audio service: karaoke (if audio coding mode = 2/0 or higher)

Table 6.2 Bit Stream Mode Types

Audio Coding Mode

This parameter defines the number of full-bandwidth audio channels within the encoded bit stream and also indicates the channel format. The Audio Coding Mode is designated as two numbers, m/n, with m indicating the number of front channels, and n indicating the number of rear (surround) channels. Table 6.3 lists all eight modes and defines which input channel is used for encoding based on the selected mode.

If the mode is set to 1+1, then two completely independent program channels (dual mono) are encoded into the bit stream, and are referenced as Ch1 and Ch2. In this case, additional Bit Stream Information items become active on the Dolby AC-3 control menu which are used to fully describe Ch2. The default setting for this parameter is "3/2."

Audio Coding Mode	Encoded Channels
1+1	L (Ch1), R (Ch2)
1/0	C
2/0	L, R
3/0	L, C, R
2/1	L, R, Ls
3/1	L, C, R, Ls
2/2	L, R, Ls, Rs
3/2 (default)	L, C, R, Ls, Rs

Table 6.3 Audio Coding Mode

Low Frequency Effects Channel On/Off

This menu item enables or disables the Low Frequency Effects (LFE), or sub-woofer channel. The LFE channel can only be enabled if the audio coding mode is 3/0 or higher. The default setting is LFE "on."

6.1.2 Input/Output Control

Input Channel Assignment

This parameter is used to configure the mapping of the physical digital input signals to the proper Dolby-AC-3 encoded channel assignment. Table 6.4 lists the different mappings that can be selected. The number of input channels that are actually encoded depends on the Audio Coding Mode and the LFE setting; it is not determined from the number of digital signals present at the input connector. The encoder reads the channels to be encoded according to the mapping defined in Table 6.4. The default setting for this parameter is "mode 1."

Example
Consider the case of Audio Coding Mode set to 2/0, and LFE channel off. If the Input Channel Assignment is set to mode 1, the encoder will read both Left (L) and Right (R) channel inputs from AES/EBU input number 1; if the Input Channel Assignment were set to mode 2, the encoder would read the Left channel input from Ch A of AES/EBU 1, and the Right channel input from Ch A of AES/EBU 2.

Note AES/EBU 1 is the master input, and must always be present in order for the encoder to lock to the input signals.

When the Audio Coding Mode is set to 1/0, the center channel (C) is encoded. For this configuration, it may be convenient to choose Input Channel Assignment mode 6, because the center channel is mapped to Channel A of AES/EBU 1.

Input Channel Assignment			
Mode	(master) AES/EBU 1 Ch A/Ch B	AES/EBU 2 Ch A/Ch B	AES/EBU 3 Ch A/Ch B
1	L/R	Ls/Rs	C/LFE
2	L/C	R/Ls	Rs/LFE
3	L/Ls	C/Rs	R/LFE
4	L/R	C/LFE	Ls/Rs
5	L/C	Rs/R	Ls/LFE
6	C/L	R/Ls	Rs/LFE

Table 6.4 Input Channel Assignment Mapping

Output Format

This switch selects one of four possible output data formats:

consumer mode;
professional mode 32-bit (A and B frames) data packing;
professional mode 16-bit (A frame, or Ch1) data packing; or,
professional mode 16-bit (B frame, or Ch2) data packing.

These output formats are described in detail in the accompanying ATSC documentation entitled "Annex B, AC-3 Data Stream in IEC 958 Interface;" please refer to this for detailed specifications of the data formatting.

After selecting the desired output format, the user will be prompted as to whether or not to include time code data packets in the output stream. The user should respond with a "y", or "n", to indicate a choice. The default setting for this parameter is 32-bit professional mode, including time code packets.

DC Filter

This switch turns a DC filter either on or off for all input channels. The default setting is "on."

LFE Lowpass Filter

This switch can be used to activate a 120 Hz lowpass filter applied to the Low Frequency Effects input channel. If the digital signal fed to the LFE's input does not contain information above 120 Hz, this filter may be disabled. The default setting is "on."

Surround Input 3 dB Attenuation

This switch can be used to activate a 3 dB attenuation applied to the input of the surround channel(s). The default setting is "off."

Surround Channel Phase-Shift

This switch can be used to activate a 90° phase-shift applied to the surround channel(s) at the input. This feature is useful for generating multi-channel AC-3 bit streams which may be down-mixed in an external 2-channel decoder to create a true Dolby Surround compatible output. The default setting is "on."

Digital Input/Output Settings

This menu item activates a sub-menu which supplies the user with information about the state of the digital audio input signals which are applied to the Model DP561. The sub-menu also allows the user to change certain bits in the AES/EBU or S/PDIF channel status of the DP561 digital output signal containing the AC-3 bit stream.

6.1.3 Bit Stream Information

Bit Stream Information parameters may be modified by typing the number or letter next to the menu item, and selecting the desired value from a list that will appear on the monitor screen. The presence of certain parameters within the bit stream depends on the value of one of the other parameters. In such cases, the dependent menu item will be inactive and cannot be modified until the parameter it depends upon is set to a particular value.

Example

The Dolby Surround Mode menu item is not present within the AC-3 bit stream and may not be changed on the screen unless the Audio Coding Mode is set to 2/0 mode.

Definitions and valid ranges for all control parameters are given below:

Center Downmix Level

This parameter indicates the nominal downmix level of the center channel with respect to the left and right channels. This parameter appears in the bit stream and the menu item is active only when three front channels are in use, i.e., only when the Audio Coding Mode is set to 3/0, 3/1, or 3/2. Table 6.5 lists the values that the user can select for Center Downmix Level; the default value is "0.707" (-3.0 dB).

Center Downmix Level	Mix Level
1 (default)	0.707 (-3.0 dB)
2	0.596 (-4.5 dB)
3	0.500 (-6.0 dB)

Table 6.5 Center Downmix Level Values

Surround Downmix Level

This parameter indicates the nominal downmix level of the surround channel(s) with respect to the left and right channels (consistent with the ITU BR specification). This parameter appears in the bit stream and the menu item is active only when a surround channel is in use, i.e., only when the Audio Coding Mode is set to 2/1, 2/2, 3/1, or 3/2. Table 6.6 lists the values that the user can select for Surround Downmix Level; the default value is "0.707" (-3.0 dB).

Surround Downmix Level	Mix Level
1 (default)	0.707 (-3.0 dB)
2	0.500 (-6.0 dB)
3	0 (-∞ dB)

Table 6.6 Surround Downmix Level Values

Dolby Surround Mode

This parameter indicates whether or not a 2-channel Dolby AC-3 bit stream is conveying a Dolby Surround encoded program. This parameter appears in the bit stream and the menu item is active only when operating in the two channel mode, i.e., only when the Audio Coding Mode is set to 2/0. This information is not used by the Dolby AC-3 decoding algorithm, but may be used by other portions of the audio reproduction equipment, such as a Dolby Pro Logic Surround decoder. Table 6.7 lists the values that the user can select for Dolby Surround Mode. Because the default Audio Coding Mode is 3/2, this parameter is not present in the bit stream after power-up, and "none" is displayed. If the 2/0 mode is selected, then the default setting for this parameter is "not indicated."

Dolby Surround Mode	Indication
1	not indicated
2	NOT Dolby Surround encoded
3	Dolby Surround encoded

Table 6.7 Dolby Surround Mode Indications

Dialog Normalization

This parameter indicates how far the average dialog level of the encoded program is below digital 100%. Valid values are 1 to 31, which are interpreted as -1 dB to -31 dB with respect to digital 100%. The value of Dialog Normalization affects the sound reproduction level. Please refer to the ATSC standard AC-3 document for a thorough definition of the "dialnorm" parameter. The default value is "27" (-27 dB).

A second value for this parameter type will become active when the Audio Coding Mode is 1+1; this second item applies to the second independent channel (Ch2) residing within the bit stream.

Language Code

This is an eight bit code which represents the language of the main audio service. Valid values are hexadecimal 0x0 to 0xff (256 values). The default setting is "0x0," which indicates that the language code is unknown or not applicable.

A second value for this parameter type will become active when the Audio Coding Mode is 1+1; this second item applies to the second independent channel (Ch2) residing within the bit stream.

At present, there are no professional or commercial applications that require the AC-3 elementary bitstream language code. For this reason, Dolby has disabled access to this feature.

Audio Production Info Exists

This menu item indicates whether the Mixing Level and Room Type parameters exist within the bit stream. The default setting is "no."

A second value for this parameter type will become active when the Audio Coding Mode is 1+1. This second item applies to the second independent channel (Ch2) residing within the bit stream.

Mixing Level

This parameter appears in the bit stream and the menu item is active only when the Audio Production Info Exists parameter is set to "yes." This parameter indicates the acoustic sound pressure level of the dialog level during the final audio mixing session. The valid range is 0 to 31. Because the default state for Audio Production Info Exists is "no", this parameter is not present in the bit stream after power-up, and "none" is displayed. If Audio Production Info Exists is changed to "yes", then the default value for this parameter is "25."

A second value for this parameter type will become active when the Audio Coding Mode is 1+1 *and* the Audio Production Info Exists parameter for Ch2 is set to "yes." This second item applies to the second independent channel (Ch2) residing within the bit stream.

Room Type

This parameter appears in the bit stream and the menu item is active only when the Audio Production Info Exists parameter is set to "yes." This parameter indicates the type and calibration of the mixing room used for the final audio mixing session. The value of Room Type is not typically used within the Dolby AC-3 decoding algorithm, but may be used by other parts of the audio reproduction equipment. Table 6.8 lists the values that the user can select for Room Type. Because the default state for Audio Production Info Exists is "no", this parameter is not present in the bit stream after power-up, and "none" is displayed. If Audio Production Info Exists is changed to "yes", then the default value for this parameter is "small room."

A second value for this parameter type will become active when the Audio Coding Mode is 1+1 *and* the Audio Production Info Exists parameter for Ch2 is set to "yes." This second item applies to the second independent channel (Ch2) residing within the bit stream.

Room Type	Indication
1	Not indicated
2	Large room, X curve monitor
3	Small room, flat monitor

Table 6.8 Room Type Indications

Copyright Bit

This parameter sets the value of a single bit within the Dolby AC-3 bit stream. If this bit has a value of 1, the information in the bit stream is indicated as protected by copyright. It has a value of 0 if the information is not copyright protected. The default setting is "copyright protected."

Original Bit Stream

This parameter sets the value of a single bit within the Dolby AC-3 bit stream. This bit has a value of "1" if the bit stream is an original. It has a value of "0" if the bit stream is a copy of an original bit stream. The default setting is "original."

Compression

This parameter selects the Dynamic Range Compression mode used by the encoder to generate Dynamic Range Gain Words within the Dolby AC-3 bit stream. Table 6.9 lists the available modes and their intended application.

Please refer to the ATSC Standard AC-3 document for a detailed description of AC-3 Dynamic Range Compression.

Compression	Indication
1	Off
2	Film, Standard
3	Film, Light
4	Music, Standard
5	Music, Light
6	Speech

Table 6.9 Dynamic Range Compression Modes

A Dolby AC-3 decoder will use the Dynamic Range Gain Word to reduce the audio program's dynamic range unless the feature is disabled on the decoder by the user who desires program reproduction with the original dynamic range.

6.1.4 Encoder Status

The fields listed under Encoder Status supply the user with information about the current status of the Model DP561 encoder. The meaning of each field is described below.

Dolby AC-3 Encoder Version

This field displays the AC-3 software version running on the DSPs.

Encoder State

This field indicates the actual state of the encoder. The primary states are "encoding" and "stopped", although the encoder may pass through other states briefly as it transitions between these two states.

Sampling Frequency

The Model DP561 will automatically detect the sampling frequency of the digital audio input signal applied to the input. The detected value will be displayed in this field. The encoder currently supports two sampling frequencies: 48 kHz, and 44.1 kHz. The digital audio signal applied to the encoder's input must operate at one of these rates.

The Dolby AC-3 encoder software will select internal data tables and data structure sizes that correspond to the detected input sampling frequency. The sampling frequency of the output serial data stream will track that of the input signal.

Current Audio Bandwidth

This field indicates the bandwidth of the encoded audio signal. The DP561 will automatically adjust this parameter for optimal performance based on the user-selected data rate and audio coding mode. In general, decreasing the data rate for a given number of coded audio channels may cause the encoder to reduce the audio bandwidth in order to maintain audio quality.

Digital Input Status

This field indicates whether the digital audio inputs exhibit any error conditions, such as loss of lock, low signal integrity, input bit errors, or mismatch between measured input sample frequency and the channel status sample rate information. If this field indicates an error, the underlying cause can be found by checking the Digital Input/Output Settings screen.

6.1.5 Time Code Fields

The fields in the top left corner of the screen describe the current time code status and parameters. The meaning of each field is described below.

Current Time

This field displays the most recent time code value received by the SMPTE reader card, as well as the current SMPTE time code format. The time code format will be one of the following: "24 fps", "25 fps", "29.97 fps", "29.97 fps df", "30 fps", or "unknown" if the format can not be determined. If a time code signal is not present at the input to the SMPTE reader card, this field will read "Time Code Input not active".

Start Time

This field indicates the time code at which the DP561 will automatically start encoding (assuming it was previously stopped).

Stop Time

This field indicates the time code at which the DP561 will automatically stop encoding (assuming it was previously running).

For more information on using SMPTE time code to control the DP561, please see Section 6.2.

6.1.6 Saving the Parameter State

It is not mandatory that the user quit the control program, "ac3ctrl", before powering the DP561 down. However, in order to save the state of the encoding parameters between power cycles, the user should quit the control program by typing upper case "Q" before shutting the power off. This is because quitting the control program causes it to write a state, or configuration file to the PC's disc.

When the DP561 is turned on again, it will recall the encoding parameters and state from the saved configuration file. This is described in further detail below in the Command-Line Control section.

If the user wishes to reset all encoding parameters to the factory default settings, this can be accomplished by deleting the configuration file. This is because when the control program is invoked, it looks for the configuration file; if it is not found, then all parameters will be set to the default values. The name of the configuration file is "ac3state.cfg", and it is written in the directory where the ac3ctrl program resides. From the factory, this directory is "c:\code\ac3ctrl."

The latest version of the DP561 software uses a different file format for its configuration files than was used in the original software release. The new software is able to read older configuration files, but will only write configuration files in the new format. If the DP561 finds itself about to overwrite an old-style configuration file, you will be asked whether or not you want to proceed.

If you are upgrading your DP561, you will encounter this issue very quickly. The first time you run the new software after upgrading, it will read in the encoder parameters from an old-style configuration file. Then, when you quit the new software for the first time, the DP561 will attempt to save the state in the new format to the same file. As a result, you will be asked if you want to overwrite the old configuration file. If you decline, the old configuration file will not be modified in any way. There is no harm in updating the file format (unless you plan to continue using the old DP561 software).

6.2 Time Code Operation

The user may choose to start and/or stop the encoding process at specific points by supplying time code as an input to the encoder. This is particularly useful in the case where the audio source resides on multiple tapes, and it is desired to create a single output file which is transparent to the source tape change-over(s).

If an EBU/SMPTE Longitudinal Time Code (LTC) signal is applied to the encoder, the reader card will forward time code values to the control program running on the host CPU, which will in turn forward these to the DSP network. In this case the current time code will be displayed on the computer screen. The user may enter start and/or stop times via the control program; the desired hour, minute, second, and frame are specified by the user. The action that the encoder will take if either of the user-entered start or stop times matches the current time depends on the encoder state, which is also displayed on the control screen. If the encoder state is "encoding", and the stop time matches the current time, the encoder will stop after clearing out the processing pipeline. After this happens, the encoder state will display "stopped", and IEC 958 output will be filled with all zeros (no AC-3 or time code packets). While in the stopped state, if the start time matches the current time, the encoder will start coding the incoming audio data. Note that the user may manually start or stop the encoder at any time using the "enter" key.

There is an additional parameter appended to the displayed start and stop times; it is delineated by a colon following the frame number. This number represents PCM audio samples, and is used to specify start and stop times to a finer resolution than would be possible otherwise. The need for this parameter arises from the fact that LTC time code values represent video frames, and video frames do not exactly match the boundaries of AC-3 frames. Therefore, even though the starting time for video and audio frames may be time-aligned, the stopping points in general will not line up. To account for this, the sample number is added to the start/stop times as additional information.

The sample numbers may be used to generate sequential bit streams which are contiguous with each other, even though they have been encoded at different times. An example follows: The user may choose to start encoding segment "A" exactly on a video frame (time code) boundary. In this case, the sample number for the start time should be set to zero when the user enters the start time, and the first sample to be encoded will be the one present at the input at the instant when the start time was received by the time code reader card. The desired stop time should also be entered, and when the encoder stops encoding segment A, it will return a sample number value to the displayed stop time. This is the amount of time, given in PCM samples, that the encoder "ran over" from the user-entered stop time for segment A. If there is a continuing segment B to the source material, the encoder can be made to start encoding at the exact ending point of A by typing "r", (resume from stop time), on the control screen. This simply copies the stop time directly to the start time, including the sample number (this may also be entered manually). The user may then enter a new stop time for segment B. The user may not specify a stop sample number, as this must be determined by the encoder. Next, when the start time code is received for segment B, the encoder will delay encoding by the same number of samples that segment A ran over. This will result in an output stream for segment B that is effectively a continuation of the previous segment A.

This method, if combined with external equipment which can recognize the IEC 958 time code and AC-3 data packets, can be used to generate a contiguous data file for a complete audio program residing on separate source media.

NOTE At this time, non-drop frame (30 frames/sec), drop-frame (29.97 frames/sec), PAL (25 frames/sec), and film rate (24 frames/sec) time code formats are supported by the AC-3 encoder.

6.3 Command-Line Control

The Dolby AC-3 control program (ac3ctrl) may be invoked using command-line arguments that will cause the encoder to run for an interval specified by time code start and stop times, and then exit automatically, returning control to the operating system (DOS prompt). This allows the user to create a batch file that includes sequential commands for encoding several audio segments automatically.

Normally, the control program is invoked during the power-up sequence (from the autoexec.bat batch file) without additional command-line arguments, i.e., by calling "ac3ctrl". In this case, the control program assumes manual use, and will continue running regardless of the encoder state. Whenever any of the valid command-line arguments are specified, the control program operates in the automatic mode. In this mode, the control program will automatically exit (quit) whenever the encoder transitions from the "encoding" state to the "stopped" state.

6.3.1 Configuration Files

Running the control program with command-line arguments makes use of configuration files, which are binary (non-readable/editable) files which may be used to set the mode of operation and parameters for the DP561 encoder. The configuration file specifies all encoding parameters including Audio Service Configuration, Input/Output Control, Bit Stream Information, and also time code start/stop times and the encoder state.

The user may create custom configuration files which configure the encoder for a desired mode. When one of these files is used as a command-line argument to the control program, the encoder parameters and state will be set according to data in the file at the time when ac3ctrl is invoked.

A configuration file may be created by manually setting all parameters to the desired values, and then saving these to a user-defined file name. The menu item,

S - Save Settings

will prompt the user for a file name, which can be any valid DOS name, i.e., up to eight characters for the base name, and up to three characters for the extension.

The user may recall any saved settings using the menu item,

R - Recall Settings

This command will prompt the user for an existing file name; if the configuration file does not exist, then the encoder settings will not change.

The control program will automatically save a state configuration file whenever the program exits, i.e., whenever the user chooses

Q - Quit,

or whenever the program exits in the automatic mode. The name of this file is "ac3state.cfg." Each time the control program is invoked without the "-f" command-line argument (this parameter is described in the section immediately below), it will search for a file by this name in the current directory and restore the encoder state using this file.

Note The default state configuration file, ac3state.cfg, will only be created or updated when the user quits the control program by typing "Q ." Therefore, in order to save the state of the encoder between power cycles, the user should quit ac3ctrl before shutting the power off.

If the state configuration file does not exist, the control program will set the encoder into the default state as defined in Section 6.1 above.

Finally, as described in Section 6.1.6, the current DP561 software uses a different file format for its configuration files than was used by the original software release. If the new DP561 software finds itself about to overwrite an old-style configuration file, you will be asked whether or not you want to proceed. The new software can read old-style configuration files, but will only write them using the new format. The new file format includes several new parameters that are important when running the DP561 in remote control mode.

6.3.2 Command-Line Usage

The command-line usage for the control program, "ac3ctrl," is given as follows:

```
ac3ctrl [-c] [-f<file name>] [-r] [-x<start time>] [-y<stop time>]
```

-c = Set default control to remote control
This parameter causes the ac3ctrl program to start up in remote control mode. See Section 6.4 for details.

-f = DP561 Configuration File
This parameter identifies a configuration file for the encoder.

Note Do not put a space between the "f" and the file name.

-r = Resume from State File Stop Time
This switch causes the encoder to load the time code start time with the value of the stop time, including sample number, that is stored in the default AC-3 state configuration file. See below for details.

-x = Time Code Start Time
This parameter is used to specify the encoder start time code value.
The format for the start time is HH:MM:SS:FF:SAMPLE, where:

HH is hours ranging from 00 to 23.
MM is minutes ranging from 00 to 59.
SS is seconds ranging from 00 to 59.
FF is video frames ranging from 00 to 29.
SAMPLE is audio sample number ranging from 0 to 1535.

Notes Do not put a space between the "x" and the time code value.

Always include two digits for the HH, MM, SS, FF fields, e.g., "0" should be specified as "00," and "1" should be "01."

The SAMPLE field may range from one to four digits (with a value of 0 to 1535).

-y = Time Code Stop Time
This parameter is used to specify the encoder stop time code value.
The format for the stop time is HH:MM:SS:FF, where:

HH is hours ranging from 00 to 23.
MM is minutes ranging from 00 to 59.
SS is seconds ranging from 00 to 59.
FF is video frames ranging from 00 to 29.

Notes Do not put a space between the "y" and the time code value.

Always include two digits for the HH, MM, SS, FF fields, e.g., "0" should be specified as "00", and "1" should be "01."

All command-line arguments are optional. If a configuration file is not explicitly specified, the default name, "ac3state.cfg" will be used. If time code start and stop times are given as command-line arguments, these will be used instead of the values contained in the configuration file. If the "-r" argument is used, then any command-line start time specified with "-x" will be ignored.

If the user wishes to encode several segments of a single program, then the same configuration file may be used each time ac3ctrl is invoked. In this case, only the command-line time code start and stop times need to change for each encoded segment.

Furthermore, if a segment should begin at the exact stop time of the previous segment, then the "-r", (resume) switch may be used. This will load the previous stop time, including the sample number, into the current start time. The stop time is recalled from the default Dolby AC-3 state configuration file, which is updated each time the program exits. This method assumes that the resumed audio segment is to be encoded immediately following the prior audio segment.

If the user wishes to resume encoding from a particular segment at some later time, then the default state configuration file should be copied and saved under some other name. Later, this saved file may be renamed to the default state name, "ac3state.cfg", and the encoder will be able to resume from the previous stop time.

Example 1

Consider an audio program which is to be encoded into 4 segments: A, B, C, and D. The time code start and stop times for each segment are given as:

Segment	Time Code Start	Time Code Stop
A	00:00:01:00	00:14:24:00
B	00:14:24:00	00:27:45:12
C	00:27:45:12	01:05:00:22
D	01:05:00:22	01:30:38:00

Assume that a custom configuration file named "mystate.cfg" has been created. In addition to the data rate, number of channels, and other items that may be set, the user should also make sure that the encoder is in the "stopped" state before saving the settings into the custom configuration file. This is because each time ac3ctrl is invoked from the command line, the encoder should be stopped and waiting for the time code start point.

The segments may be encoded automatically by creating the following batch file:

```
ac3ctrl -fmystate.cfg -x00:00:01:00:0 -y00:14:24:00
ac3ctrl -r -y00:27:45:12
ac3ctrl -r -y01:05:00:22
ac3ctrl -r -y01:30:38:00
```

This procedure assumes that the operator provides either manual or automated control of the audio source machine, so that each encoded segment has sufficient time code pre-roll.

Example 2

Consider the case where segments A and B are to be encoded during one session, and then C and D during some later time. Two batch files could be created as follows:

```
ac3ctrl -fmystate.cfg -x00:00:01:00:0 -y00:14:24:00
ac3ctrl -r -y00:27:45:12
copy ac3state.cfg mystate.cfg

copy mystate.cfg ac3state.cfg
ac3ctrl -r -y01:05:00:22
ac3ctrl -r -y01:30:38:00
```

6.4 Remote Control Mode

The DP561 control program has two mutually exclusive modes of operation: local control mode and remote control mode. In local control mode, the user changes encoder parameters using the keyboard directly attached to the Model DP561. Encoder status and the currently active values of the encoding parameters are displayed on the DP561 monitor.

In remote control mode, encoder parameters are changed via an external Windows PC running the Model DP561 Remote Control software, which sends commands to the DP561 over an RS-232 serial link. In this mode, encoder status and parameters continue to be displayed on the DP561 monitor, but cannot be changed using the DP561 keyboard. The remote control software is described in detail in Chapter 8.

The DP561 starts up in the local control mode by default. To switch to the remote control mode, simply choose

C – Remote Control

from the DP561 local user interface. Once the remote control mode has been selected, choose

C – Local Control

to switch back to local control mode. The main screen of the DP561 operating in remote control mode is shown in Figure 6.2.

Dolby AC-3 Encoder: Remote Control

Current Time: 00:00:00:00 30 fps	**Encoder Status**
Start Time: 00:00:00:00:0	
Stop Time: 00:00:00:00:0	Dolby AC-3 Encoder Version: 6.3.0
	Encoder State: encoding
	Sampling Frequency: 48.0 kHz
Audio Service Configuration	Current Audio Bandwidth: 20.3 kHz
	Digital Input Status: no errors
Data Rate: 448 kbps	
Bit Stream Mode: complete main	**Bit Stream Information**
Audio Coding Mode: 3/2	
Low Frequency Effects Channel: on	Center Downmix Level: 0.707
	Surround Downmix Level: 0.707
Input/Output Control	Dolby Surround Mode: none
	Dialog Normalization: -27 dB
Input Channel Assignment: mode 1	Language Code: none
Output Format: pro, 32bit + TC	Audio Production Info: no
DC Filter (all channels): on	Mixing Level: none
LFE Lowpass Filter (120 Hz): on	Room Type: none
Surround Input 3 dB Atten.: off	Copyright Bit: copyright protected
Surround Channel phase-shift: on	Original Bit Stream: original
o - Digital Input/Output Settings	Compression: Film, Standard
Q - Quit C - Local Control	

Figure 6.2 AC-3 Control Program Main Screen (remote control mode)

Note In the remote control mode, the only commands that may be entered via the DP561 keyboard are "C" to switch to local control mode, "o" to see (but not modify) the Digital Input/Output Settings, or "Q" to quit.

6.4.1 Remote Parameter State Control

In the remote control mode, all parameters are controlled by the Model DP561 Remote Control software running on an external PC. Any encoding parameters set while in local control mode are discarded in favor of the settings provided by the remote control software. If the DP561 is then switched back to local control mode, the encoder parameters will retain the values set by the remote control software (until they are modified via the local user interface).

Like the local control program on the DP561, the remote control program on the external PC maintains a state file on its hard disk that remembers the values of all encoding parameters between sessions. The remote control software also allows the user to save and recall custom configuration files. It is important to note that the format of the remote control software state file is different than that of the local user interface; however a utility has been provided to translate local configuration files from the DP561 into the format required by the remote control software on the external PC. For more information about remote configuration files and this translation utility, see Section 8.3.

6.4.2 Changing the Default Control Mode

If desired, the DP561 can be configured to automatically boot up in remote control mode. To make this change, edit the autoexec.bat file in the root directory of the Model DP561's hard disk. Change the last line of this file from "@AC3CTRL" to "@AC3CTRL -c". This will cause the DP561 to automatically enter remote control mode on power-up.

Note If you plan to control the DP561 exclusively via the remote control software, you should make the change described above. Furthermore, because the remote control software is in control of all encoding parameters, the local power-up configuration file "ac3state.cfg" described in Section 6.1.6 is essentially ignored in remote control mode. Thus, it is not necessary to quit the ac3ctrl program before powering down the DP561. In fact, it is not even necessary to connect a keyboard or monitor to the DP561 at all!

7.1 Software Updates

The architecture employed in the DP561 allows for convenient software updates as the Dolby AC-3 encoder algorithm is improved from time to time. When significant changes are made to the encoding process, Dolby Laboratories will supply users with a self-installing update diskette.

The software is updated by placing the 3.5" update diskette in the floppy drive, and re-booting the unit. A power-on, reset, or warm boot (ctrl-alt-del) will work. The autoexec.bat file on the floppy diskette will copy the new C4x executable code to the DP561 hard drive. The machine will beep three times to indicate completion of the update.

Remove the diskette and reboot the computer; this will cause the new code to be loaded, and will restart the DP561 control program.

8.1 Software Installation

To install the remote control software on an external Windows PC, simply run the "setup.exe" program from the Model DP561 Remote Control installation disk. This will install all necessary files to the hard disk of the Windows PC, and create new items in the Start Menu under the heading "**Dolby Laboratories**".

The installation process also copies a "readme.txt" file to your hard disk. For the most current information, please consult this file before using the remote control software.

8.2 Using the Remote Control Program

The Model DP561 Remote Control program is a Windows application that allows complete control over the DP561 from an external PC via a serial port link. The remote control program provides access to all encoding parameters as well as immediate display of any changes in encoder status. The main screen for the remote control program is shown in Figure 8.1.

Note Once started, the remote control program tries to stay in constant communication with the DP561. For this reason, it is important that you not disconnect the serial link or shut down the Model DP561 while the remote control program is running - doing so will result in an error message on the Windows PC. For more information about remote control error messages, see Section 8.4.

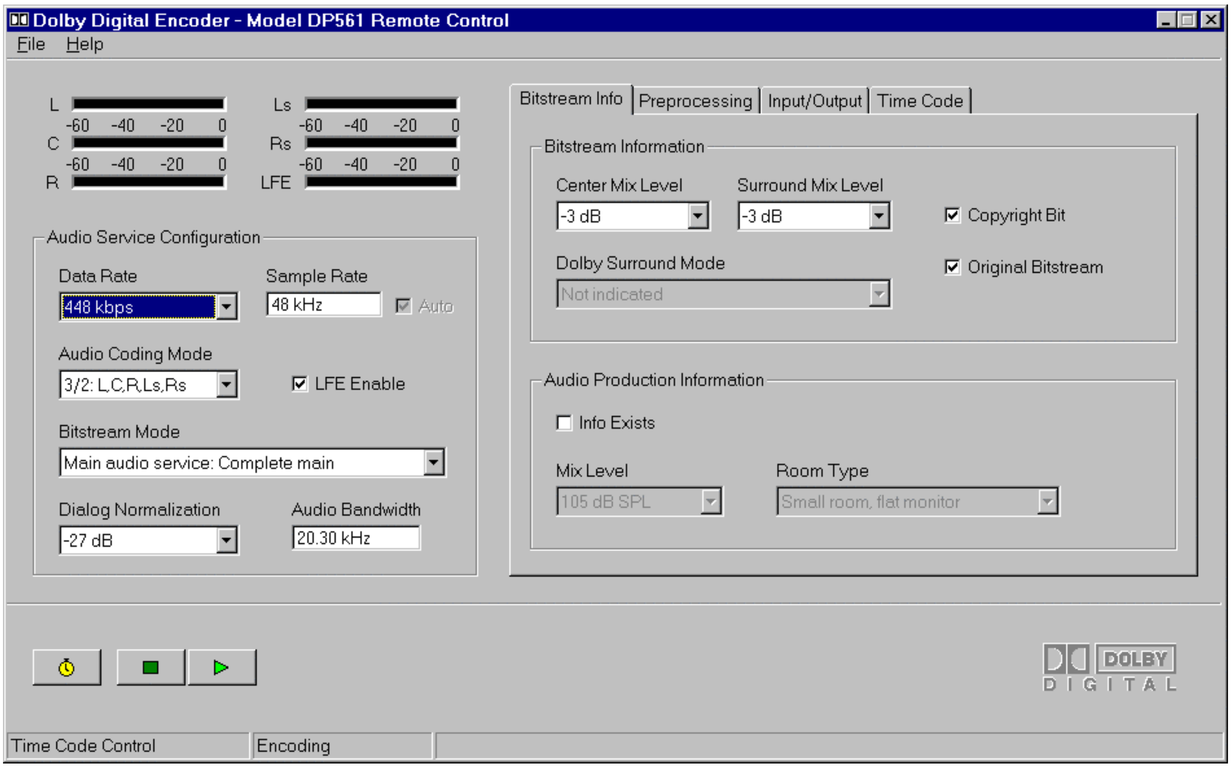


Figure 8.1 Model DP561 Remote Control Program Main Screen

The screen is divided into several distinct elements, including the audio level meters, the Audio Service Configuration fields, the parameter pages, the state control buttons, the status bar along the bottom, and the menu bar along the top. Each of these elements is discussed in the following sections. More information on the individual parameters and their usage can be found in Chapter 6.

8.2.1 Audio Level Meters

In the top left corner of the main screen are six level meters, one for each coded channel. The meters represent the peak value of each channel after all input filter preprocessing, and thus display the signal level that is actually being encoded. Also, the meter channels reflect the channel assignments, which may be different than the channel ordering on the DP561's AES/EBU inputs. That is, the meter labelled "L" corresponds to the left coded channel, regardless of which AES/EBU input the left channel happens to be assigned to.

If the audio coding mode is set to a mode other than 3/2, not all channels will be coded in the bitstream. In this case, the unused meters become inactive and are grayed out. Similarly, the LFE channel meter is grayed out if the LFE channel is not enabled. Finally, for audio coding modes with only one surround channel (i.e., 2/1 or 3/1), the mono surround level is shown on the meter labelled "Ls".

8.2.2 Audio Service Configuration

Just below the audio level meters is the Audio Service Configuration section, which contains fields for all of the most important encoding parameters. These include:

- Data Rate
- Sample Rate
- Audio Coding Mode
- LFE Enable
- Bitstream Mode
- Dialog Normalization
- Audio Bandwidth

8.2.3 Parameter Pages

To the right of the audio level meters and Audio Service Configuration section are the remaining parameter pages. Each page has its own tab along the top of the group – simply click on the desired tab to call up the appropriate page.

Normally, there are four tab pages available to choose from: the Bitstream Info page, the Preprocessing page, the Input/Output page, and the Time Code page. If the audio coding mode is set to 1+1, then a page for Dual Mono is also available.

Bitstream Info Page

This page, shown in Figure 8.2, contains fields for parameters that provide additional information about the coded bitstream, including:

- Center Mix Level
- Surround Mix Level
- Dolby Surround Mode
- Copyright Bit
- Original Bitstream
- Audio Production Info Exists
- Mix Level
- Room Type

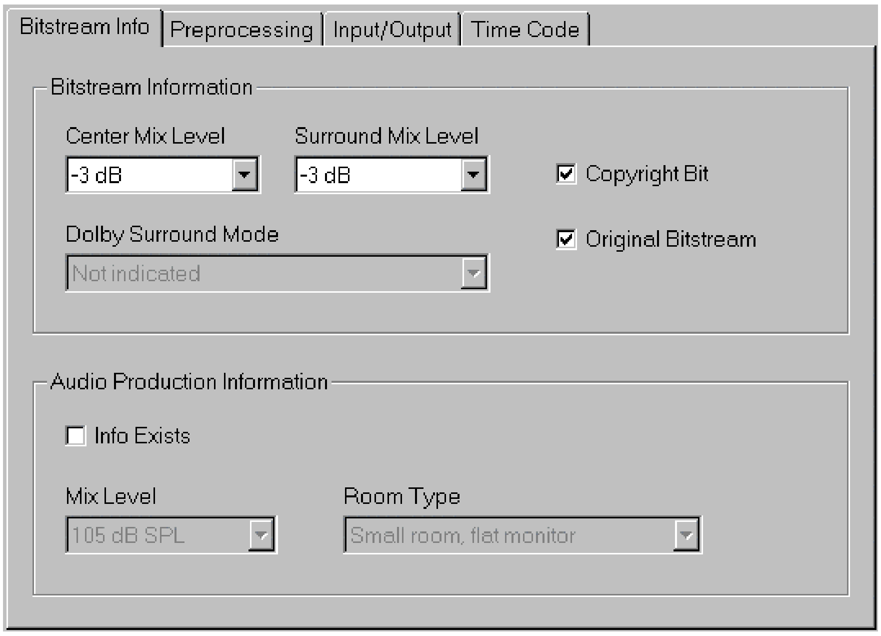


Figure 8.2 The Bitstream Info Page

Preprocessing Page

This page, shown in Figure 8.3, contains fields for parameters that control the input filtering and surround channel processing that can be applied prior to encoding, as well as the built-in dynamic range control processing. These fields include:

- DC Highpass Filter
- LFE Lowpass Filter
- Surround Channel 90 Degree Phase Shift
- Surround Channel 3 dB Attenuation
- Dynamic Range Compression Characteristic
- Compression Input Level Meter
- Compression Line Mode Meter
- Compression RF Mode Meter

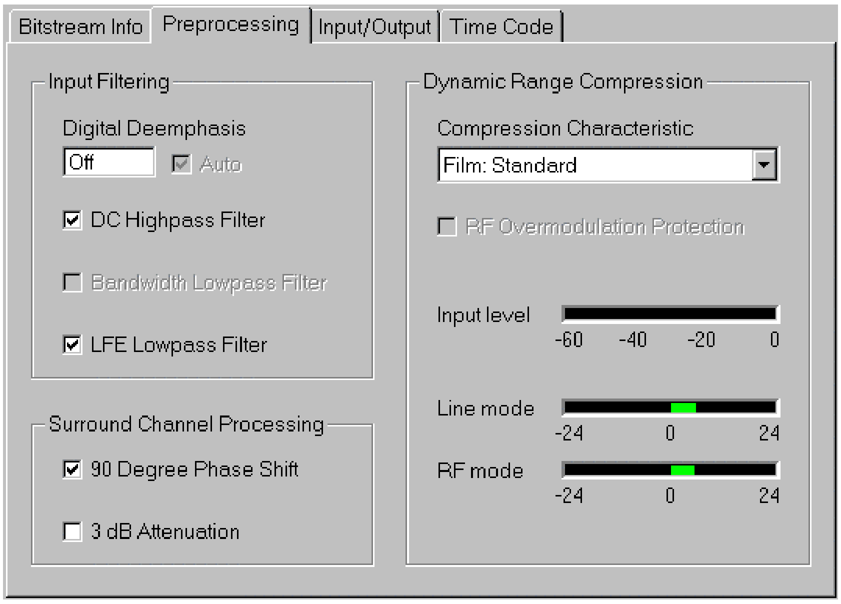


Figure 8.3 The Preprocessing Page

Note The Digital Deemphasis, Bandwidth Lowpass Filter, and RF Overmodulation Protection features are not available on the Model DP561. These controls have been disabled, and the Digital Deemphasis status will always indicate "Off".

The compression input level meter displays the "overall" level of all coded channels, which is used by the dynamic range compressor to determine the appropriate amount of boost or cut. This overall level is determined from the individual channel levels, and then modified based on the setting of the dialog normalization parameter. Dialog normalization values closer to -1 dB will cause more attenuation at the decoder, thus the compression input level is lowered accordingly.

The compression line mode and RF mode meters show the amount of boost or cut that will be applied by a decoder in Line Out mode or RF mode, respectively. These meters provide a useful means to see the extent of compression applied by each of the available compression characteristics. They can also be used to determine the correct dialog normalization setting. To do this, first select the desired compression characteristic for your project (other than "None"). Then feed the encoder with audio material that represents average loudness, and adjust dialog normalization up or down until the line mode and RF mode meters show no boost or cut.

Dual Mono Page

This page, shown in Figure 8.4, contains fields that apply to Ch2 when the audio coding mode is set to dual mono (1+1). If the audio coding mode is not set to dual mono, then this page is not available for selection. The dual mono fields include:

- Dialog Normalization

- Audio Production Info Exists
- Mix Level
- Room Type
- Compression Input Level Meter
- Compression Line Mode Meter
- Compression RF Mode Meter

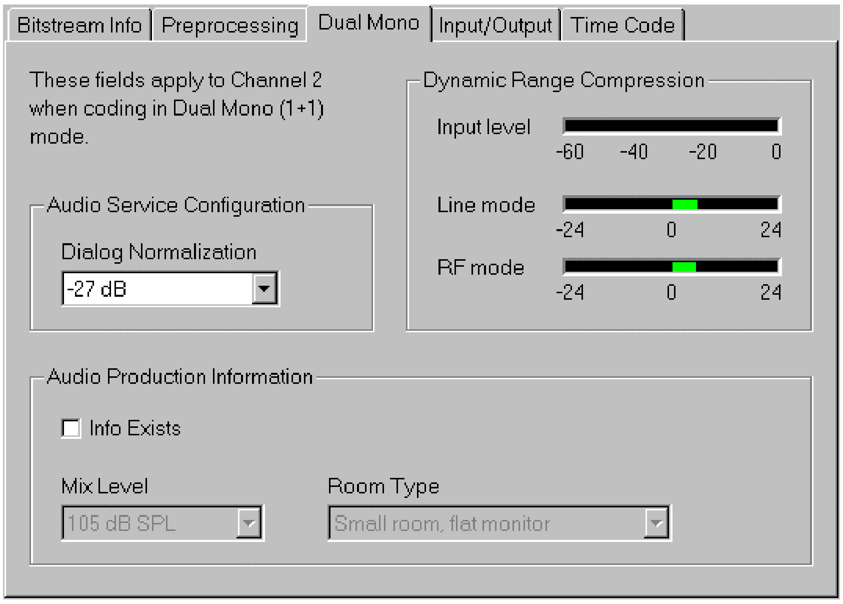


Figure 8.4 The Dual Mono Page

Note In dual mono mode, the two channels have independent dynamic range compression behavior, including independent dialog normalization settings, and independent compression input level and line mode/RF mode meters. However, they both make use of the same dynamic range compression characteristic, as selected from the Preprocessing page.

Input/Output Page

This page, shown in Figure 8.5, contains several fields that display the status of the incoming digital audio input signal, as well as control the digital bitstream output format. These fields include:

- Input Lock Status
- Input Sample Rate (measured)
- Input Integrity
- Input Subcode Format
- Input Subcode Sample Rate (from channel status bits)
- Input Subcode Emphasis
- Input Subcode Audio Bit
- Output Format
- Output Time Code Packets
- Output Audio Bit

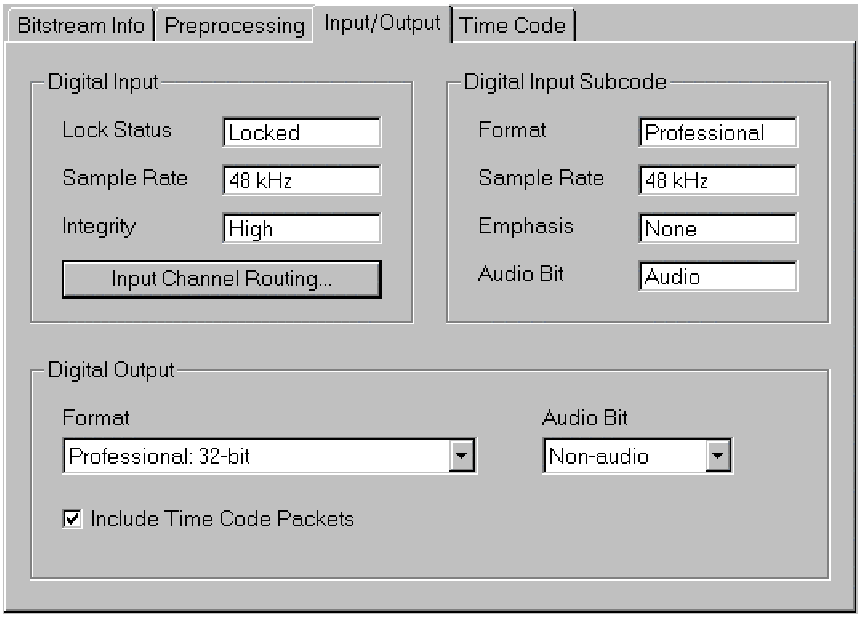


Figure 8.5 The Input/Output Page

Note The measured input sample rate should match the input subcode sample rate in a properly-configured system. If they do not match, the fault is generally in the device that is feeding the DP561. Note that the DP561 uses the measured sample rate to determine the sample rate used for the encoding process – which is displayed in the Audio Service Configuration section.

Also, since the DP561 does not support Digital Deemphasis, you should not use the DP561 to encode material which has been preemphasized (i.e., make sure that the input subcode emphasis indicator reads "None" before encoding).

The "Input Channel Routing…" button brings up a separate window, shown in Figure 8.6, in which you can specify the input channel assignments. There are six channel assignment presets for commonly-used formats, as well as an option for custom channel assignment, in which arbitrary pairings of coded channels to AES/EBU inputs are possible.

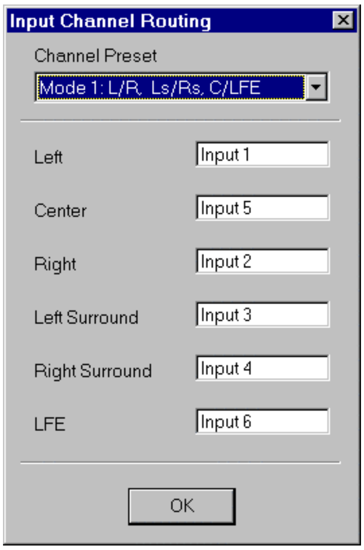


Figure 8.6 The Input Channel Routing Window

Time Code Page

This page, shown in Figure 8.7, contains fields that display the time code parameters, including:

- Time Code Input Status
- Current Time Code
- Current Time Code Format
- Start Time Code
- Stop Time Code

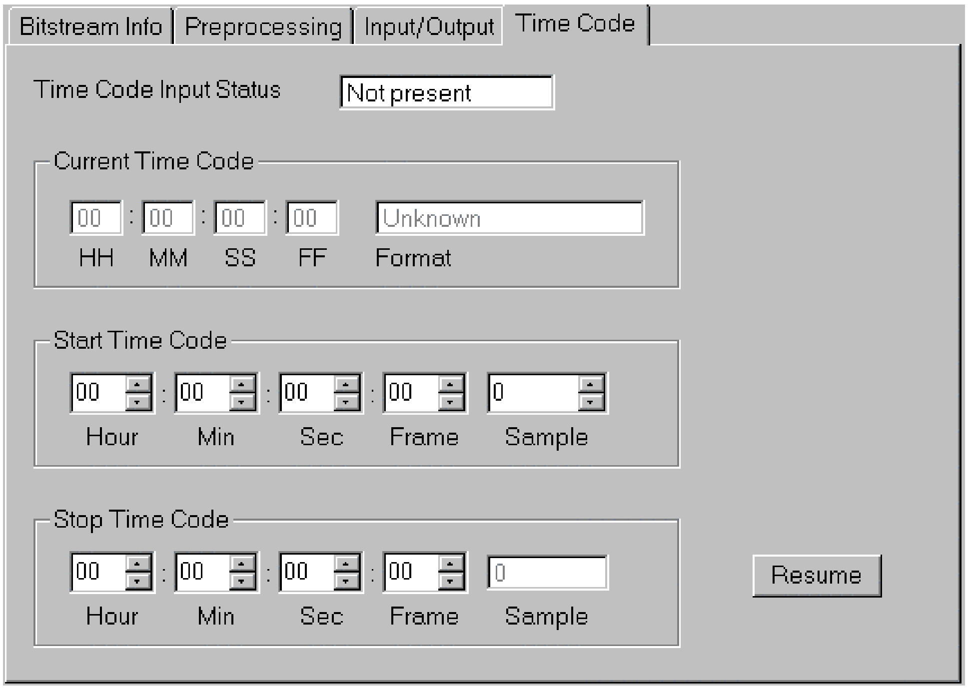


Figure 8.7 The Time Code Page

Note If the time code input is not present, the time code status fields will be shaded light gray, as shown in Figure 8.7.

The time code page also provides a "Resume" button, which copies the stop time code (including stop sample) into the start time code fields in one step.

8.2.4 State Control

The DP561 has two primary operating states: "encoding" and "stopped". Two other states, "prerolling" and "clearing", occur during transitions between the encoding and stopped states. The DP561 state may change based on start/stop time codes, or by manual command from the user.

State Control Buttons

The state control buttons at the bottom left corner of the main screen allow manual control of the DP561 state. There are two separate buttons for manual stop (square) and manual start (triangle) functions. Furthermore, these buttons contain "LED"-style lights that indicate the current state of the DP561. If the start button is lit, then the DP561 is in the "encoding" state.

A third button indicating time code control (stopwatch) is always lit. This serves as a reminder that the DP561 is always "on-line" with respect to time code – that is, it can initiate state changes based on time code regardless of manual start/stop commands.

The Status Bar

The status bar, which runs along the bottom of the main screen, provides text information on the current state of the DP561. Because the DP561 is always "on-line", the first segment of the status bar will always read "Time Code Control". The second segment indicates the current operating state, which will usually be either "Encoding" or "Stopped".

8.2.5 Menu Commands

The Model DP561 Remote Control software provides two drop-down menus at the top left corner of the main screen: the File menu and the Help menu. These menus contain several useful commands, some of which can also be accessed using simple keystrokes.

File Menu Commands

The file menu contains commands relevant to high-level encoder control, configuration file access, and quitting the remote control program. The specific commands in this menu, along with their keyboard shortcuts (if applicable) include:

- Select Encoder
  This menu item is currently disabled. In the future, it may allow a user to control more than one DP561 from a single external PC.

- Reset Encoder
  This menu item allows you to reset all DP561 parameters to their default state with one command.

- Load Config… (Ctrl-L from the keyboard)
  This menu item loads all DP561 parameters from a configuration file stored on the external PC hard disk.

- Save Config… (Ctrl-S from the keyboard)
  This menu item saves the current DP561 parameters to a configuration file on the external PC hard disk.

- Exit (Alt-X from the keyboard)
  This menu item quits the remote control program. The current DP561 parameters will be automatically saved to a configuration file, which will be loaded the next time you start the remote control program.

Help Menu Commands

The help menu contains commands provide additional help or information about the DP561 and the remote control program. The specific commands in this menu, along with their keyboard shortcuts (if applicable) include:

- Help (F1 from the keyboard)
  This menu item brings up the online help for the remote control program. This feature has not yet been implemented.

- About
  This menu item brings up the About Box for the remote control program, as shown in Figure 8.8. The About Box shows the version of the remote control program in the text fields to the right of the Dolby icon. Also, to the left of the "OK" button, it shows the version of the encoder software running on the DSP cards in the DP561.

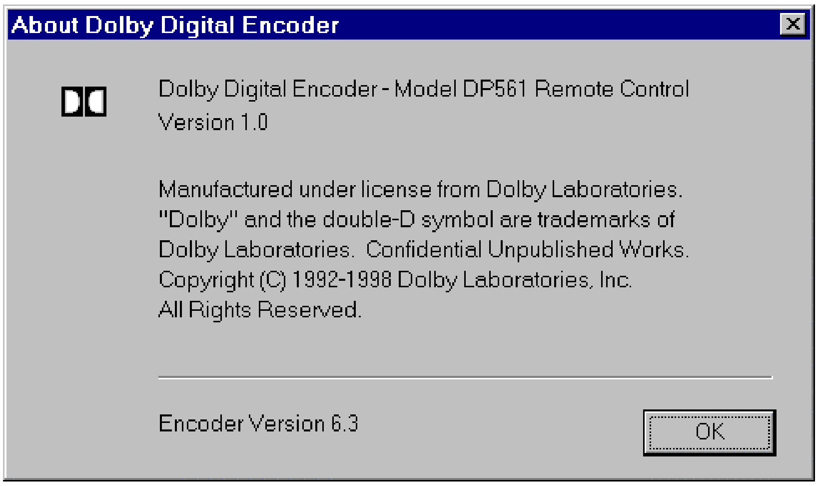


Figure 8.8 The About Box

8.3 Remote Control Configuration Files

The Model DP561 Remote Control software is able to save all DP561 encoding parameters to a configuration file on the external PC’s hard disk. In fact, whenever you quit the remote control program, it saves the current parameters to a file named "ac3state.cfg. (A second file, "ac3enc.ini", remembers where you placed the remote control program main screen on your desktop.)

In addition to using configuration files for remembering parameter settings between sessions, the remote control software also lets you save and recall your own configuration files. This feature makes it easy to set up snapshots that are suited to particular encoding situations, such as one for 2/0 stereo encoding, another for 5.1 channel encoding, and perhaps a third for Karaoke-mode encoding. The "Load Settings…" and "Save Settings…" menu items in the File menu allow you to specify the name and location of your custom configuration file.

8.3.1 Converting Local Mode Configuration Files

The configuration files used by the remote control program are not the same format as those used by the "ac3ctrl" program in the local control mode of the DP561. If you have created your own configuration files using the local mode user interface, you will need to convert them into the remote program format in order to use them with the remote control program.

This can be easily done using the configuration file conversion program, called "cfg_cvt.exe". This program is located in the "c:\code\ac3ctrl" directory of the Model DP561's hard disk. The command-line usage for this program is as follows:

```
cfg_cvt <DP561CFG> <WINDOWSCFG>
```

DP561CFG = DP561 configuration file
This is the name of the DP561 local mode configuration file, e.g. "local.cfg".

WINDOWSCFG = Remote control program configuration file
This is the name of the remote control program configuration file, e.g. "remote.cfg".

The "cfg_cvt" program is able to read old-style and new-style configuration files, as described in Section 6.3.1.

Example

Suppose you want to convert a DP561 local mode configuration file for use with the remote control program. The local mode configuration file is located in the "c:\code\ac3ctrl" directory of the DP561 hard disk, and is named "example1.cfg". The remote control configuration file is to be called "example2.cfg", and it must be copied to a floppy disk so that it may be transferred to the remote control PC's hard disk. You can do the conversion by putting a floppy disk into the DP561 floppy drive and entering the following command:

```
cfg_cvt c:\code\ac3ctrl\example1.cfg a:\example2.cfg
```

This assumes that you issued this command from the "c:\code\ac3ctrl" directory, which is where the "cfg_cvt" program is located.

8.4 Remote Control Error Messages

If the Model DP561 Remote Control program is unable to connect to the DP561, or if the connection is unreliable, you may see an error message on the external Windows PC. This section lists some of the error messages that you might encounter, along with possible causes of the underlying problem. If you get an error message not listed in this section, please contact Dolby Laboratories for additional support.

Note Any time an error message is displayed, you have the option of selecting "Retry" or "Cancel". "Retry" will attempt to reestablish the connection, while "Cancel" will quit the remote control program.

Unable to detect an appropriate COM port

This message occurs if the remote control program is unable to find the DP561 on either COM1 or COM2 when it first starts. Possible causes include:

1. The DP561 may not be connected properly to the external PC. See Section 5.6 for more information.

2. The DP561 may not be powered up and running in remote control mode. See Section 6.4 for more information.

3. The external PC may not be able to access the COM port which you have connected to the DP561. This may happen if there is a problem with your Windows installation, or if another program is already using the COM port.

Remote hardware not responding

This message occurs if the remote control program has successfully connected to the DP561 and the connection is subsequently lost. Possible causes include:

1. The DP561 may have been turned off, or the DP561 control program may have been quit or switched back into local control mode.

2. The serial link cable between the DP561 and the external PC may have come unplugged.

3. The external PC may not have reliable serial port hardware. The remote control program expects to communicate with the DP561 at 38400 baud without any transmission errors. This may be impossible with some older PCs, or with improperly designed serial port hardware. You might try using a different external PC to see if this is the problem.

Busy poll status timeout

This message occurs if the DP561 takes too long in responding to a message from the remote control program. This generally means that the DP561 microprocessor is not operating fast enough to keep up with all of the requirements of the DP561 system. If you encounter this message, check to see that no other programs (such as network drivers, TSR programs, etc.) are running in the background. Note that the DP561 is not designed to run other software in tandem with the local user interface, and no other programs are included as shipped from the factory.

Unexpected poll status
Message no longer available

These messages generally indicate that the remote control program has lost synchronization with the DP561. This is usually due to an unreliable serial port connection, but can also occur if the serial link connection is lost and then recovered (e.g., if you select "Retry" after getting the "Remote hardware not responding" error message). In some cases, you may have to quit the remote control program (by selecting "Cancel") and then restart it to establish a reliable link.

In some applications, it is necessary to capture Dolby Digital bitstreams as hard disk files for subsequent mastering. This section outlines one method of recording the output bitstream of a Model DP561 to disk, using a Unix workstation and a IEC-958 SCSI capture device.

There is also another method of recording Dolby Digital bitstreams to disk, using a Windows-compatible sound card with digital audio input and output capabilities. This method uses the "Dolby Digital Recorder" program, available separately from Dolby Laboratories.

A.1 Interface Description

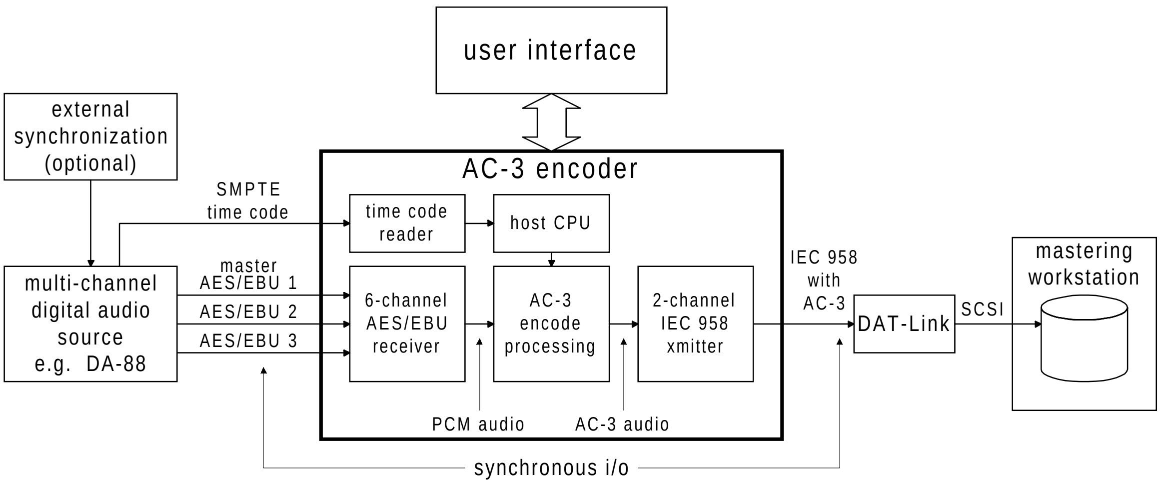


Figure A.1.1 AC-3 Encoder/Mastering Computer Interface

Figure A.1.1 depicts the entire encode mastering chain from the digital audio source to the mastering computer. The encoder is driven from an external digital audio source, which may also include time code.

The encoded output data stream is fed to an external unit, called a "DAT-Link+," manufactured by Townshend Computer Tools. This unit is in essence an IEC 958-to-SCSI converter. The DAT-Link+ is connected to the mastering computer via SCSI bus. A software driver may be invoked on the mastering computer which will write the packetized data to disc.

The following sections describe the use of the Model DP561 in conjunction with the DAT-Link+ and driver software to record (write) and playback (read) Dolby AC-3 disc files which may optionally include associated time code stamps.

A.2 Encoder Output Format

The IEC 958 output on the DP561 encoder has a considerably higher bandwidth than is required to carry a single Dolby AC-3 data stream, so each AC-3 sync frame is loaded into a packet which is "burst" out of the encoder at the frame rate. The output may also carry associated time code packets which are marked as such in the burst packet information field. The output format is described in detail in the included documentation entitled "Annex B, AC-3 Data Stream in IEC 958 Format." An example of the burst structure is given in Figure A.2.1 below.

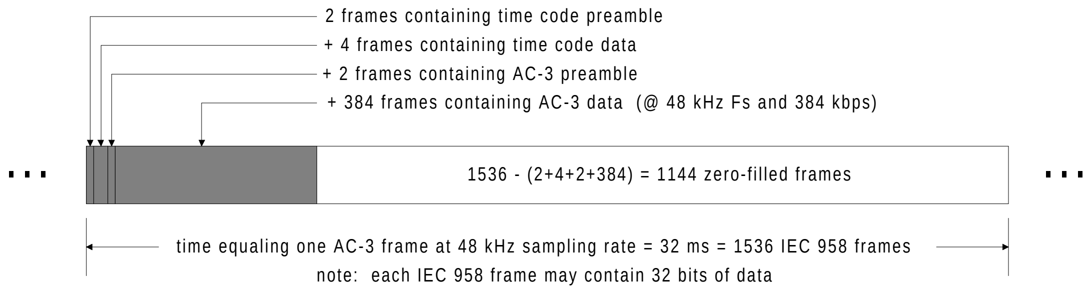


Figure A.2.1 AC-3 & Associated Time Code Data Burst

The figure depicts how the time code and Dolby AC-3 data are packed into the IEC 958 frames relative to the number of "zero" packed frames. Each time code packet is sent prior to the AC-3 frame to which it applies.

In the specific example shown above, the time code packets occur immediately prior to the AC-3 packets. The minimum time code packet length is 0x0060, (96 bits, or 3 IEC 958 frames), as defined in Annex B. The DP561 includes 32 bits of additional information in the time code packet which may be useful for disc mastering applications, and so the time code packet length is 0x0080, or 128 bits. At this time, the additional information includes a file type descriptor and length, and the Model DP561 encoder software version number.

The AC-3 data packet length in the above example is valid for a data rate of 384 kbps and a sampling frequency of 48 kHz. The packet length changes for other data rates and sampling frequencies.

A.3 Software and Dolby AC-3 File Format

The DAT-Link+ includes software which allows the user to record PCM audio data over the SCSI bus to a disc file. Playback capability is also provided. This software is intended to be used in a system which is recording PCM audio data only, and therefore records all received data from the IEC 958 input. However, Dolby Laboratories has modified the software so that it will recognize Dolby AC-3 and/or time code packets, like those provided from the DP561 output, and record only the packet payloads to the disc file.

The name of the record program is called "recac3." Dolby supplies the source and executable code for this program to the users of the DAT-Link+.

When the DAT-Link+ and the "recac3" program are used in conjunction with the Model DP561, the structure of the resultant Dolby AC-3 file written to the mastering computer's disc is described in the next manual section (A.4) entitled "AC-3 Disc File Formats." This system will create *file type 0x01,* as described in that section.

The usage for "recac3" is given as:

```
recac3 [-u server:unit] [-s a|t] <output_file>
```

-u = DAT-Link+ unit to use. (Default = local unit 0)
This parameter essentially defines the SCSI identification number for the DAT-Link+. The default setting for this parameter is "dolby:0", which is the name of Dolby Lab's server to which the DAT-Link+s are connected, followed by the unit number "0." This option must be specified for other systems if recac3 is not modified to set this parameter to a different default server name.

-s = Skip the specified packet type:
a = AC-3
t = time code
This option may be used to skip either Dolby AC-3 packets or time code packets (i.e., they will not be written to disc). By default, the recac3 program will write both packet types to disc in the order which they are received from the IEC 958 interface.

Dolby Labs also supplies a playback utility for the AC-3 disc files. The program is named "playac3", and will play a Dolby AC-3 file from disc out the DAT-Link+'s audio output. The program reads AC-3 disc files, and formats the data into the same type of IEC 958 output as described above for the DP561. In this case, the output stream will come from the DAT-Link+'s output connectors. This may be useful as a monitoring tool for the recorded AC-3 disc files.

The usage for "playac3" is given as:

```
playac3 [-u server:unit] [-l | -r] [-a] <input_file>
```

-u = DAT-Link+ unit to use. (Default = local unit 0)
This parameter essentially defines the SCSI identification number for the DAT-Link+. The default setting for this parameter is "dolby:0", which is the name of Dolby Lab's server to which the DAT-Link+s are connected, followed by the unit number "0." This option must be specified for other systems if playac3 is not modified to set this parameter to a different default server name.

-l = Single pack output data into left channel.

-r = Single pack output data into right channel.

Default = double pack into both channels.

Either the l *or* r option may be used to specify that the AC-3 and time code packet data occupy only one of the IEC 958 channels. Left and right correspond to A frame and B frame, respectively. By default, the playac3 program will format the data into both channels of the IEC 958 output stream.

-a = Clear channel status _audio_ bit.

This parameter may be used to clear the _audio_ bit of the IEC 958 channel status. The default value for this bit is "one", indicating non-audio data in the channel, as required by "Annex B:, AC-3 Data Stream in IEC 958 Format." However, some equipment may not accept a data stream with this bit set, so the user may clear it with this command-line option.

NOTE The Dolby-modified DAT-Link+ software, "recac3" and "playac3", have been written and tested under SUN OS4. SUN OS5 (Solaris 2.0) has not been tested at this time, although the DAT-Link+ release software is said to support SUN OS5.

A.4 AC-3 Disc File Formats

Introduction

This document contains specifications of four different AC-3 file formats which are currently in use, and partial specifications for some other formats. This specification considers the AC-3 files to be a sequence of *dataframes*. Each dataframe consists an AC-3 *syncframe* and may include a *header* containing a time stamp or other information. Each AC-3 syncframe is the encoded representation of 1536 audio samples.

The initial byte of an encoded AC-3 file may be used to determine the dataframe (or file format) type. Two of the file types are simply a sequence of AC-3 syncframes (dataframe equal to AC-3 syncframe). These two file types differ only in that one is byte-pair reversed. The other two file formats include headers containing SMPTE time-stamp information (dataframe equals header plus AC-3 syncframe).

Specification

The psuedo code below shows the AC-3 file to be a sequence of dataframes. The type of dataframes in the file may be determined by testing the value of the intial byte of data in the file.

Syntax

```
AC-3_file()
{
    if(initialbyte==0x0B)
    {
        while(!EOF)
        {
            dataframe_type_0x0B() ;
        }
    }
    else if(initialbyte==0x77)
    {
        while(!EOF)
        {
            dataframe_type_0x77() ;
        }
    }
    else if(initialbyte==0x00)
    {
        while(!EOF)
        {
            dataframe_type_0x00() ;
        }
    }
    else if(initialbyte==0x01)
    {
        while(!EOF)
        {
            dataframe_type_0x01() ;
        }
    }
} /* end of AC-3 file */
```

Dataframe_type_0x0B

The AC-3 syncframe is defined as a sequence of bits of data. The length of a syncframe is always an integer multiple of 16. The first 16 bits of the syncframe is the AC-3 syncword. This syncword has a value of ‘0000 1011 0111 0111’ (0x0B77), where the left (or most significant) bit is transmitted first. The AC-3 syncframe bit sequence may be converted into a byte sequence by forming bytes from each sequential set of 8-bits. The MSB of byte 0 is bit 0 in the AC-3 syncframe, the LSB of byte 0 is bit 7, etc.

An AC-3 file type 0x0B consists simply of the sequence of bytes which result from this conversion. There are no headers or time stamps. The dataframe is simply the byte sequence representation of the AC-3 syncframe.

AC-3 stream: byte 0, byte 1, byte 2, byte 3, byte 4, byte 5...

AC-3 file type 0x0B: byte 0, byte 1, byte 2, byte 3, byte 4, byte 5 ...

Syntax			
Field	**Bytes**	**Value**	**Comments**
dataframe_type_0x0B()			
{			
syncframe() ;			AC-3 syncframe
}			End of this dataframe

Dataframe_type_0x77

An AC-3 file type 0x77 is identical to the file type 0x0B, except that pairs of bytes are reversed. The first byte in this file is the second byte of the AC-3 byte stream. This file type is generated by some encoders running on Intel computers, and which form the AC-3 bit stream as a sequence of 16-bit words which are written to disc as 16-bit integers. (Some AC-3 development was done in this way.)

AC-3 stream: byte 0, byte 1, byte 2, byte 3, byte 4, byte 5...

AC-3 file type 0x77: byte 1, byte 0, byte 3, byte 2, byte 5, byte 4, ...

Syntax			
Field	**Bytes**	**Value**	**Comments**
dataframe_type_0x77()			
{			
syncframe() ;			AC-3 syncframe in byte-pair reversed order
}			End of this dataframe

Dataframe_type_0x00

The AC-3 file type 0x00 includes a 16-byte header at the beginning of each dataframe. The balance of the data frame is the AC-3 syncframe with sequential byte ordering.

Syntax			
Field	**Bytes**	**Value**	**Comments**
dataframe_type_0x00()			
{			
file_type_code ;	1	0x00	Indicates frame type 0x00
smpte_hours ;	1		BCD encoded, see note 1
marker_byte ;	1	0x00	
smpte_mins ;	1		BCD encoded, see note 1
marker_byte ;	1	0x00	
smpte_secs ;	1		BCD encoded, see note 1
marker_byte ;	1	0x00	
smpte_frames ;	1		BCD encoded, see note 1
smpte_ts_sample ;	2		See ATSC A/52, Annex B
smpte_ts_flags ;	2		See ATSC A/52, Annex B
frame_count ;	2		See note 2
version ;	2		See note 3.
syncframe() ;			AC-3 syncframe
}			End of this dataframe

Note 1: The values of hours, minutes, seconds, and frames are BCD encoded nibbles. The 4 high bits in the byte represent a 4-bit binary integer whose value should be multiplied by 10, and then added to the 4-bit integer in the 4 lower bits of the byte.

value = 10 * (8*bit7 + 4*bit6 + 2*bit5 + bit4) + 8*bit3 + 4*bit2 + 2*bit1 + bit0

Note 2: The frame_count is a 16-bit unsigned integer which has a value of zero in the first dataframe of the file, and which increments by one for each following dataframe. This number will “wrap around” to zero approximately every 35 minutes (the time it takes for 65536 frames at 48 kHz sample rate). The value of the 16-bit integer is formed from the two bytes as follows:

value = 256 * first_byte + second_byte

Note 3: The value of version represents the version of the AC-3 encoder code used to encode the AC-3 frames in this file. The value is interpreted as a two digit number, followed by a decimal point, and then another 2 digit number.

value = u v . x y

where u, v, x, and y are BCD encoded values; u is in the upper 4 bits of the first version byte, v is in the lower 4 bits of the first version byte, x is in the upper 4 bits of the second version byte, and y is in the lower 4 bits of the second version byte.

Dataframe_type_0x01

The AC-3 file type 0x01 includes a 16-byte header at the beginning of each dataframe. The balance of the data frame is the AC-3 syncframe with sequential byte ordering.

Syntax			
Field	**Bytes**	**Value**	**Comments**
dataframe_type_0x01()			
{			
file_type_code ;	1	0x01	Indicates frame type 0x01
hdr_len ;	1	0x10	Header length, 16 bytes
smpte_ts_word[0] ;	2		See Annex A
smpte_ts_word[1] ;	2		See Annex A
smpte_ts_word[2] ;	2		See Annex A
smpte_ts_word[3] ;	2		See Annex A
smpte_ts_word[4] ;	2		See Annex A
smpte_ts_word[5] ;	2		See Annex A
version ;	2		See note 3 above.
syncframe() ;			AC-3 syncframe
}			End of this dataframe

Dataframe_type_0x02 thru dataframe_type_0x3F

A number of file_type_codes have been reserved, and partially specified. Software which is designed to work with file type 0x01 may easily be configured so that is will also work with future file types 0x02-0x3F. The key to providing this flexibility is to make provision for the potential existence of the new_byte[i] data bytes. The initial 16 bytes of the dataframes in these file formats will always be identical to the initial 16 bytes found in file type 0x01. The new_byte[i] data bytes may have differing meanings in these future file types. Software which understands file format 0x01 will understand the meaning of the initial 16 bytes of all of these dataframe types. Software should be designed to use the value of hdr_len to skip over any new_byte[i] data which occurs in any of these file types.

Syntax			
Field	**Bytes**	**Value**	**Comments**
dataframe_type_0xA()			File format A (0x01<A≤0x3F)
{			
file_type_code ;	1		Indicates frame type
hdr_len ;	1		Header length
smpte_ts_word[0] ;	2		See Annex A
smpte_ts_word[1] ;	2		See Annex A
smpte_ts_word[2] ;	2		See Annex A
smpte_ts_word[3] ;	2		See Annex A
smpte_ts_word[4] ;	2		See Annex A
smpte_ts_word[5] ;	2		See Annex A
version ;	2		See note 3 above.

for(i=0;i<n;i++)
{
new_byte[i] ;
}
	n		n = hdr_len - 16
			
			additional bytes of information

syncframe() ;			AC-3 syncframe
}			End of this dataframe

A.5 The SMPTE Time Stamp

An excerpt from Annex B of ATSC Document A/52 (“Digital Audio Compression Standard (AC-3)”) is reproduced below. The information in Section 4.8.2 (“Time Stamp Payload”) shows the contents of the smpte_ts_word[n] (where n = 0 to 5) which are included in some of the AC-3 file formats.

The Time Stamp Data_type

Time stamps are useful in applications where time information must be kept closely associated with encoded audio data (see Figure 1). An example of this would be in a digital audio/video transmission system where both audio and video sources have SMPTE time code. When the audio and video are digitally compressed it is useful if each output compressed bit stream contains the original SMPTE time code information. When a time stamp is included in this interface, its value applies to the single coded audio access unit which immediately follows.

Values of SMPTE time code occur only once per picture frame, and thus have a resolution in their value of approximately 33 ms (for 30 Hz frame rate). Audio samples occur much more frequently, approximately once every 21 μs (48 kHz sample rate). The AC-3 audio access units occur every 32 ms (48 kHz sample rate). It would be desirable for the time stamp to precisely indicate the time of the first audio sample contained in each audio access unit, but this is not practical due to the coarse nature of the source of the timing information (SMPTE time code). The method adopted here is to let the time stamp contain both a SMPTE time code value, and an indicator as to the audio sample within the following audio access unit to which the time code value applies. Since, in general, there is not an exact integer relationship between the frequency of time code values and the frequency of audio samples, there will always be an inherent ambiguity of exactly when in the audio stream the exact time code value is valid, since the exact point of validity will typically be between two audio samples. Depending on the precise time code frame rate, and the audio access unit frequency (which depends on the audio sample rate), it is possible for all audio samples within a single audio access unit to be between two sequential time code values. In this case, the time stamp cannot point to a sample in the audio access unit, but must point to a sample in the following audio access unit. It is also possible for two time code values to occur within a single audio access unit. In this case, the time code value which applies to the earliest sample in the access unit shall be used. It should be recognized that the time stamp values will inherently have a small amount of jitter. The sources of the jitter will be: the inherent + or - 1 sample ambiguity as to which sample the time code value applies; bandwidth limitations in some sources of linear time code; and interrupt latencies in some hardware implementations. In some applications (such as converting the time stamp values to values of MPEG-2 PTS), this jitter may have to be removed by subsequent processing equipment.

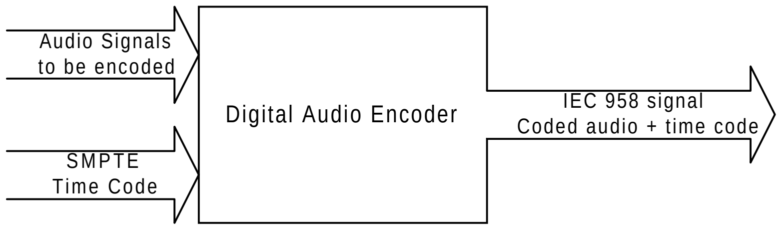


Figure 1. Encoding audio with time code.

Preamble values

Time stamps are conveyed by data bursts with a data_type value of 0x2. The value of data_type_dependent shall be set to 0x0 for the payload defined below. (In the future, other payload types may be defined for different values of data_type_dependent.) The length_code shall indicate the actual length of the time stamp payload.

Time stamp payload

The time stamp payload, shown in Table 8, has a minimum length of six 16-bit words which have a defined meaning. Additional 16-bit words may be optionally added, but the meaning of these words is not specified.

Table 8 Time Stamp Payload

Time Stamp Payload Word		MSB 15	14	13	12	11	10	9	Bit Number 8	7	6	5	4	3	2	1	LSB 0
0	Usr8, Usr7, **Hours**	[63]	[62]	[61]	[60]	[55]	[54]	[53]	[52]	[59]	[58]	H20	H10	H8	H4	H2	H1
1	Usr6, Usr5, **Minutes**	[47]	[46]	[45]	[44]	[39]	[38]	[37]	[36]	[43]	M40	M20	M10	M8	M4	M2	M1
2	Usr4, Usr3, **Seconds**	[31]	[30]	[29]	[28]	[23]	[22]	[21]	[20]	[27]	S40	S20	S10	S8	S4	S2	S1
3	Usr2, Usr1, cf, df, **Frames**	[15]	[14]	[13]	[12]	[7]	[6]	[5]	[4]	[11]	[10]	F20	F10	F8	F4	F2	F1
4	Sample Number	S15	S14	S13	S12	S11	S10	S9	S8	S7	S6	S5	S4	S3	S2	S1	S0
5	Reserved, Flags	R	R	R	R	R	R	R	R	R	R	a3	a2	a1	a0	f1	[10]

<u>Table Entries</u>

[63]	Bit number 63 of SMPTE time code word
H20	This bit has a value of 20 hours
R	Reserved bit, set to ‘0’
Usr8	The 8th group of user bits in the SMPTE time code word
cf	Color frame flag bit
df	Drop-frame flag bit
f1	Flag bit number 1
S15	Sample number, bit 15
a3	Frame rate code, bit 3

The first four words contain an hours, minutes, seconds, frame count. Space is available to carry the user group, color frame flag, drop frame flag, and unassigned bits from a SMPTE time code word. Flag bit f1 (in word 5) is set to a ‘1’ if this information has been copied from a source of SMPTE time code into the upper bits of payload words 0-3. If flag bit f1 is set to a ‘0’, this information has not been provided, and the upper bits of payload words marked [ ] are all set to ‘0’. The sample number in word 4 is an unsigned integer which indicates the sample number (Sn in Figure 2) to which the time code value applies. The sample number does not have to be exactly correct, but should indicate an audio sample within ±0.5 ms of the ideal value. Word 5 contains 10 reserved bits (in bits 6-15), a 4-bit frame rate code (a3-a0), the f1 flag bit, and the drop-frame flag bit (bit 10 of the SMPTE time code word) if the timing source is SMPTE time code. The drop-frame flag bit is always provided in bit 0 of word 5; its presence in bit 6 of word 3 is conditional on the value of the f1 flag bit. The meaning of the frame rate code is shown in Table 9.

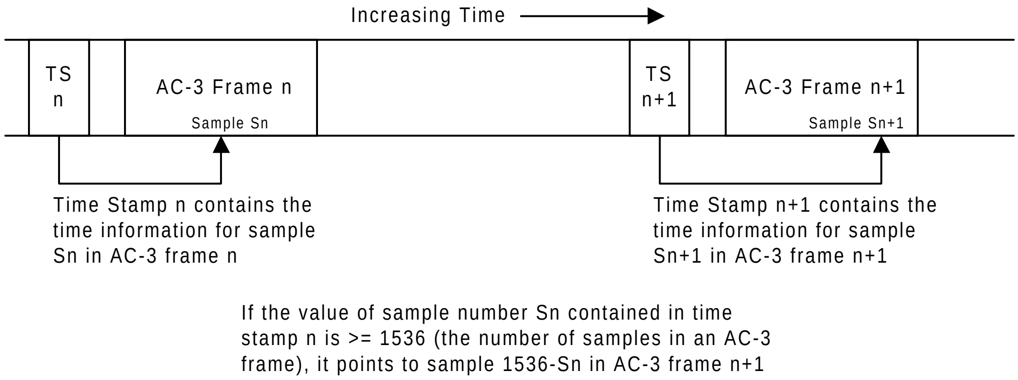


Figure 2. Time stamps and AC-3 frames in the IEC958 data stream.

Table 9 Frame Rate Code

Frame Rate Code				Frame Rate
a3	a2	a1	a0	
0	0	0	0	not indicated
0	0	0	1	24 ÷ 1001 (23.98)
0	0	1	0	24
0	0	1	1	25
0	1	0	0	30 ÷ 1001 (29.97)
0	1	0	1	30
0	1	1	0	50
0	1	1	1	60 ÷ 1001 (59.94)
1	0	0	0	60
-	-	-	-	reserved
1	1	1	1	reserved

Additional payload words containing arbitrary information may be optionally provided. The meaning of any additional payload information is not specified. Receivers should be capable of operating whether or not additional information is present. The presence of additional information may be determined by the value of the length_code in the burst preamble. If the value length_code is 0x0060 then no additional information is present. If the value of length_code is greater than 0x0060 then additional information is present.

A.6 Operation with a DAT-Link+

The following steps describe a procedure for creating a master disc file using a Tascam DA-88, a Model DP561, and DAT-Link+ connected to a mastering computer. In this example, two PCM master tapes (containing a movie sound track, for example), are used to create a single disc file. It is assumed that the DA-88 tapes are striped with time code which corresponds to the video elements of the movie, and that the two tapes have a small overlap region to allow for pre-roll on the second tape. It is also assumed that the DAT-Link+ distribution software and recac3 program have been installed and configured on the mastering computer.

The system elements in this example are connected together as shown in Figure A.1.1 above. After power-up, the DP561 will be in the "encoding" state, and the "locked" and "pause" LEDs on the DAT-Link+ should be lit.

The first DA-88 tape should be loaded and rewound to a few seconds before the desired start time. If the display on the DA-88 machine is in the "TC" mode, then the displayed numbers will match those on the DP561 monitor screen.

Next, stop the DP561 by pressing "enter" on the DP561 keyboard. In this mode, no time code or AC-3 packets will be output. Now the start and stop times may be entered on the DP561 for the first DA-88 tape.

Next, invoke the recac3 program by typing "recac3 *filename*," (filename may be any valid file name) on the mastering computer keyboard. The computer should display "recording..." on the screen, and the "record" LED on the DAT-Link+ should be lit. Only data packets will be written, and since the DP561 is stopped, the recac3 program is waiting for valid data packets, (i.e., nothing is being written to disc at this time).

Now the DA-88 machine may be started by pressing "play." When the start time is encountered, the DP561 will immediately start encoding, and subsequently start outputting time code and AC-3 data packets. These packets will be identified by recac3, and written to the specified disc file.

When the stop time is encountered by the DP561, it will stop encoding automatically, and no further data will be written by recac3. The DA-88 machine may now be stopped.

If there is only one DA-88 tape to encode, then the recac3 program may be stopped by typing "ctrl-c" on the mastering computer keyboard. If there is an additional segment to be encoded, then do NOT interrupt recac3, and load the second DA-88 tape.

Locate the second tape to a few seconds prior to the stop time of the first tape. Next, type "r" on the DP561 keyboard; this will copy the stop time as the new start time, including the sample number; then manually enter the new stop time for the second tape.

The second tape may now be started, and when the new start time is reached, recac3 will resume writing to disc. When the new stop time is reached and the DP561 has stopped encoding, stop the recac3 program by typing "ctrl-c." The program will display the number of time code and Dolby AC-3 packets written to disc. The resultant file will contain contiguous AC-3 data frames for the entire encoded program.

B.1 System Specifications[2]

Audio Coding Method

Dolby Digital (AC-3)[3].

Data Rates and Channel Modes

Total Data Rate (kbps)	Channel Modes[4]
56	1/0
64	1/0
80	1/0
96	1+1, 1/0, 2/0
112	1+1, 1/0, 2/0
128,160	1+1, 1/0, 2/0, 3/0, 2/1
192	1+1, 1/0, 2/0, 3/0, 2/1, 3/1, 2/2
224,256,320,384, 448,512,576,640	1+1, 1/0, 2/0, 3/0, 2/1, 3/1, 2/2, 3/2

Audio Sampling Rates

44.1 kHz and 48 kHz.

Frequency Response

Lower limit: < 20 Hz. Upper limit varies with data rate and channel mode. Tolerance 0.25 dB.

Distortion[5]

Less than 0.003 % at 1 kHz.
Less than 0.01 %, 20 Hz - algorithm upper limit.

Dynamic Range

Greater than 90 dB.

Time Delay

Approximately 400 ms. Varies with sample rate. Contact Dolby Laboratories for further information.

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[2]Specifications subject to change without notice.
[3]Patents pending and issued worldwide.
[4]Modes follow the convention X/Y, where X represents the number of front channels, and Y represents the number of rear channels. The mode 1+1 represents a dual mono mode. 2/0 represents either conventional stereo L/R or Dolby Surround encoded Lt/Rt.
[5]Measured with an Audio Precision® System One.

B.2 Hardware

Hardware Configuration

Rack mount
PC-compatible platform (CPU, disk controllers, keyboard, serial and parallel ports, RAM):

DP561B:	On motherboard
DP561:	On host CPU plug-in card

DSP card
Digital I/O card
Power supply
3.5" floppy disk drive
Hard disc drive
VGA display card (monitor not included)

Digital Audio Input

3 ea. AES/EBU inputs, 25-pin D-sub connector, 110 ohms ±5%. (adapter cable to 3 ea. XLR female connectors provided).

Encoded Data Output

BNC connector, 75 ohms, can be adapted to AES/EBU or SMPTE 276M.

Remote Serial Interface

RS-232, 38400 baud. The DP561 COM1 port is normally configured as a DTE device; using the null modem adapter switches this to a DCE device.

Windows® Compatibility

Utilities available for remote control and hard disk file storage with PCs and workstations operating under Windows 95 or Windows NT (4.0 or later)

B.3 General

Power Requirements

100-130 / 200/260 Vac Manual Select, 50 to 60 Hz, 160 Watts max.

Dimensions and Weight

7 x 19 x 20 inches (17.8 x 48.3 x 50.8 cm), 33 pounds (72.6 kg) net.

Shipping Dimensions and Weight

33.5 x 28 x 20 inches (85 x 71 x 51 cm), 49 pounds (108 kg) gross.

Regulatory Notices

USA: This unit complies with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules.
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.

Environmental Requirements

Temperature Operating: 5^{o}C to 45^{o}C, integral fan cooling.
20-85% Relative Humidity (non-condensing).

Warranty - 1-year limited, parts and labor.