Dolby®

Model 363
Two-Channel
SR/A-Type Processor
Users' Manual

MAIN

Users' Manual

For

Model 363

Two Channel SR/A-Type Processor

(Serial No. 650 Onwards)

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WARRANTY INFORMATION - USA: Warranty on the product covered by this manual is subject to the limitations and disclaimers set forth in the warranty disclaimer originally shipped with the product and also printed on the back of the invoice.

All requests for repairs or information should include the unit model number and serial number to assure rapid service.

ISSUE 8
W96/034
Dolby Part No. 91122

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SECTION 1
INTRODUCTION AND SPECIFICATIONS

1.1 Introduction

The Dolby Model 363 is a two-channel signal processing unit designed to accept modules providing either A-type noise reduction or SR Spectral Recording processing, or a combination of the two. The unit can operate with a mixture of module types, or with only a single module installed.

The unit occupies 1U (44 mm, 1.75") of standard 19" rack space. Provision is made for up to two plug-in modules for signal processing. The Cat. No. 450 module provides A-type noise reduction, the Cat. No. 350 SR processing, and the Cat. No. 300 contains switchable SR/A-type. One module is required for each channel in the Model 363. Circuitry in the frame allows selection and indication of the type of processing provided in the installed modules.

Units prior to serial number 650 differ from later units in that no cooling fan was fitted and the remote control for noise reduction allowed only processing in/out and not selection of type of processing.

The Model 363 is shipped with the following:

1 Model 363 mainframe

Signal processing modules as ordered
- Cat. No. 300 for switchable SR or A-type processing
- Cat. No. 350 for SR processing only
- Cat. No. 450 for A-type processing only

1 AC Power Cable

1 Cable mounting 9-pin D connector

Fuses (one 20mm, one 1.25")

1 Manual

1 Set of identification labels

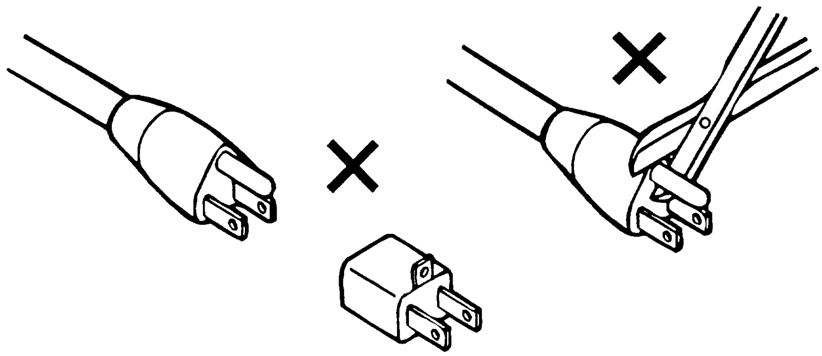
MAIN

1.2 Regulatory Notices

UL

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

The ground terminal 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.



WARNING: On multi-voltage units, check that the unit has been set to the correct supply voltage and that the correct fuse is installed. To reduce the risk of fire, replace the fuse only with the same type and rating.

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.

Safety Notices

IMPORTANT SAFETY NOTICE

This unit complies with the safety standard EN60065. The unit shall not be exposed to dripping or splashing and no objects filled with liquids, such as coffee cups, shall be placed on the equipment. To ensure safe operation and to guard against potential shock hazard or risk of fire, the following **must** be observed:

- Ensure that your mains supply is in the correct range for the input power 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

GB

IMPORTANT – NOTE DE SECURITE

Ce materiel est conforme à la norme EN60065. Ne pas exposer cet appareil aux éclaboussures ou aux gouttes de liquide. Ne pas poser d'objets remplis de liquide, tels que des tasses de café, sur l'appareil. 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

F

WICHTIGER SICHERHEITSHINWEIS

Dieses Gerät entspricht der Sicherheitsnorm EN60065. Das Gerät darf nicht mit Füssigkeiten (Spritzwasser usw.) in Berührung kommen; stellen Sie keine Gefäße, z.B. Kaffeetassen, auf das Gerät. 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

D

NORME DI SICUREZZA – IMPORTANTE

Questa apparecchiatura è stata costruita in accordo alle norme di sicurezza EN60065. Il prodotto non deve essere sottoposto a schizzi, spruzzi e gocciolamenti, e nessun tipo di oggetto riempito con liquidi, come ad esempio tazze di caffè, deve essere appoggiato sul dispositivo. 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

I

AVISO IMPORTANTE DE SEGURIDAD

Esta unidad cumple con la norma de seguridad EN60065. La unidad no debe ser expuesta a goteos o salpicaduras y no deben colocarse sobre el equipo recipientes con liquidos, como tazas de cafe. 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

E

VIKTIGA SÄKERHETSÅTGÄRDER!

Denna enhet uppfyller säkerhetsstandard EN60065. Enheten får ej utsättas för yttre åverkan samt föremål innehållande vätska, såsom kaffemuggar, får ej placeras på utrustningen." 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

S

BELANGRIJK VEILIGHEIDS-VOORSCHRIFT:

Deze unit voldoet aan de EN60065 veiligheids-standaards. Dit apparaat mag niet worden blootgesteld aan vocht. Vanwege het risico dat er druppels in het apparaat vallen, dient u er geen vloeistoffen in bekers op te plaatsen. 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

NL

DOLBY 363

NOISE REDUCTION UNIT

DOLBY 363 SR/A NOISE REDUCTION UNIT

The Dolby Model 363 provides two channels of noise reduction switchable between Dolby SR (Spectral Recording) and Dolby A-type. The economical 1-U high unit is intended for all kinds of audio facilities: music recording, video postproduction, broadcast, multimedia, and film.

Record/playback changeover of each channel can be controlled individually from the front panel, a tape recorder, or a remote control, allowing a single Model 363 to serve for stereo recording applications. Front-panel toggle switches select Dolby SR, Dolby A-type, or no processing. A setup button and four-element LED calibration displays allow quick alignment using internally generated Dolby tone for A-type or Dolby noise for SR. As with all other implementations of Dolby SR, the Model 363 features Auto Compare for verifying the performance of the audio system.

The Model 363 is also ideal for applications requiring dedicated encode or decode operation. A pair of 363 units can be used for simultaneous record and playback on two-channel tape recorders, or for transmission systems where the encoder and decoder are physically separated. In single-channel recording applications, a single unit can be used for simultaneous record and playback.

The Model 363 incorporates electronically balanced transformerless input and output circuits. Independent level adjustments for the record and playback signal paths allow accurate matching of existing line levels. A Check Tape switch allows monitoring of either the line-in signal or the encoded tape while recording. Bypass buttons make it possible to remove either or both channels from the entire audio system for studio alignment.

The Model 363 is normally supplied with two Cat. No. 300 modules, which contain both Dolby SR and Dolby A-type processing. A Dolby SR-only version using Cat. No. 350 modules is also available. Note: Cat. No. 22 and Cat. No. 280 modules cannot be used in the Model 363.

For multichannel applications, the Dolby SRP Series is available to provide up to 24 channels of Dolby SR noise reduction.

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Dolby Noise Reduction

Dolby noise reduction is a family of signal processes that reduce the noise inherent in analog recording media, without affecting the sound being recorded. While they differ in performance and details of operation, all Dolby NR systems are complementary processes that first encode the music when it is recorded, then decode it when it is played back. They also treat soft signals separately from loud ones, and vary the NR action with frequency to avoid the side effects typical of other systems.

Dolby A-type noise reduction, introduced in 1965, was originally intended for use by professional recording studios to make quieter master tape recordings. In the early 1970s its use was extended to film studios and motion picture release prints to make films sound better.

Dolby SR (spectral recording), introduced in 1986, was designed not only to provide more noise reduction than A-type, but also to enable analog master recordings that equal or surpass 16-bit digital recordings with respect to overall dynamic range. Today, analog with Dolby SR is still a preferred format for some musicians, producers, and recording and mastering facilities, and is the standard format for the analog soundtracks of virtually all feature films.

363 Specifications

Frequency Response[1]
20 Hz–20 kHz ± 1 dB encode/decode, at any level

Overall Distortion (THD)[1]
≤0.2% at Dolby level

Processor Headroom
+21 dB above Dolby level

Overall Dynamic Range[1]
SR:
105 dB, clipping level to CCIR/ARM noise level
105 dB, clipping level to NAB A-weighted noise level;[2] 95 dB, clipping level to unweighted noise level, 20 Hz–20 kHz[3]
A-type:
104 dB, clipping level to CCIR/ARM noise level
105 dB, clipping level to unweighted noise level, 20 Hz–20 kHz

Typical Obtainable Dynamic Range
SR: 90–95 dB; A-Type: 75–80 dB; typical at 38 cm/s (15ips) tape speed

Matching Between Units
±l dB at any level and any frequency, 20 Hz–20 kHz

Crosstalk
<–100 dB, 20 Hz–20 kHz processor off, or encode/decode

Signal Delay
6 µs for a single channel, 12 µs for overall encode/decode system

Analog Audio Inputs (0 dBr = 0.775 V_{RMS})
XLR connectors, balanced, 20 kΩ; common mode rejection: >55 dB, 50 Hz–10 kHz; maximum input level: +27 dBr balanced, +21 dBr unbalanced; input line levels from –10 dBr to +10 dBr can be adjusted to give Dolby level
Built-in Dolby tone and Dolby noise generators for calibration and channel identification

Analog Audio Outputs
XLR connectors, balanced, 20Ω; output level balance: within 1 dB into symmetrical 600Ω load; output float[4] better than –40 dB, 50 Hz–1 kHz; maximum signal level into 600Ω: +26 dBr balanced, +21 dBr unbalanced; output line levels in the range –10 dBr to +10 dBr can be adjusted to give Dolby level

Remote Control
9-pin male D-connector for individual-channel record/play changeover for remote control by the tape recorder; connector may be wired to allow remote-only or combined local and remote operation; changeover time: 3 ms maximum
15-pin female D connector for remote control of SR/Off/A in/out and Setup functions; also provides a remote status indication of A/SR and Auto Compare function

Front-Panel Controls and Indicators
Individual channel controls:
A/OFF/SR—Selection of processing type
NORMAL/CHECK TAPE—Allows checking of the nondecoded signal from tape during recording or playback
RECORD/PLAY—Control of record/play switching
BYPASS—Provides relay-controlled "hard" bypass of all circuitry
LEVEL CONTROLS—Four multi-turn level controls per channel for setting levels to and from the console, and to and from the tape recorder
SETUP—For use during alignment and calibration
Calibration Display:
Four-LED display per channel for setting Dolby level

Dimensions and Weight
1-U Rackmount: 44 x 483 x 285 mm (1.75 x 19 x 10.2 inches); a further 65 mm (2.5 inches) required for standard XLR connectors
Net: 6.3 kg (14 lbs) including two Cat. No. 300 modules

Power Requirements
230 V version: 198–264 VAC, 50–60 Hz, 40 W; uses one 20 mm T250 mA fuse
Multi-voltage version: User selected 85–132 VAC, 50–60 Hz, 40 W; uses one 1.25-inch 500 mA slow-blow fuse, or 187–264 VAC, 50–60 Hz, 40 W, uses one 20 mm T250 mA fuse

Operating Conditions
Up to 40° C (104° F)

Regulatory Notices
US: This unit is UL listed.
Europe: The 230 Volt unit complies with the requirements of Low Voltage Directive 73/23/EEC and EMC Directive 89/336/EEC.

Warranty
One-year limited, parts and labor

All specifications apply with input and output controls set for Dolby level equal to +4 dBr = 1.23 V_{RMS}, balanced source and load.
0 dBr is defined as 0.775 V without regard to impedance.
[1] Two units back-to-back, encode/decode
[2] Weighting filter supplemented by 25 kHz 4-pole lowpass filter to ensure that only audible noise is measured
[3] Average-responding or RMS meter, 4-pole filters
[4] Output float is the level across a balanced load relative to an interfering signal injected at one end of the load

Specifications subject to change without notice.

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SECTION 2
INSTALLATION

2.1 Installation

STEP 1

Unpack the unit from its box and check for any damage. Be sure to check the packing material for the processing module(s), mains cord, manual, and other accessories. The processing modules are separately packed in individual boxes.

STEP 2

Mount the unit appropriately in a rack or in a tape recorder. Ensure that there is air flow around the unit, and that it is not mounted directly above any other heat-producing equipment. The unit will operate within specifications up to an ambient temperature of 40°C (104°F); note that the ambient temperature inside a poorly-vented rack or inside a tape recorder may be considerably higher than that in the room.

STEP 3

Model 363 - 230V Unit (for multi-voltage units, skip this step)

Open the fuse compartment door in the AC mains input connector with a small flat-blade screwdriver as shown below and check that the fuse has the correct rating (T250 mA 20mm time-lag). The fuse carrier must be inserted into the compartment with the orientation as shown. Do not force the carrier into the compartment - damage will result. Close the compartment door, making sure it clicks firmly into place.

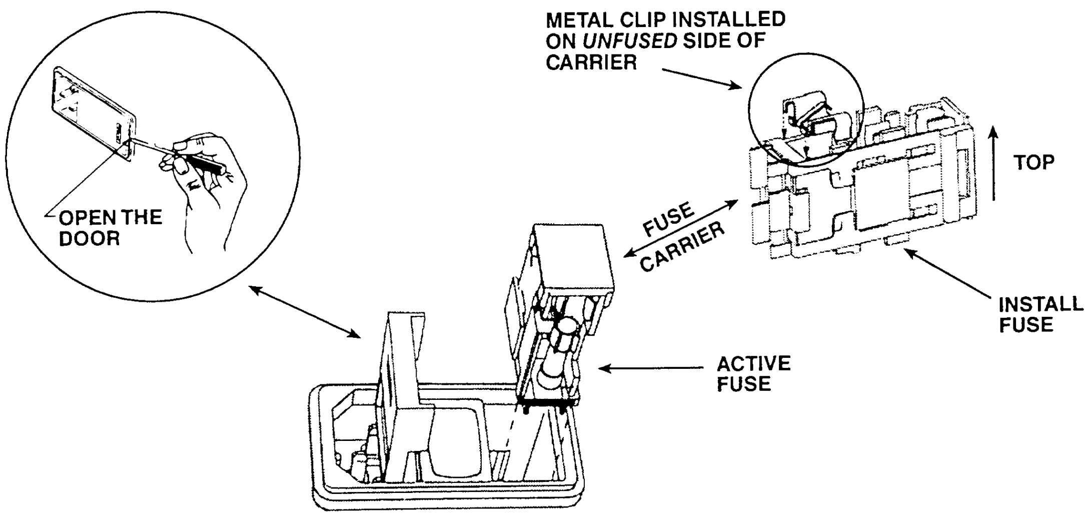


Figure 2.1

Model 363 - Multi-Voltage Unit

Open the voltage selector/fuse compartment door in the AC mains input connector with a small flat-blade screwdriver as shown below. Rotate the voltage selector drum until it reads the correct mains voltage. (The drum may also be removed and replaced with the correct voltage displayed. It will only fit one way around.)

The compartment has positions for two fuses and will accept carriers for either 20mm or 1.25" fuses. Only the lower fuse position is electrically connected. Select the appropriate fuse and carrier, and insert into the lower position with the arrow in the carrier in the same direction (downwards) as the arrows inside the compartment door. A spare fuse of the same type and rating can be placed in the upper position. Close the compartment door, making sure it clicks firmly into place.

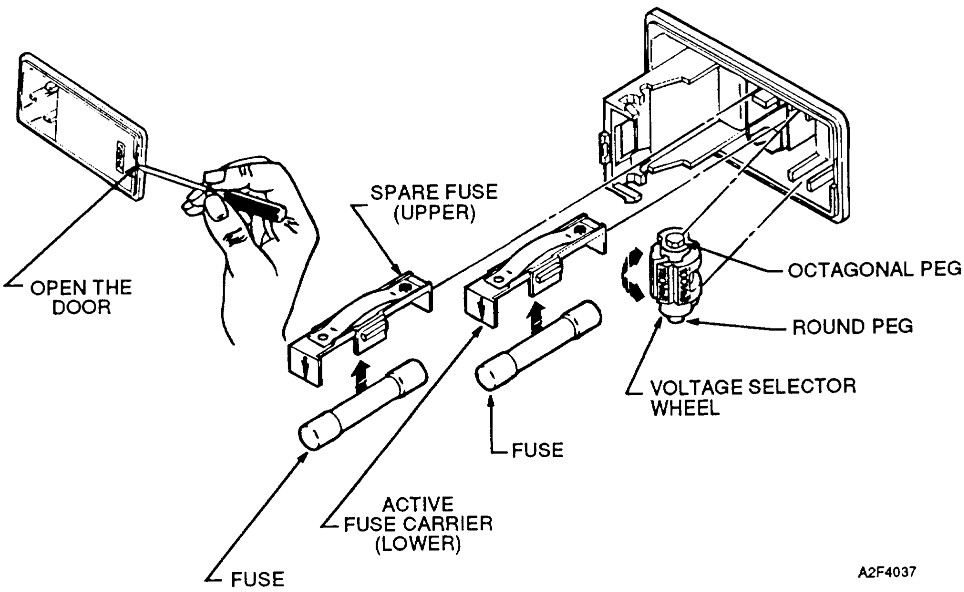


Figure 2.2

2.2 Connections

STEP 1

Connect the Model 363 to mains power

STEP 2

Connect audio cables to the inputs and outputs. For optimum EMC performance, wire the shields of the audio cables to the outer shells of the XLR connectors, not to pin 1 which should, if possible, be left open. If other items of equipment connecting to the Model 363 have unbalanced inputs or outputs, use twin-screened cable and wire the connectors at the Model 363 end exactly as if the circuits were balanced; perform the unbalancing at the other end.

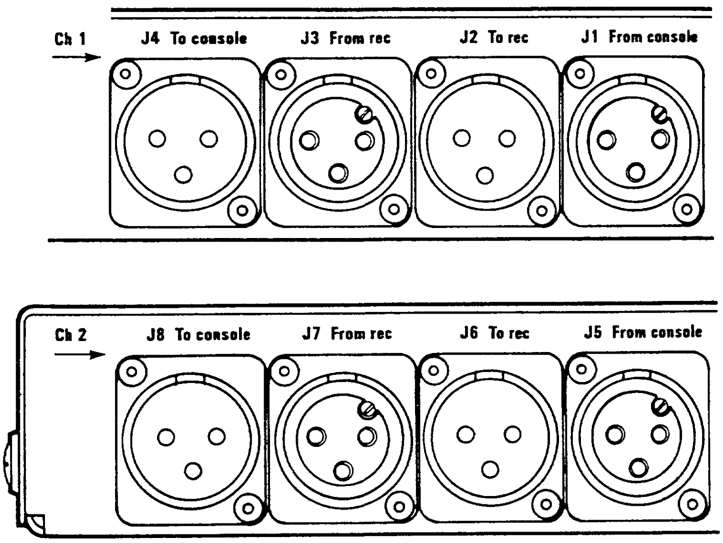


Figure 2.3

NOTE 1: Current IEC wiring convention calls for XLR pin 2 to be "high/hot" and pin 3 "low/cold". In a balanced system, the distinction is arbitrary provided there are no phase inversions through the unit; the Model 363 maintains phase. As the Model 363 is fully floating, it is unimportant which of the two signal pins is grounded, so long as the same convention is used on all inputs and outputs..Nevertheless, in the interests of maintaining international standardization, we suggest the IEC recommendation is followed.

NOTE 2: It is normal practice to connect program (audio) ground to power line ground for many reasons, including safety. On occasion (particularly with remote location recording) induced hum can sometimes be reduced by separating the two grounds. Link J1101 provides this feature. This link is accessible through the left hand side of the left hand front panel cut-out after first removing the processing module. Long-nose pliers provide the easiest method of moving the jumper from its normal closed position (towards the front of the unit) into the open positions (towards the rear; see Figure 2.4). Note that there is always a 1kohm resistor across the link, which remains in the open position so that the audio ground is never totally isolated from the chassis ground. Also note that the chassis is always connected to the ground pin of the power line cord. for safety reasons, this ground should NEVER be disconnected.

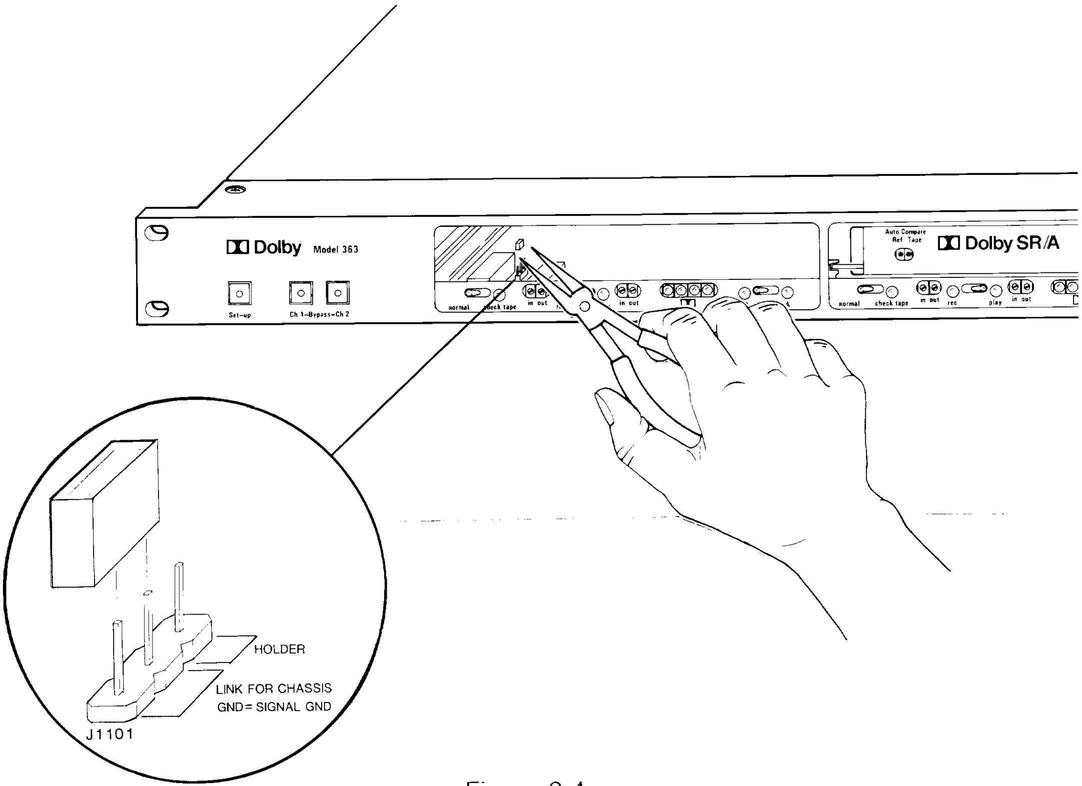


Figure 2.4

STEP 3

If remote control of the record/playback functions is required, wire up a standard 9-pin female cable-mounting D-connector (connected to J9, **rec/play remote**) as follows (Figure 2.5).

Each channel can be switched from the playback mode to record mode by applying a voltage in the range 4 - 25 V dc to pins 3 and 4 for channel 1, and pins 7 and 8 for channel 2, with polarities as shown in Figure 2.5 (channel 2 pin numbers shown in parentheses). Many tape recorders have suitable switching voltages available; note the switching input on the Model 363 uses an opto-isolator to provide a floating input. The input current is roughly 5 mA, essentially independent of voltage. Switching voltages higher than 25 V can be accommodated by adding an appropriate series external resistor.

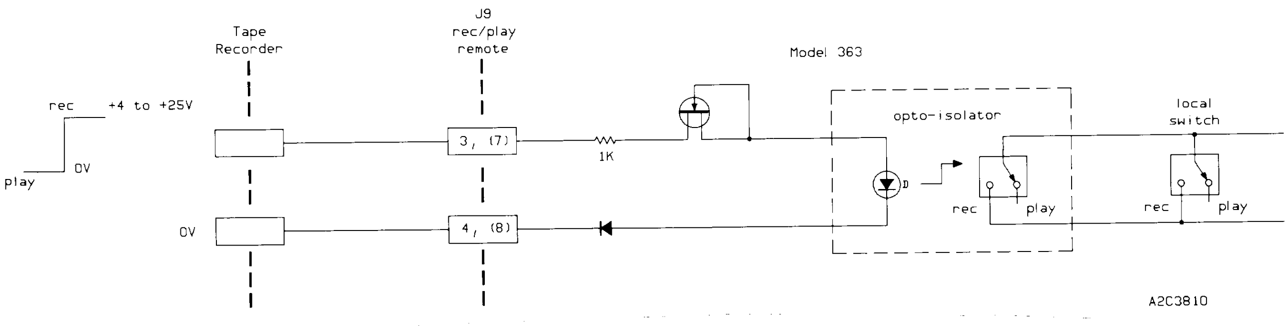


Figure 2.5 Remote control of mode (voltage-operated)

MAIN

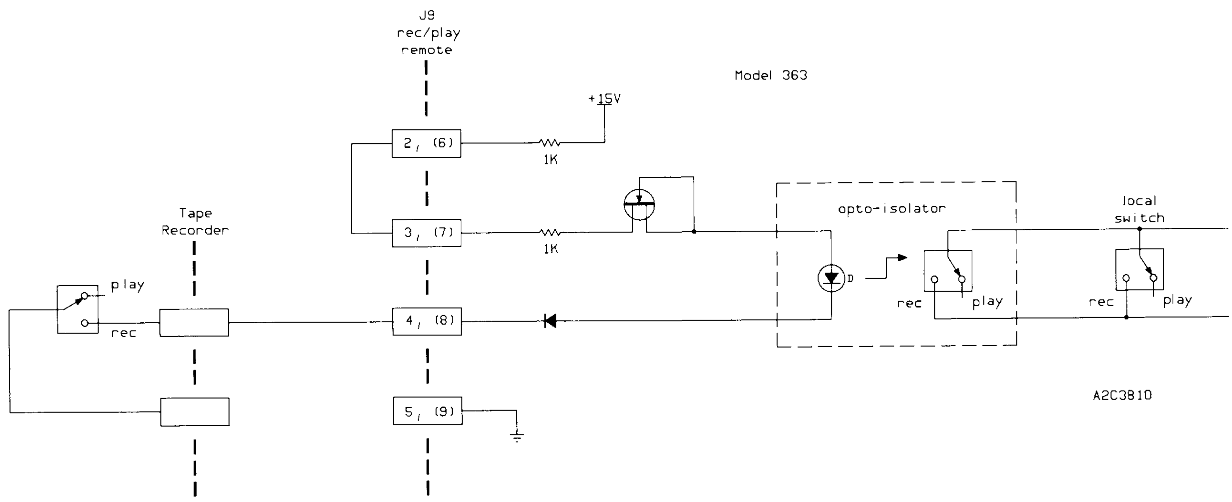


Figure 2.6 Remote control of mode (using isolated contact)

Alternatively, some recorders provide an isolated make contact which can be used by wiring the connector to pick up the supply voltage in the Model 363 (Figure 2.6).

Wired in either manner, the unit is switched to Record using either the local **rec/play** switch, or by the remote connection provided the local switch is left in the **play** position. By connecting pin 1 to pin 5 or 9, the local control is inhibited and only the remote switching system will effect the change (Figure 2.7).

At the end of this section (starting on page 2.8) there are specific diagrams for several of the commoner two-track recorders.

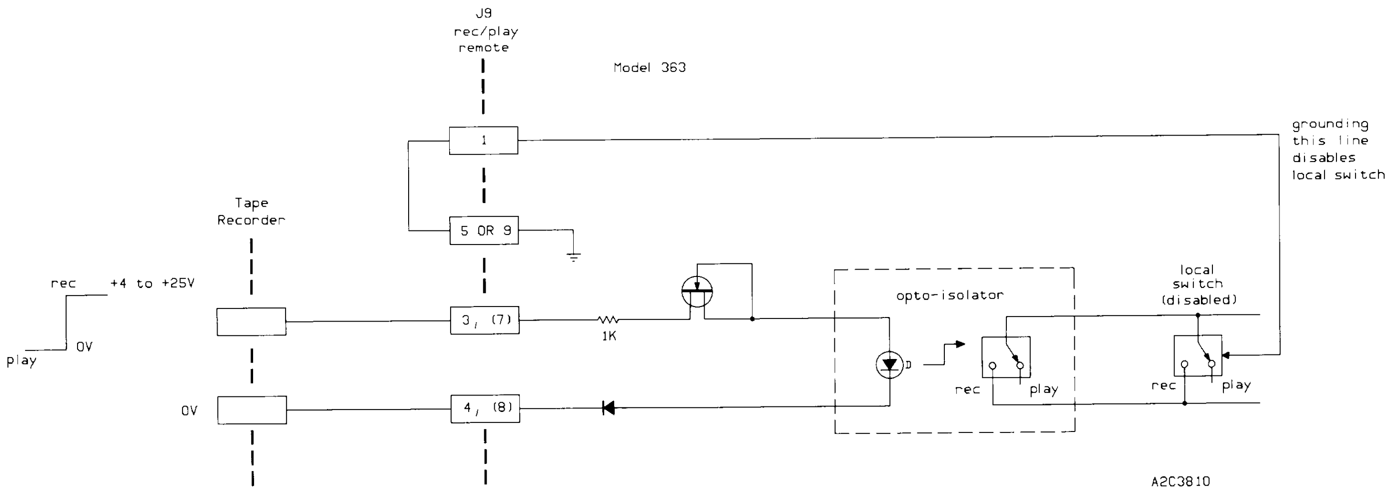


Figure 2.7 Disabling local 363 rec/play switch

MAIN

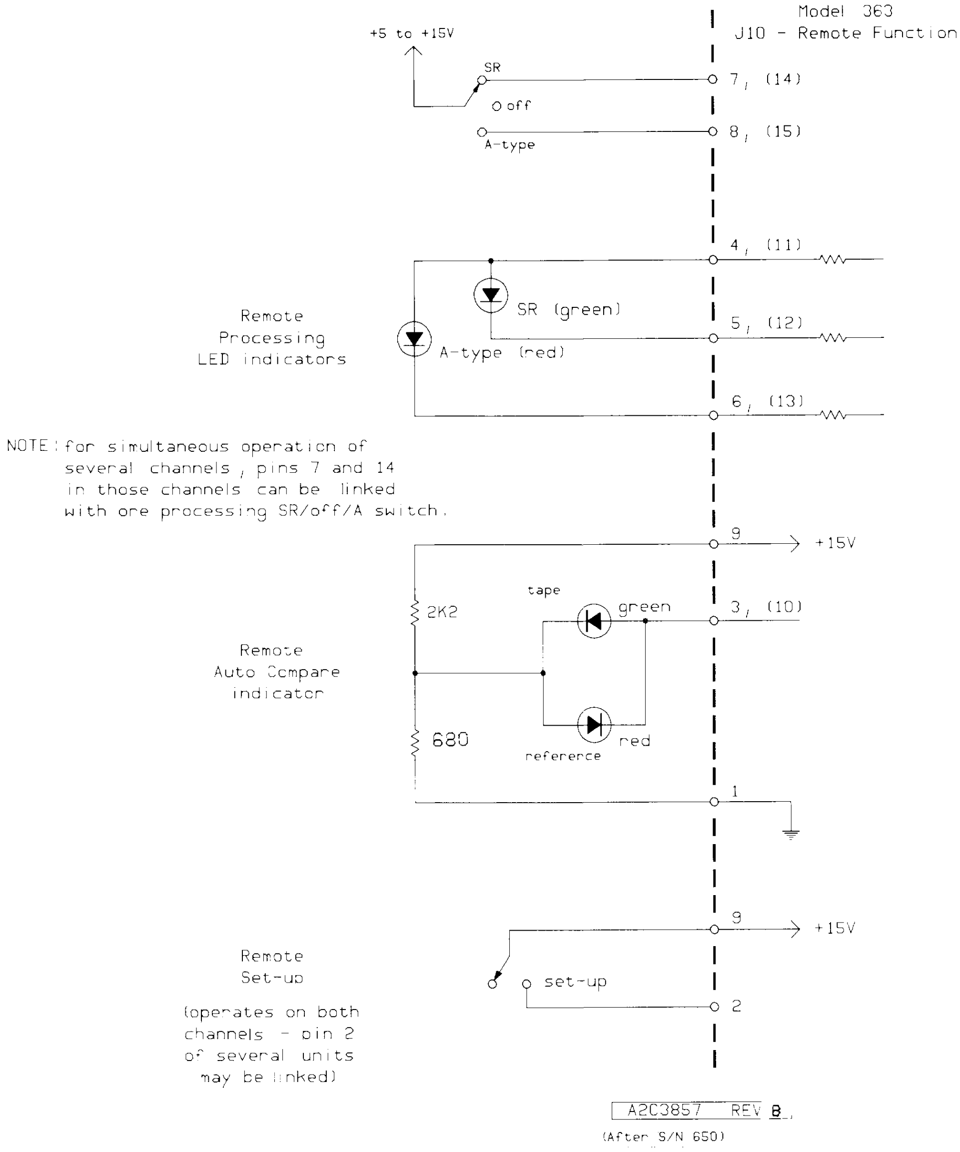


Figure 2.8 Remote function wiring

STEP 4

Other remote functions can be obtained by wiring a standard 15-pin male D-connector to J10 (function remote) as shown Figure 2.8. Remote processing selection, remote indication of signal processing selected (depends on which processing module(s) are installed), and remote mimic of the state of the Auto Compare function are provided for each channel independently. Remote Set-up control is also provided, and (as with the local control on the Model 363) is common to both channels.

It is also possible to link the processing selection and Set-up functions on several Model 363 units using the J10 connector. For common control of processing selection, link all pins 7 and 14 together, and separately link all pins 8 and 15 together. The remote switch is wired as above to any pins 7 and 8. For remote operation, the local switch must be in the "off" mode. If the local switch is moved to either A-type of SR, it will take precedence over the remote switch and the unit (and both local and remote indicators) will switch to the mode selected by the local switch.

For common control of the Set-up function, link all pins 2 together, and connect a switch between pin 9 on any unit and these linked pins. In this case, both the remote switch and <u>any</u> local switch will switch all the linked units into the Set-up mode; since the switches are locking switches, it is important to release the correct switch to cancel Set-up.

STEP 5

Insert the signal processing module(s) into the guides visible through the front panel. The modules may be either Cat. No. 300, Cat. No. 350, or Cat. No. 450. The unit will operate with only one channel of signal processing installed, or with a mixture of module types. Circuitry in the unit frame recognizes which module is in each channel, and prevents incorrect indications being given for the channel switches. For example, if the Cat. No. 350 (which only has SR processing) is installed, switching to A-type will not illuminate the green LED, and the module will remain in the "off" mode.

STEP 6

Track identification stickers are supplied which can be used to differentiate between multiple units mounted together. We suggest these are stuck on the front extrusion to the left of the left module cut-out and to the right of the right cut-out. There is also space on the rear panel to add stickers; a double set is supplied with each unit for labelling both back and front.

NOTE

Prior to unit serial number 599, the remote mode selector only selected processing in/out, and was wired as shown on the next page.

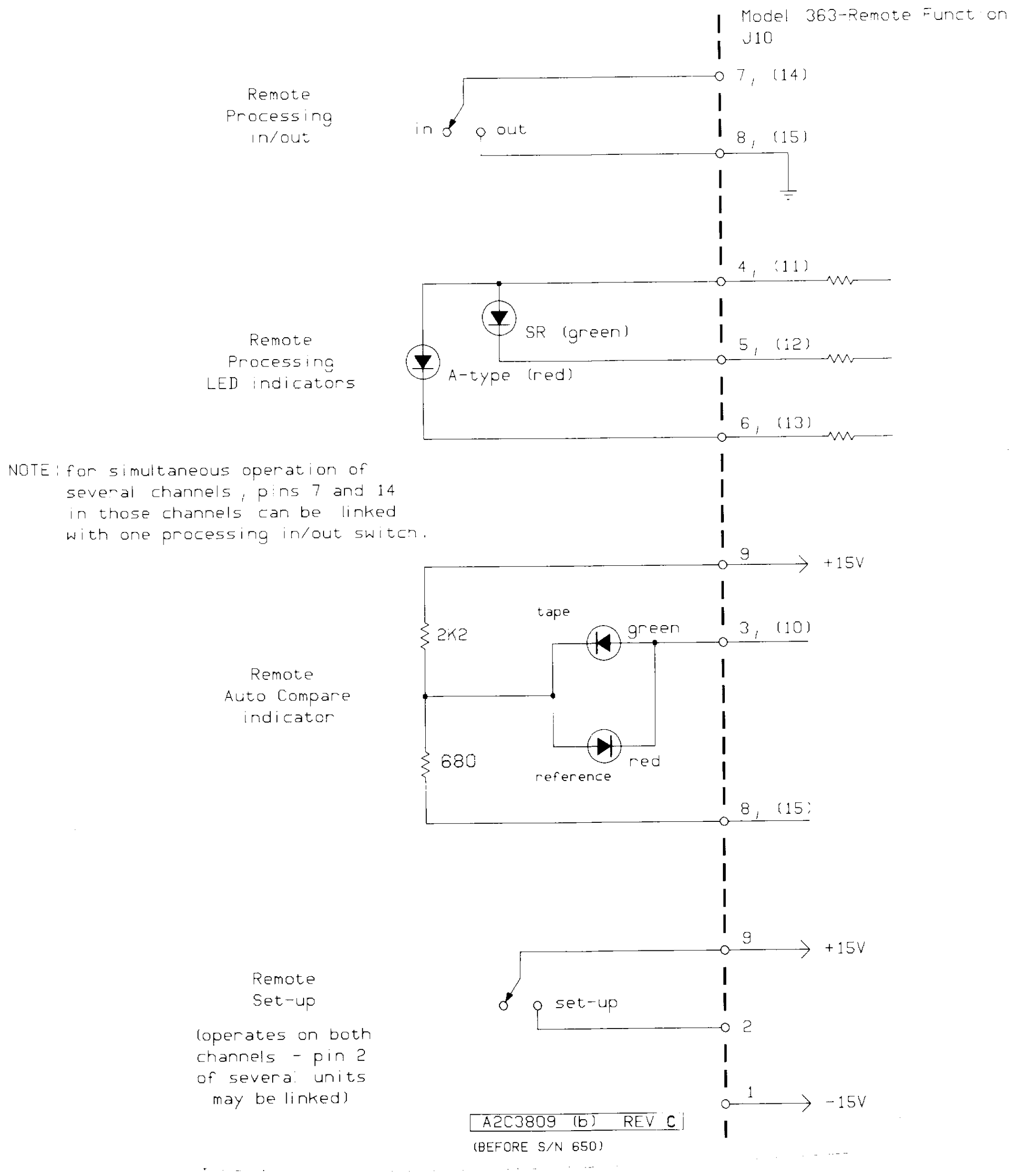


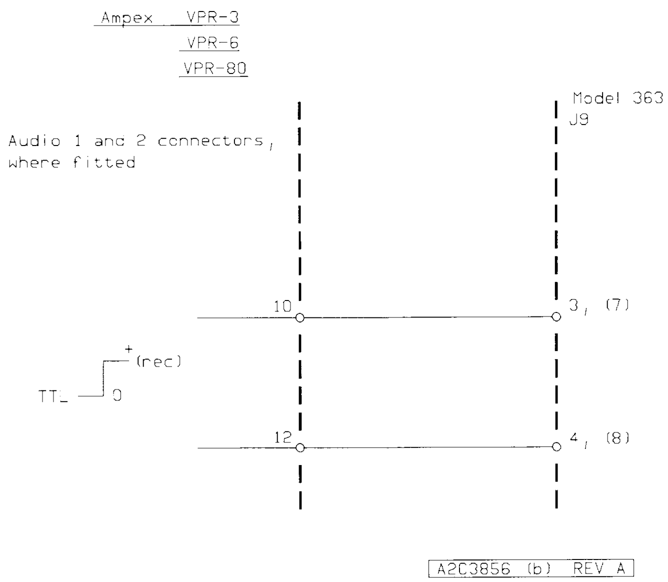
Figure 2.9 Remote function wiring (prior to unit number 650)

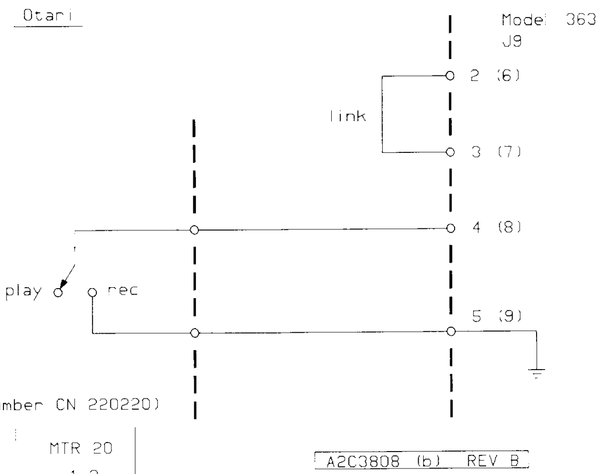
Note that the operation is somewhat different from that described previously, especially in that relationship between local and remote controls. In addition, connections for the remote Auto Compare indicator are different. It is possible to link this older processing on/off control on several Model 363 units using the J10 connector. For common control of processing in/out, link all pins 7 and 14 together and connect a switch between pin 8 or 15 on any unit to these linked pins. Note the difference from later units; with the master remote switch in the "on" position, the local switch on each channel can still be used to turn off that particular channel. With the master switch in the "off" position, the local control is disabled and processing remains switched off.

If the wiring for an older unit is connected to a later unit (or vice versa), no damage will be done, but control will not be effective. The local and remote mode and nr type indicator of any unit will always indicate the actual operating mode of the unit.

MAIN

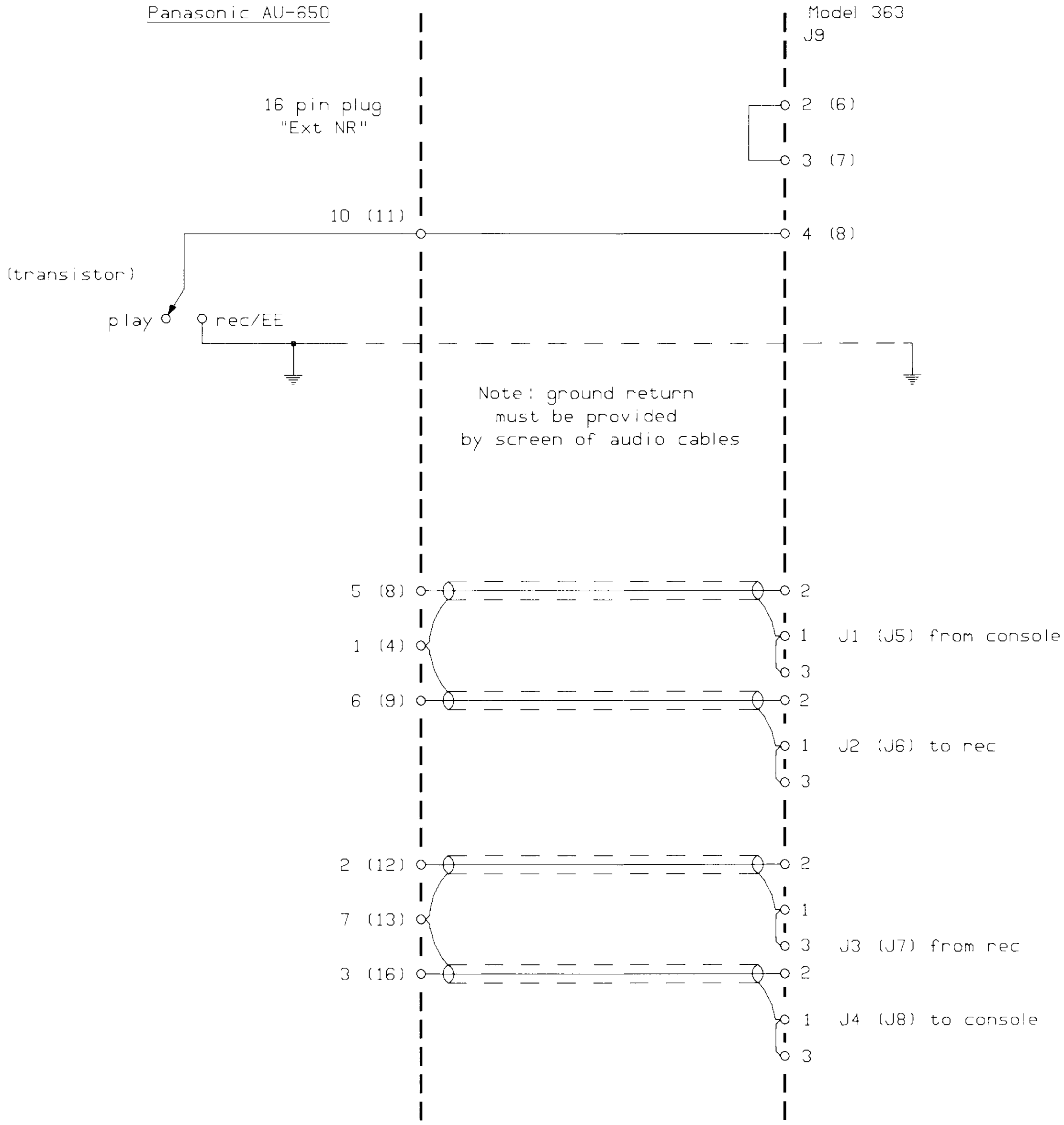
2.3 Connection Diagrams for Specific 2-Track Recorders





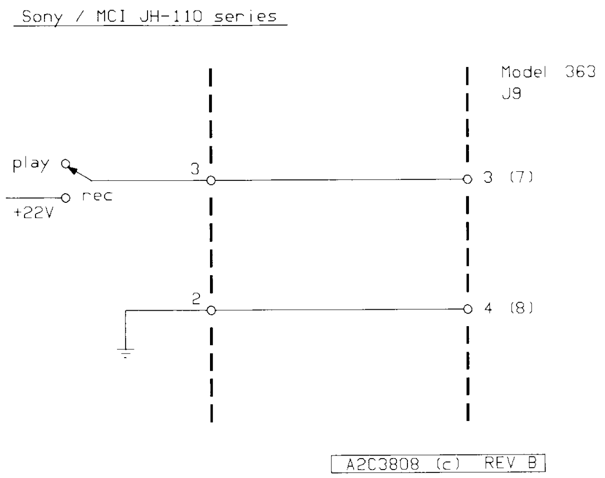
(use Otari connector part number CN 220220)

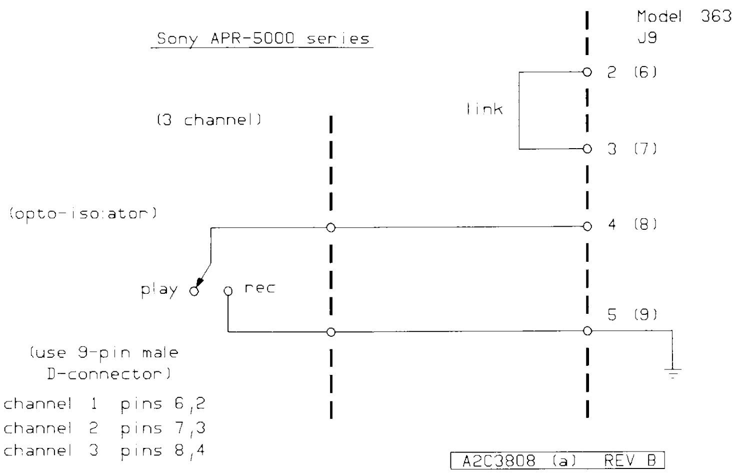
		MTR 10/12	MTR 20
channel	1	pins 1,2	1,2
	2	pins 3,4	3,4
	3	pins 11,12	5,6
	4	pins 13,14	7,8

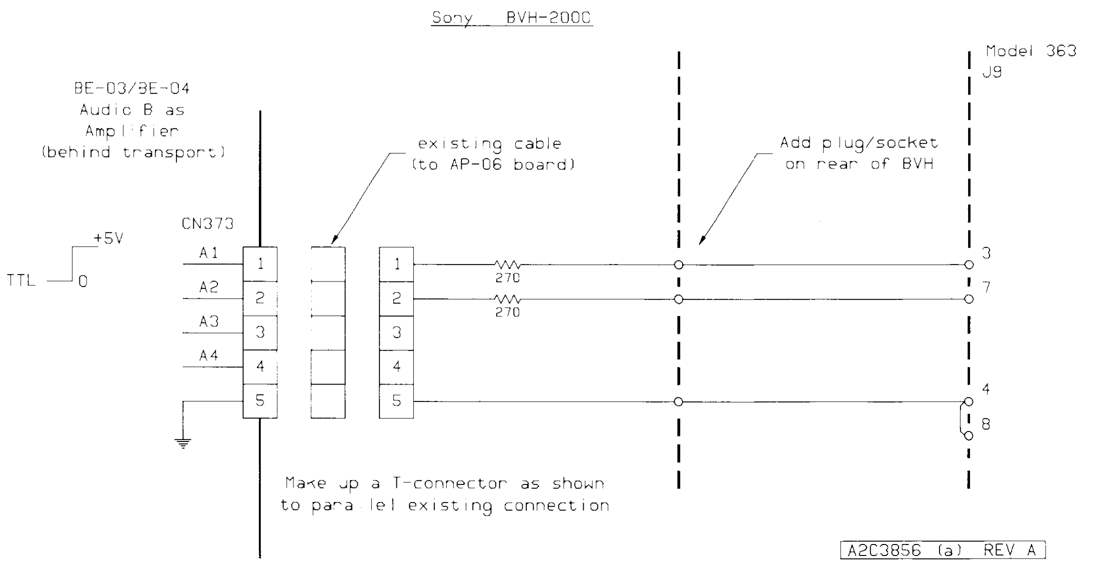


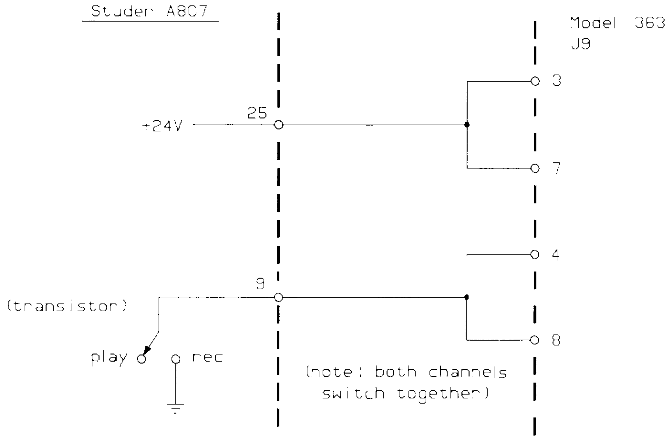
A2C3855 REV A

MAIN

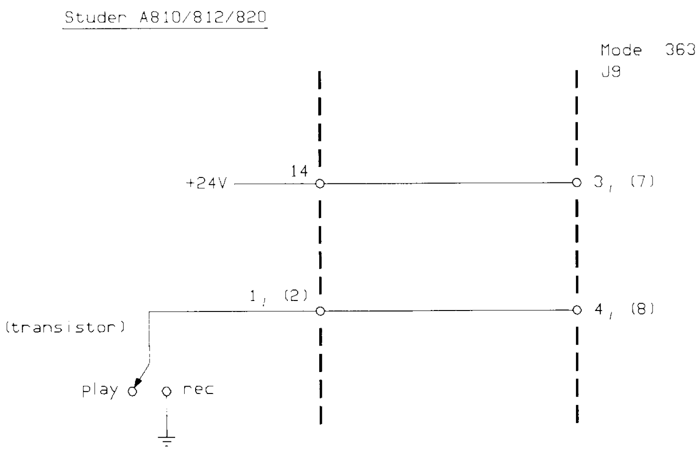

Sony / MCI JH-110 series
Model 363
J9
play
rec
+22V
3
2
3 (7)
4 (8)
A2C3808 (c) REV B


Sony APR-5000 series
Model 363
J9
(3 channel)
link
2 (6)
3 (7)
(opto-isolator)
4 (8)
play
rec
5 (9)
(use 9-pin male
D-connector)
channel 1 pins 6,2
channel 2 pins 7,3
channel 3 pins 8,4
A2C3808 (a) REV B


Sony BVH-2000
Model 363
J9
BE-03/BE-04
Audio B as
Amplifier
(behind transport)
existing cable
(to AP-06 board)
Add plug/socket
on rear of BVH
CN373
+5V
TTL
0
A1
A2
A3
A4
1
2
3
4
5
270
270
3
7
4
8
Make up a T-connector as shown
to parallel existing connection
A2C3856 (a) REV A



A2C3808 (d) REV B



use Studer connector
part number 20.020.303.08

A2C3809 (a) REV B

SECTION 3
APPLICATIONS

There are two ways of using the Model 363 (in addition to a bypass condition). These are:

1. Switched operation, where the mode is switched between record and play. One Model 363 unit (two channels) can therefore be used for a two-track recorder.

and

2. Dedicated Encode or Decode operation, where the channels are permanently assigned. Two Model 363 units are required for a two-track recorder.

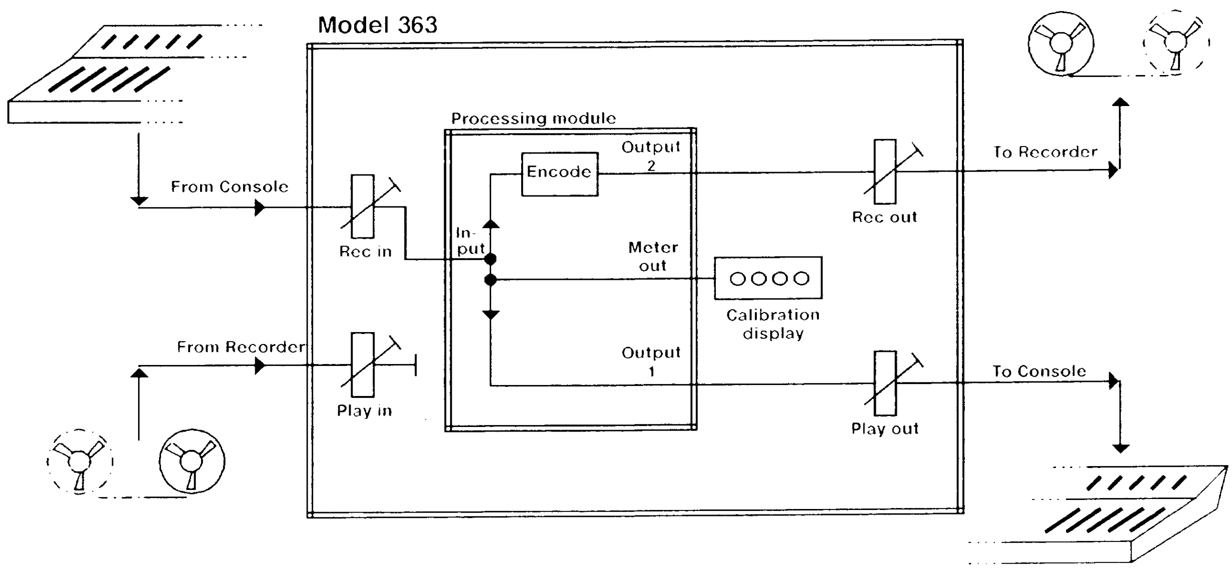
3.1 Switched operation

The processor channel is switched either into the RECORD signal path or into the PLAY signal path. The switching may be controlled manually from the front panel of the Model 363, or more often automatically by the tape recorder. (Controls on the Model 363 are identified on the fold-out drawing at the end of Section 4.)

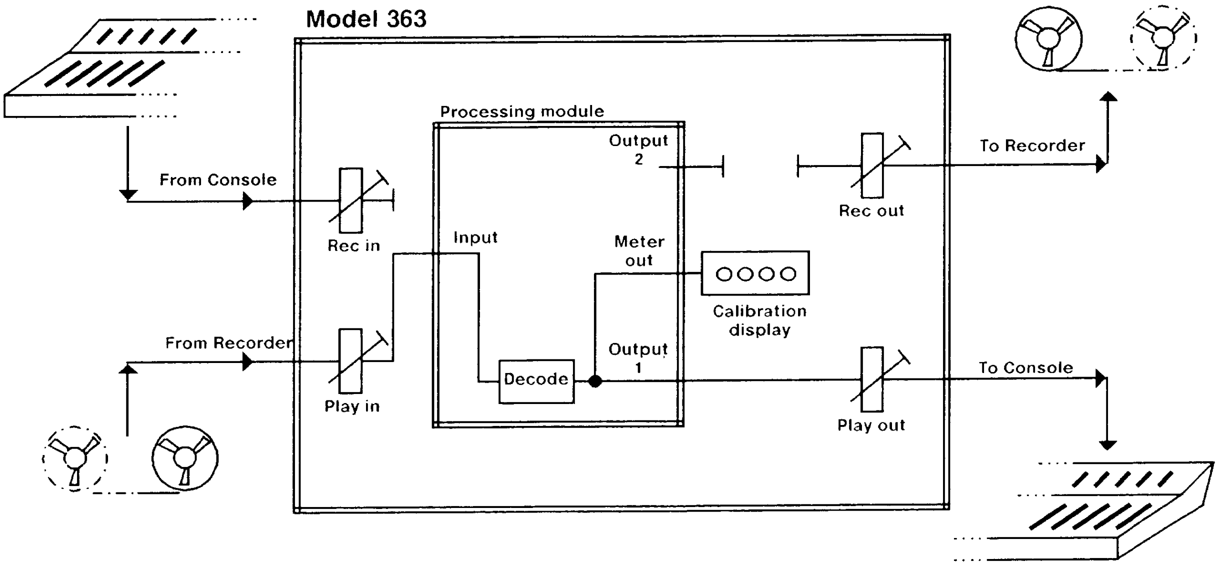
Since each channel of the recorder has only one processor it is not possible to decode the signal "off tape" during recording. In normal operation the console output is heard during recording. A switch labelled **normal-check tape** is provided to allow the operator to hear the playback signal directly without any processing. In both record and play modes, the encoded tape can be monitored by using the **check tape** position of the switch. The switch should usually be left in the **normal** position. The output signal is then the unprocessed "line-in" signal while recording, or the properly decoded signal during playback.

This switched operation is commonly used with two track tape recorders in recording studios, and sometimes with video recorders.

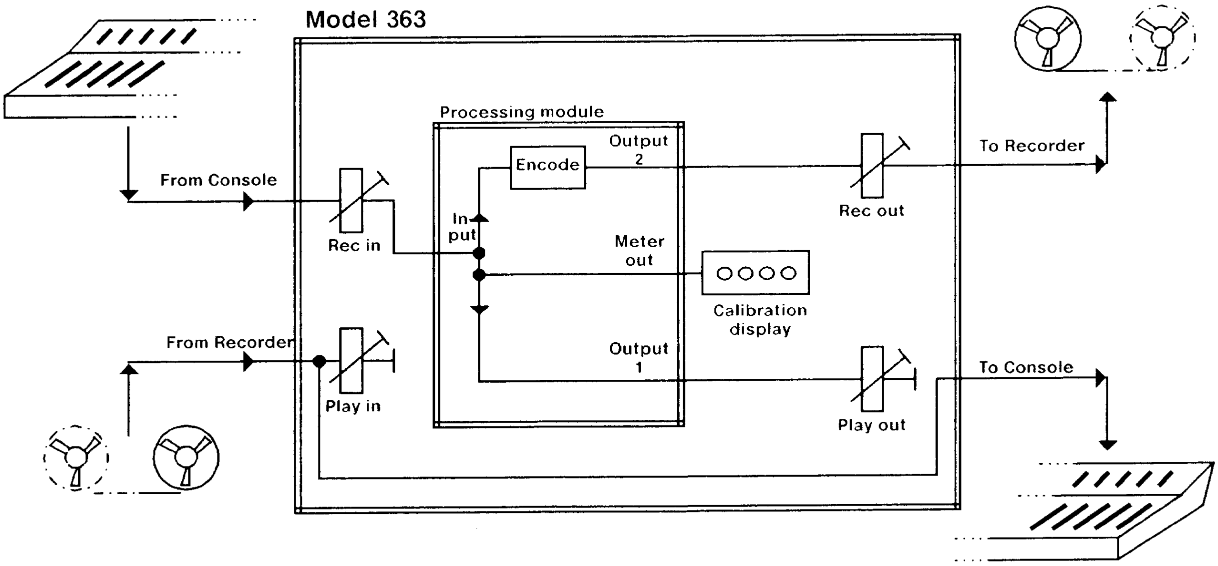
3.1.1 Signal paths in **RECORD**



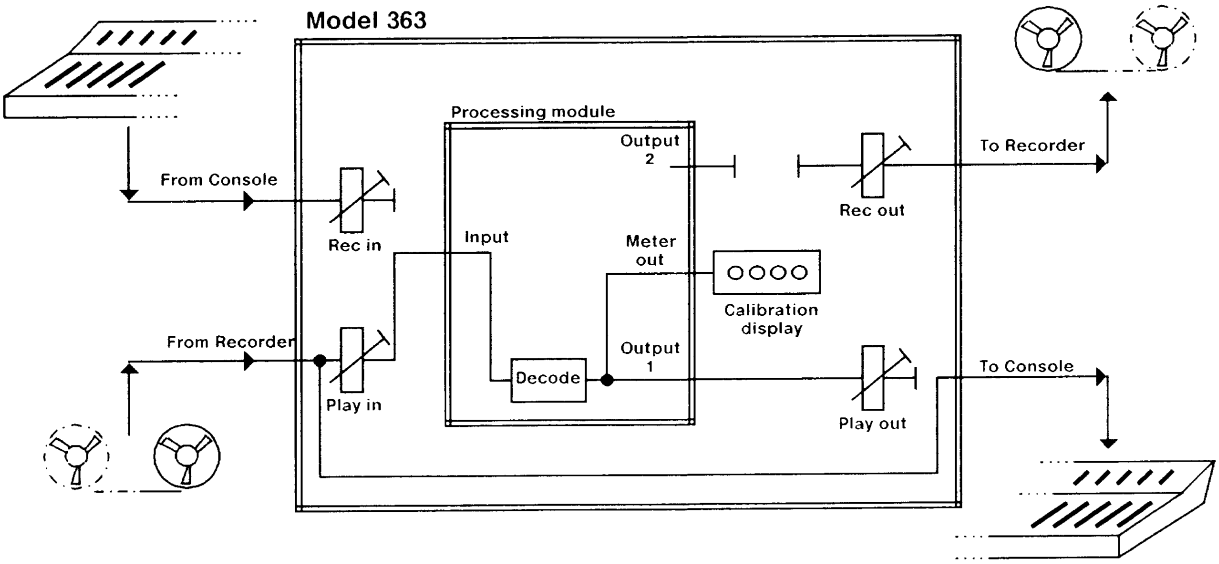
3.1.2 Signal paths in **PLAY**

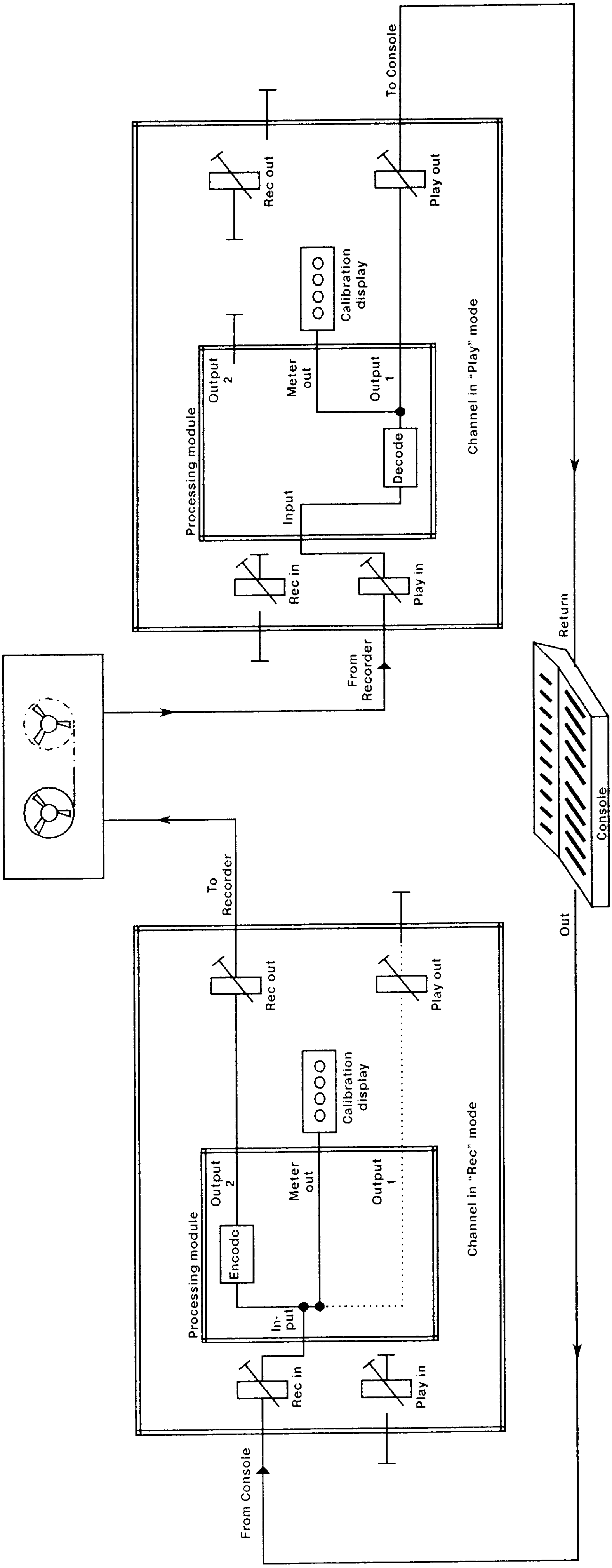


3.1.3 Signal paths in **RECORD and CHECK TAPE**



3.1.4 Signal paths in **PLAY and CHECK TAPE**





3.2.1 Encode/Decode operation

MAIN

3.2 Encode/Decode Operation

Two processors are needed for each channel of the recorder. One processor is permanently in the record signal path, and one in the play signal path. This arrangement is used where it is important to listen to the signal “off tape” or “E-to-E” during a recording, or where encoding and decoding must be carried out simultaneously.

This method is used with film and most video recorders and transmission lines, and may also have applications with master tape recording.

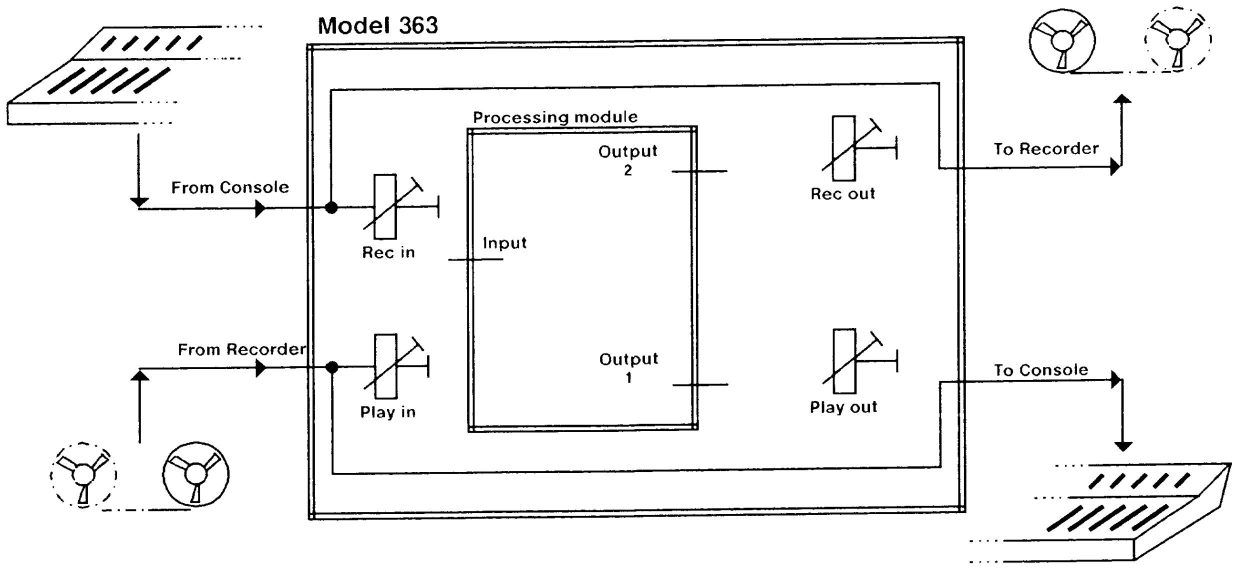
When using two model 363s with a two-track recorder, it is possible to use one Model 363 to provide two channels of encoding and the other to provide two channels of decoding, or alternatively to use one Model 363 to provide encoding and decoding for track 1 and the other for track 2. We would recommend the latter approach (as shown on the next page) which has some advantages in the Set-up mode (notably in the ease of entering this mode).

3.3 Bypass

Bypass is used during alignment or to remove the Model 363 from the audio paths. When the power is removed, the unit is bypassed automatically.

In **bypass**, the Model 363 inputs are connected in parallel across the outputs of the console and recorder. The inputs to the recorder and console are connected by relay contacts to these inputs, with the noise reduction process disconnected from the signal path as shown below.

The impedance in shunt across the two inputs is about 20 kohm with power on, and about 10 kohm with power off.



SECTION 4
LEVEL STANDARDIZATION AND OPERATION

4.1 Level standardization for Dolby systems

Dolby A-type and Dolby SR are complementary noise reduction systems; that is, the processing applied during playback is a mirror image of that applied when recording. In addition, the processing is dependent on both the level and frequency of the signal being recorded. This gives great flexibility in the way that the processing adapts to the incoming signal, but it means that for correct operation the levels in the playback processor must be the same as those in the record processor. In most studios all this really means is that the tape recorder should be at unity gain, which is the normal situation.

To ensure that the play processor gets the same signal level as the original record processor (which often is in a different studio), it is necessary to record a test signal onto the tape to indicate how the Dolby processing was set when the recording was made. For Dolby A-type this signal is called Dolby tone, and for Dolby SR it is called Dolby noise.

4.2 Level standardization for Dolby A-type noise reduction

Every Dolby A-type noise reduction unit contains an oscillator that produces a distinctive reference signal at a closely defined internal level (called Dolby level) and a calibration display to indicate this Dolby level. When the Set-up button is pressed, Dolby tone is fed into the record signal path and the calibration display indicates the level in the play processor. (In products designed before the introduction of Dolby SR, the Set-up function is controlled by a button labelled Dolby tone.) Dolby tone is made distinctive by frequency modulating the basic 850 Hz frequency upward by 10% for 30 ms every 750 ms. This easily recognized signal indicates that the tape has been recorded with Dolby A-type noise reduction, and also represents the tape fluxivity corresponding to Dolby level used on the recording.

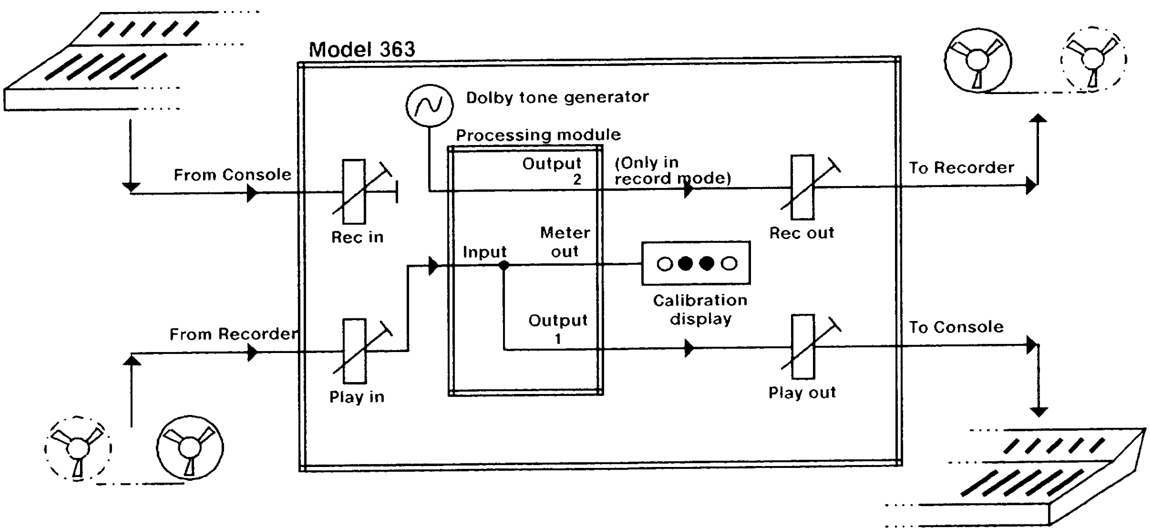
For correct operation, Dolby tone recorded on tape should produce a reading of Dolby level on the calibration display when the Set-up button is pressed. Many different parts of the audio industry have standardized their Dolby levels to ease interchange of material. The list below gives some examples.

Typical Dolby levels

Application	**--------Dolby level-------**	
	Meter reading	Level
Recording Studio, Europe	0 VU	320 nWb/m
Recording Studio, USA	0 VU	250 nWb/m
Most video formats	----	100 nWb/m
35mm Magnetic film	----	185 nWb/m
35mm Optical film		50%

If no standard is listed for your application, please contact Dolby Laboratories for discussions on an appropriate Dolby level. The choice of Dolby level is influenced by system noise, system headroom, type of metering systems, and type of transmission or recording method. If you have to make a quick decision, check with other possible local users, or choose a level which is easy to read on your studio meters.

4.2.1 **When the SR/off/A switch is in the A position,** the signal paths in Set-up are:



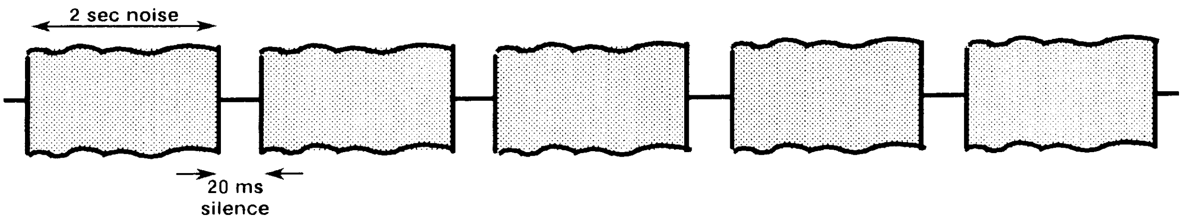
The calibration display will read the level returning from the tape recorder, and, if the Model 363 is switched to **record**, Dolby tone will be the output to the recorder.

This mode is also used to align the Model 363 to Dolby tone played back from a previously recorded tape.

4.3 Level Standardization for Dolby Spectral Recording

Dolby SR also has its own distinctive calibration signal called "Dolby noise". Dolby noise, like Dolby tone, not only serves as a reference level but also indicates that a recording was encoded with Dolby SR.

As with Dolby tone, Dolby noise has been made distinctive. It consists of pink noise interrupted every two seconds by 20 ms gaps, as shown diagrammatically below (note actually the nick is much shorter than shown, and also noise is a much less regular signal than in the diagram).



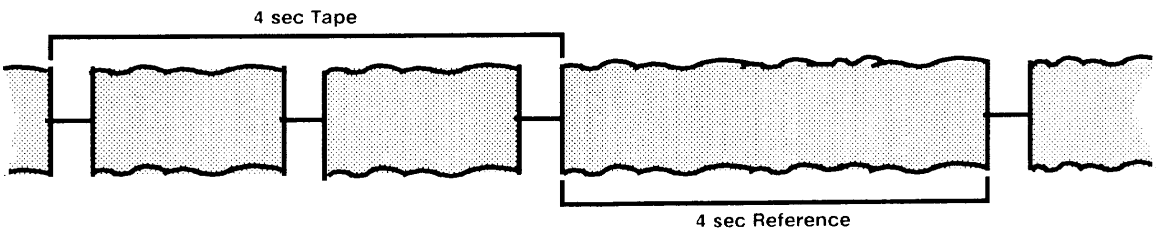
Dolby noise is used in a similar way to Dolby tone for calibration and alignment. Dolby noise recorded on tape should produce a reading of Dolby level on the calibration display when the Set-up button is pressed.

Unlike Dolby tone, Dolby noise is not recorded on the tape at Dolby level but at 15 dB* below Dolby level. Dolby level is usually around 0 VU; recording Dolby noise at this level would risk saturation at both high and low frequencies (especially at low tape speeds or when using thin oxide tapes). Thus the noise recorded on the tape would not be a true reflection of the response of the recorder. To prevent these errors, Dolby noise is recorded at a lower level.

When the **Set-up** button is pressed **and** **SR** is selected on the Model 363, the calibration display circuit gain is increased so that Dolby noise will still read Dolby level on the Model 363 display even though it is not recorded at Dolby level on the tape. (In addition the Dolby noise is filtered in the meter circuit only to remove the low-frequency components of the pink noise which would otherwise give a display signal which would be difficult to read.)

As well as providing level information, Dolby Noise also makes checking the frequency response quick and easy. Audibly comparing the Dolby noise recorded on tape with a Dolby noise generator will show up any differences caused by either gain or frequency response errors in the complete record-replay chain; this test is remarkably easy to perform, and is extremely critical.

This comparison is made even easier by a feature called Auto Compare. During Auto Compare, the monitor output of the Model 363 switches between the the signal replayed from tape and the internal Dolby noise generator, in the sequence shown below. In general, if Auto Compare sounds OK, all aspects of the complete recorder/Model 363 chain are operating and are calibrated correctly. Note that the noise from tape is Dolby noise and so has 20 ms gaps, whereas the reference noise from the generator has no gaps. This pattern makes it easy to tell the difference between "Tape" noise and "Ref" noise.



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* Note that, due to the nature of pink noise, the level indicated on any meters in the studio will depend critically on the type of meter used. Do not try to read the level of Dolby noise with your studio meters.

MAIN

The Auto Compare sequence will start whenever

...... Dolby noise is being replayed from tape;

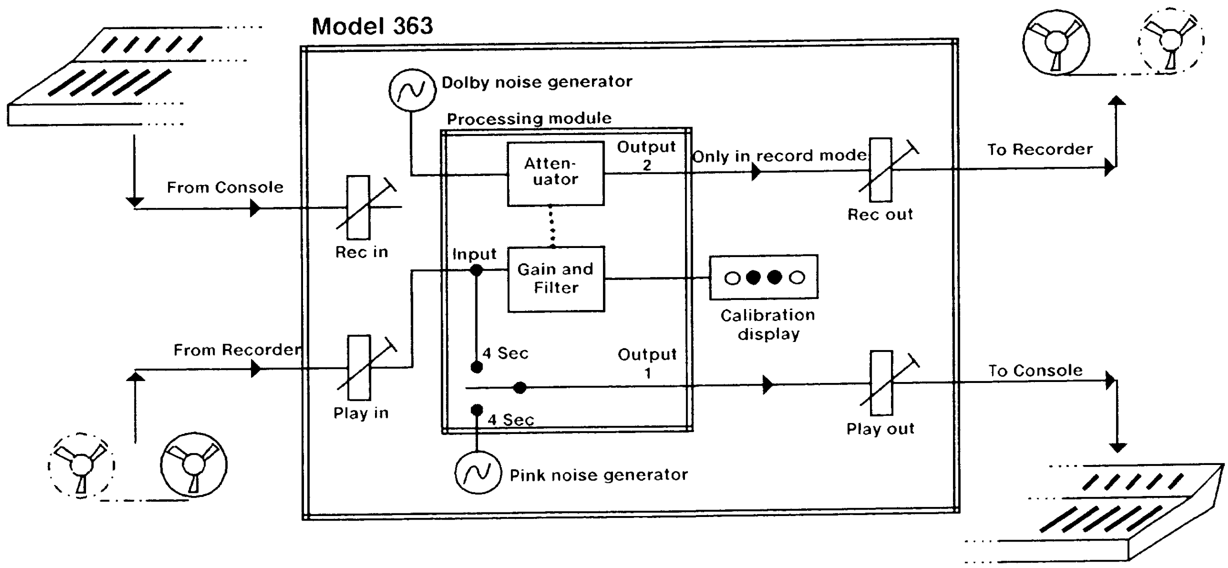
AND ...... SR processing is selected;

AND ...... the "Set-up" button is pressed.

LEDs on the front of each module indicate the source of the noise being heard. (Remote indicating LEDs can also be connected; see section 2.) The red LED indicates the internal reference noise, and the green LED indicates Dolby noise from the tape recorder.

The Auto Compare circuitry may deliver spurious and unexpected noises if it receives signals other than Dolby noise (including silence); this is normal and harmless. To avoid such anomalies it is good operating practice when playing Dolby noise to find the appropriate point on the tape before you press the Set-up button, and to release the button as soon as you have finished with the Auto Compare sequence.

4.3.1 **When the SR/off/A switch is in the SR position,** the signal paths in Set-up are:



The calibration display will read the level returning from the tape recorder, and, if the Model 363 is switched to record, Dolby noise at low level will be the output to the recorder. The gain of the calibration display is increased so that Dolby noise at the correct level will give an Dolby level indication.

This mode is also used to align the Model 363 to Dolby noise played back from a previously recorded tape.

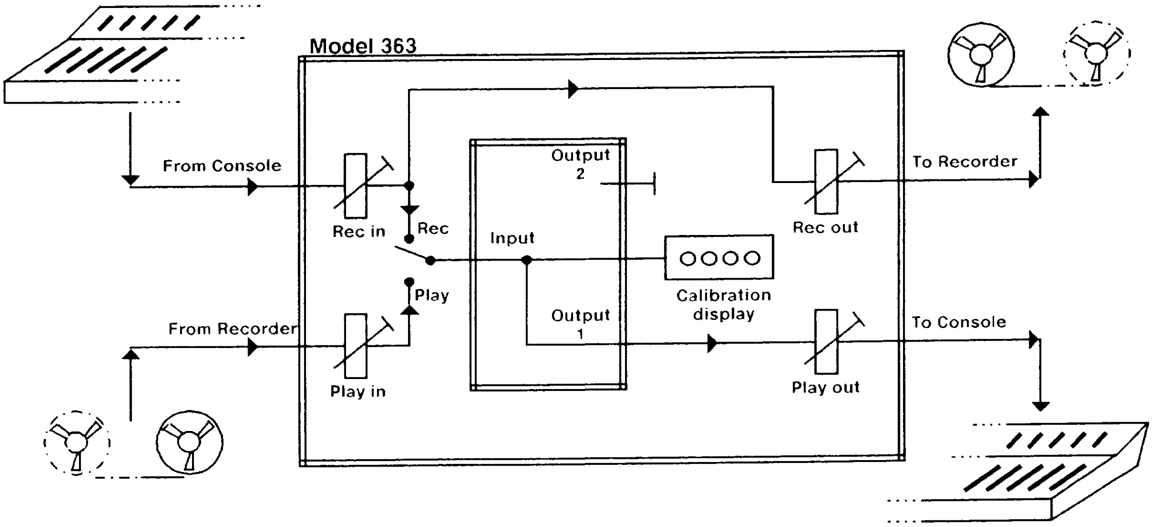
The Model 363 will start an auto-compare sequence when Dolby noise is detected at the "From recorder" input. Where both tracks in a two-channel situation are being replayed, first start the playback of the Dolby noise and then press the Set-up button; this ensures that the Auto Compare switching in the two channels is synchronized. However, we recommend listening to one track at a time.

4.4 Set-up

To help alignment of the Model 363, a Set-up button is provided. Pressing the Set-up button will set the unit to the correct alignment mode (which depends on the setting of the **SR/off/A** switch).

Initially, the Model 363 must be aligned to the normal signal levels present in the studio. To help this alignment, a special mode is selected if the **Set-up** button is pressed when noise reduction is in the **off** position.

4.4.1 **When the SR/off/A switch is in the off position,** the signal paths in Set-up are:



The calibration display may be switched between the record and play signal paths using the front panel **rec/play** switch. Any **remote rec/play** switching is disabled. This mode is used to align the Model 363 to standard studio line levels.

4.5 Alignment

During initial or routine alignment, it is simplest to send a tone at Dolby level from the console and align for unity gain through the system.

When aligning to a tape from another studio, it is necessary to take the Dolby noise or Dolby tone recorded on the tape as the starting point, and set up the unity gain structure from there. For this reason there are two alignment procedures set out here; the first sets Dolby level to a tone from the console (Console Level Reference), the second to Dolby noise or tone recorded on a tape from another studio (Tape Level Reference).

These instructions assume that the Model 363 record/play switching is being controlled manually at the front panel. Should the record/play switching be controlled automatically by the tape recorder, the manual switches must be used when both **"Set-up"** and **"off"** have been selected, since under these conditions the remote record/play switching is purposely disabled. (The various controls are shown in Figure 4.1 on page 4.11) .

The two procedures start on the following page.

CONSOLE LEVEL REFERENCE Routine in-house alignment

A: INITIAL STEPS

The following alignment steps involve sending 1 kHz tone from the console at Dolby level. In many studios this will be the normal studio line level (for example, 0 VU).

If you are not sure what your Dolby level is, read section 2.1 for more information.

1. Remove the Model 363 from the signal path by pressing the **Bypass** button for each channel.

2. Align the recorder to “in-house” standards, and check that recorder and console meters agree, or bear a known fixed relationship.

3. Restore the Model 363 unit to the signal path by releasing the **Bypass** buttons and set the **SR/off/A** switches on each channel to the center **off** position.

B: RECORD ALIGNMENT

4. Press the **Set-up** button, and set the **rec/play** switches on the Model 363 to the **rec** position (red LEDs on).

5. Send a 1 kHz test tone at Dolby level (e.g., 0 VU) from the console to the Model 363.

6. Place the recorder in record, line in, or E-to-E.

7. Adjust the **rec in** trimmers for Dolby level on the calibration displays (equal brightness of the green LEDs).

8. Adjust the **rec out** trimmers on the Model 363 for the correct reading (e.g., 0 VU) on the tape recorder meters. As a quick check, you may switch the Model 363 in and out of **Bypass** at this point and make sure the tape recorder meters don’t change.

C: PLAYBACK ALIGNMENT

9. Switch console meters to read the return signal from the tape recorder.

10. Set the **rec/play** switches on the Model 363 to **play** (green LEDs on) and adjust the **play in** trimmers for Dolby level on the calibration displays (equal brightness of the green LEDs).

11. Adjust **play out** trimmers on the Model 363 for Dolby level (e.g., 0 VU) on the console meters. As a quick check, you may switch the Model 363 in and out of **Bypass** at this point and make sure the console meters don’t change.

12. Release the **Set-up** button, and make sure the unit is out of **Bypass**.

The alignment is now complete.

D: PREPARING FOR A RECORDING

13. Select **A** or **SR** on the individual 3-position switches as appropriate.

14. Record a section of Dolby tone or noise on tape by putting the recorder into record and pressing the **Set-up** button the the Model 363 unit. (Remember that Dolby noise is recorded on tape at – 15 dB below Dolby level (e.g., 0 VU), and so will read

low on console and recorder meters. However, this level difference is automatically compensated in the Model 363 calibration display when **Set-up** is selected.)

Under these conditions, when using Dolby SR, the Cat. No. 300 or the Cat. No. 350 modules will go into the **Auto Compare** mode. **Auto Compare provides the user with an accurate audible verification that both the tape recorder frequency response and the calibration levels are set properly regardless of the indication shown on the tape recorder meters and Dolby calibration displays.** Listen for level differences between the pink noise signal coming from the output of the tape and internal Dolby noise generator in the **Auto Compare** mode. The two LEDs on the front of the Cat. No. 300 and Cat. No. 350 modules indicate whether the monitors are receiving the internal reference signal (red LED) or the Dolby noise from the tape recorder (green LED). The alignment procedure is correct when the levels between the tape and reference levels audibly match.

Note: Dolby level on tape will now have a fixed flux level which may be compared with the magnetic reference tape used for the recorder alignment: e.g., "Dolby Level is +4 dB above 185 nWb/m."

15. It is possible that the level of Dolby noise or Dolby tone recorded on the tape in this last step will produce a slightly different reading on the calibration display from that observed in step (10) above. This is due to the tolerance of many parts of the complete audio chain, the accuracy to which the recorder calibration adjustments can be made (particularly if automatic setting of these controls is used), and the agreement between meters or displays in the console, recorder, and the Model 363. If a small difference is observed (as evidenced by the two green LEDs on the Model 363 calibration display not being equally bright), we recommend that the **rec out** trimmers on the Model 363 be slightly adjusted to compensate for this build-up of tolerances.

TAPE LEVEL REFERENCE Aligning to Tapes Recorded in Other Studios

A: INITIAL STEPS

1. Remove the Model 363 from the signal path by pressing the **Bypass** button for each channel.

2. Align the recorder using the tones on the incoming tape, and check that recorder and console meters agree, or bear a known fixed relationship. If you also need to record on the tape, be sure to align the record section of the recorder as well.

3. Restore the Model 363 unit to the signal path by releasing the **Bypass** buttons.

B: PLAYBACK ALIGNMENT

4. Set the **rec/play** toggle switches on each channel to the **play** position (green LEDs on).

5. Set the **SR/off/A** toggle switches to **SR** or **A** as appropriate for the incoming tape. Replay the Dolby noise (SR) or Dolby tone (A-type) from the tape, and then press the **Set-up** button.

   When using Dolby SR, the Cat. No. 300 or the Cat. No. 350 modules will go into the **Auto Compare** mode. **Auto Compare provides the user with an accurate audible verification that the playback frequency response and decode calibration levels are set properly regardless of the indication shown on the calibration displays.** Listen for level differences between the pink noise signal coming from the tape and the internal Dolby noise generator in the **Auto Compare** mode. The two LEDs on the front of the Cat. No. 300 and Cat. No. 350 modules indicate whether the monitors are receiving the internal reference noise signal (red LED) or the Dolby noise from the tape recorder (green LED).

6. Adjust the **play in** trimmer on each channel of the Model 363 unit until the tape and reference levels audibly match in the **Auto Compare** mode or for equal brightness of the green LEDs.

7. Release the **Set-up** button and set the **SR/off/A** toggle switches to the **off** position.

8. Switch the console meters to read the return signal from the tape recorder.

9. Replay the 1 kHz alignment tone from the incoming tape and adjust the **play out** trimmers on the Model 363 for the correct reading (e.g., 0 VU) on the console meters. As a quick check, you may switch the Model 363 in and out of **Bypass** at this point and make sure the console meters don't change.

10. Reset the **SR/off/A** switches to the proper NR type for the incoming tape.

For playback only purposes the alignment is now complete. Should you be required to drop into record (overdub), continue with the following steps.

C. RECORD ALIGNMENT

11. Set the **rec/play** toggle switches on each channel to the **rec** position (red LEDs on).

12. Set the **SR/off/A** to **SR** or **A**, for the NR type being used.

13. Place the recorder in record, line in or E-to-E on an appropriate section of blank tape. Press the **Set-up** button. Dolby noise (SR) or Dolby tone (A-type) is now being sent to the tape recorder.

MAIN

14. Adjust the **rec out** trimmer on each channel on the Model 363 for equal brightness of the green LEDs.

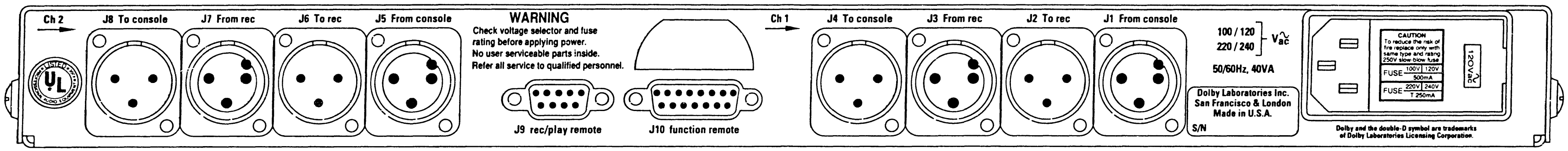
15. Set the **SR/off/A** switches to the **off** position.

16. Send a 1 kHz test tone at a convenient level (e.g., 0 VU) from the console to the Model 363.

17. Adjust the **rec in** trimmers on the Model 363 for the correct reading (e.g., 0 VU) on the tape recorder meters. As a quick check, you may switch the Model 363 in and out of **Bypass** at this point and make sure the tape recorder meters don't change.

18. Release the **Set-up** button and set the **SR/off/A** switches to the proper NR type for the recording.

The alignment is now complete.



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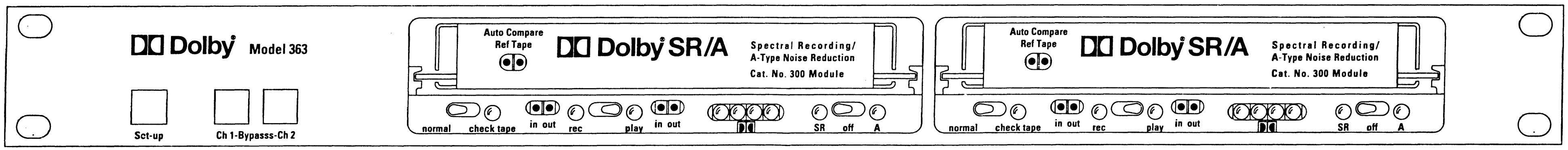


A2A3802 REV 2

Figure 4.1 Front and rear details

SECTION 5
PRINCIPLES OF NOISE REDUCTION AND SPECTRAL RECORDING

5.1 Introduction

The Model 363 may contain Dolby A-type noise reduction (using the Cat. No. 450 module), Dolby Spectral Recording (Cat. No. 350), or both (Cat. No. 300). There are two other analog Dolby systems, B-type and C-type. All employ some principles in common: for completeness, the following contains brief descriptions of all. More detailed descriptions of A-type and SR can be found in the appendices in Section 8.

5.2 General

In sound recording or transmission the higher audio frequencies are often pre-emphasized to improve the signal to noise ratio (see Figure 5.1). However the equalization characteristic must be chosen so that even in the worst cases there are no detrimental effects: material rich in high frequencies must not cause distortion. Therefore the allowable boost with fixed equalization is limited and the degree of noise reduction is modest.

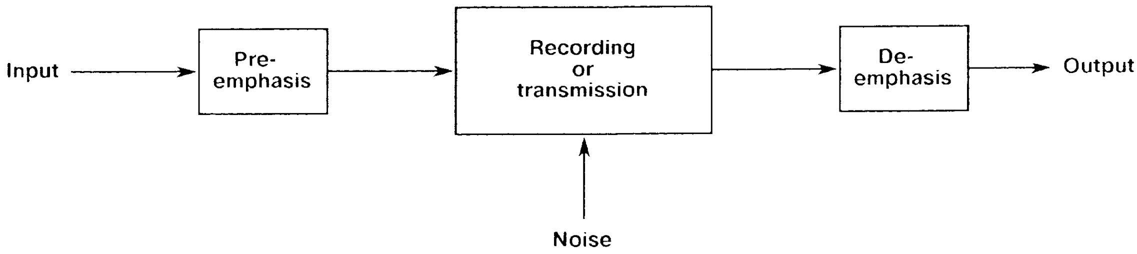


Figure 5.1 Fixed pre- and de-emphasis

Systems which improve the signal to noise ratio by compression in the encoding mode followed by expansion in subsequent decoding are known generally as compandors. Figure 5.2 shows the block diagram of a typical system.

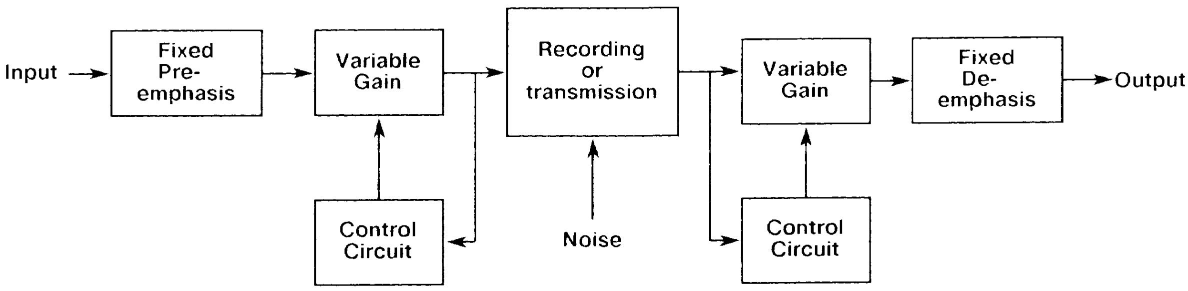


Figure 5.2 Typical compandor

The variable gain blocks change gain under the control of the signal level, most commonly with a straight-line relationship when the compressor output is plotted against its input using decibel scales; such a system is known as a constant slope compandor. The noise level at the output of the expander rises as the signal level rises. A loud signal in one area of the spectrum will mask noise in that same area, but the variation in noise in other areas may be audible. Compandors of this type are inherently susceptible to this audible variation in noise level, called noise modulation.

Noise modulation can only be eliminated by ensuring that the gains of the encoder and decoder in any particular part of the spectrum are fixed except when signals are present in that part of the spectrum at levels above a defined threshold. The output noise will then be constant in all parts of the spectrum except those where signals are present to mask the changes in noise.

A system with this property must be capable of changing gain in any one area of the spectrum without changing at others. Clearly in a conventional compandor (such as in Figure 5.2), any change in gain occurs at all frequencies equally, so that a signal at any one frequency must inevitably alter the noise level at other (unmasked) frequencies.

Constant slope compandors have no upper or lower thresholds, and hence possess the virtue that there is no need to ensure that the absolute signal levels in the compressor and expander are equal. A superficially attractive idea is to use several constant slope compressors each fed by a separate band splitting filter. Unfortunately because of the practical limits on filter slopes and the absence of compressor thresholds, each band compressor receives signals (albeit attenuated) from the other bands and responds with gain changes. It can be shown that a change in input signal at any particular frequency that causes x dB gain change in one band causes exactly x dB change in all the other bands. Thus constant slope split band compandors also inevitably lead to audible noise modulation.

The solution is to employ a defined low-level input threshold below which the frequency dependent gain or loss of the processor is constant. Together with appropriate frequency response adaptation it is then possible for the processor to keep its sub-threshold gain or loss except in those areas of the spectrum where high level signals mask the noise. If the gain or loss in unmasked regions of the spectrum is constant, then there can be no noise modulation. Such a response adaptation and low-level threshold are features of all the Dolby systems.

If the level of an input signal at a particular frequency increases abruptly, an encoder must reduce its gain in response to that new level. This gain reduction occurs over a finite time during which the encoder output level will be excessive: this excess level is known as overshoot. In general the magnitude of an overshoot corresponds to the degree of gain reduction and its duration to the response ("attack") time. Provided the overshoot does not lead to overloading of the recording or transmission system, it is harmless. However for high level signals overshoot can cause transient distortion and non-complementary behavior in the decoder. (Note in passing that it is possible to use instantaneous or near-instantaneous attack to eliminate overshoot, but such an approach can be disastrous subjectively because of wide-band modulation products.)

The Dolby systems use a dual-path configuration in which the input signal passes directly from input to output; the processing consists of the addition or subtraction of a further signal whose maximum amplitude is small compared with the maximum amplitude of the input. This method imposes an upper threshold above which gain reduction ceases (see Figures 5.3 and 5.4): note the fixed gains at low and high levels. This shape of characteristic permits overshoot suppression within the further path. The result is that overshoots resulting from high level signals are much smaller than the degree of gain reduction, and there is little danger of transient overload of the recording or transmission system.

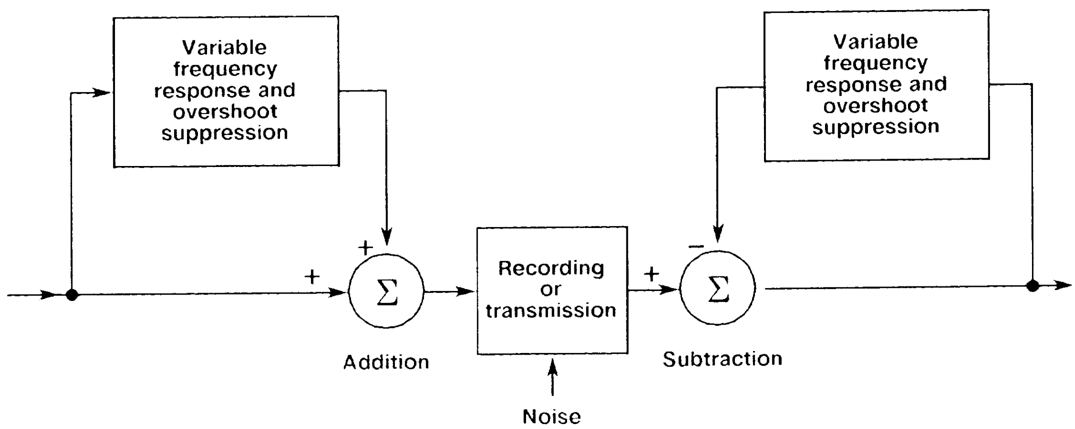


Figure 5.3 Dual-path configuration of all Dolby systems

Figure 5.4 General form of compression and expansion characteristic of all Dolby systems

5.3 Dolby A-type noise reduction

A-type noise reduction (nr) is a professional system introduced in 1966, and manufactured only by Dolby Laboratories. It provides 10 dB of nr over most of the audio spectrum rising to 15 dB at very high frequencies. The requirement for variable frequency response is met by dividing the frequency range into four bands, each with its individual compressor (see Figure 5.5). A high-level signal in one band does not affect the other bands, where noise may not be masked, and hence in general the system does not give audible noise modulation.

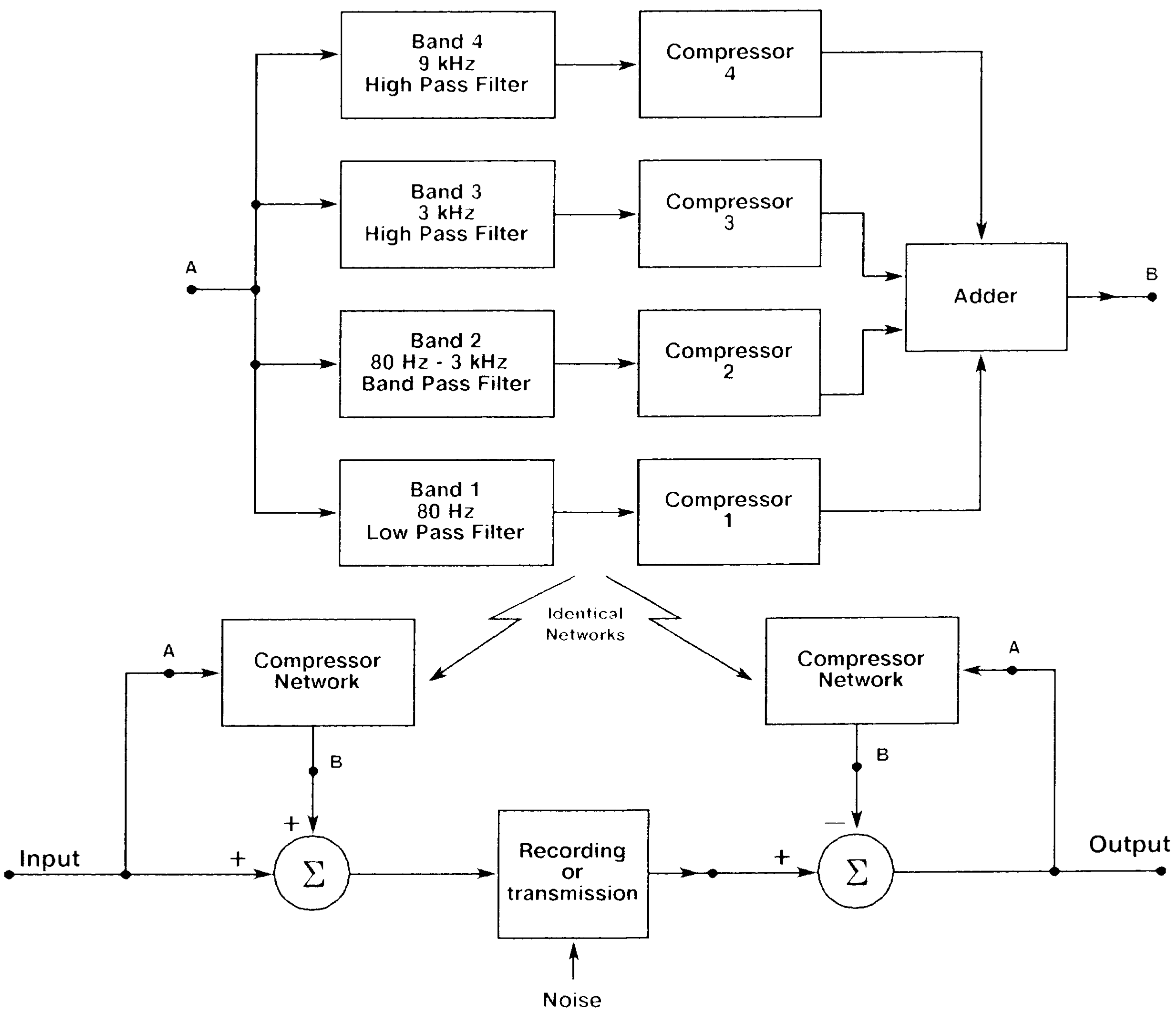


Figure 5.5 Block diagram of A-type noise reduction

A-type nr is in widespread use throughout the recording, broadcast and film industries. It is an essential ingredient in the film release format known as Dolby Stereo.

MAIN

5.4 Dolby B-type noise reduction

B-type nr is a consumer system intended primarily for use with low-speed tape, especially the Philips compact cassette. It was first introduced in 1969. It reduces tape hiss by 10 dB. Unlike A-type nr, it uses only one frequency band: instead of providing variable gain within a fixed range of frequencies it provides fixed gain (or cut in the decoder) within a band of variable width. It can be considered as a high frequency emphasis of fixed magnitude whose start and stop frequencies slide upwards along the frequency axis so as not to boost the dominant, high level spectral components of the input while providing a fixed 10 dB of nr at frequencies above those dominant components (see Figures 5.6 and 5.7). At any one frequency the output input characteristic of the encoder displays gentle compression, permitting complementary expansion in the decoder. The fixed magnitude ensures that noise not masked by the input signal has a fixed level, and therefore no noise modulation is perceived.

The vast majority of B-type circuits are built under license from Dolby Laboratories Licensing Corporation by over 250 world-wide licensees who include all the major manufacturers of consumer tape recorders. Dolby Laboratories manufactures small numbers of professional B-type processors for use in the preparation of pre-recorded tapes (audio cassettes and VHS video cassettes).

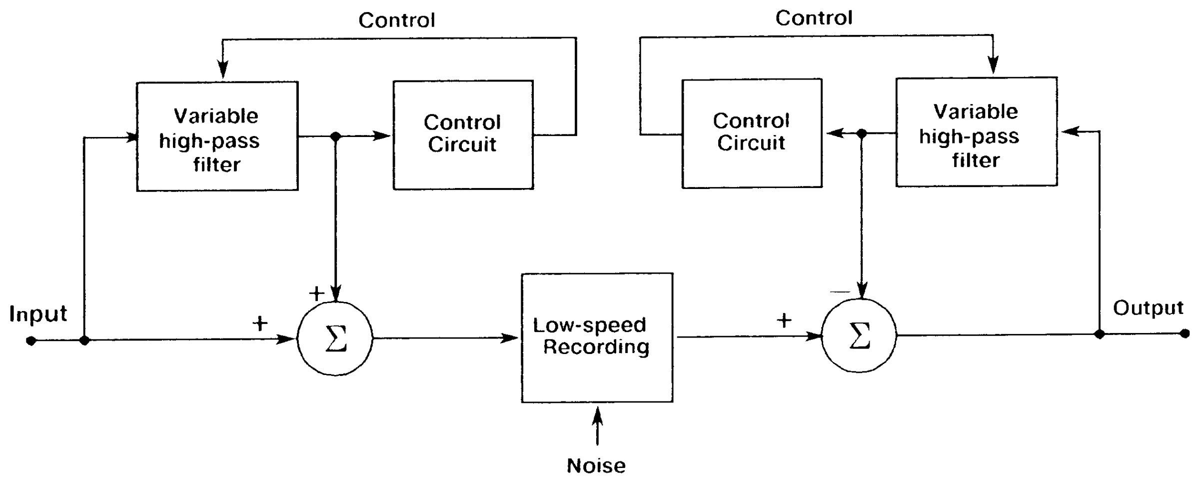


Figure 5.6 Block diagram of B-type noise reduction

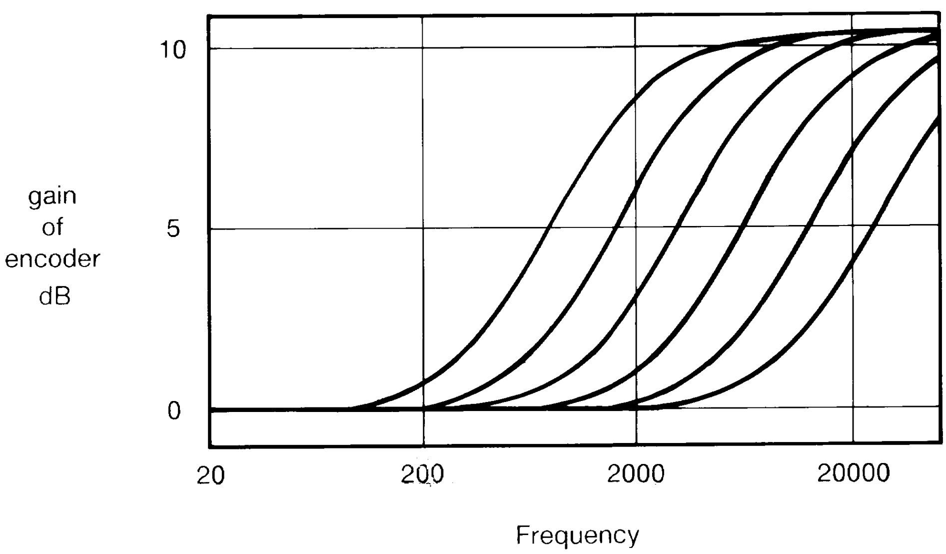


Figure 5.7 Family of response curves for B-type noise reduction

MAIN

5.5 Dolby C-type noise reduction

C-type nr was introduced in 1980 and is used in consumer audio cassette recorders and in the audio channels of professional Betacam*, MII*, and U-matic SP* video recorders. It operates in a manner similar to B-type, but offers 20 dB of nr. It achieves the steeper filter slopes required to give adequate nr at high frequencies in the presence lower frequency dominant signals by employing two overlapping processor stages in tandem, operating with offset ("staggered") thresholds and with an action extending two octaves lower than B-type (see Figures 5.8 and 5.9).

Frequency shaping ("spectral skewing") at the input of the encoder desensitizes the processor to the effects of high frequency response errors. Additional shaping in the main path ("anti-saturation") lowers the amplitude of high frequency high level signals before they are applied to the tape, reducing high frequency distortion and self-erasure.

The decoder contains complementary circuits to restore the frequency response: the amount of nr at the highest frequencies is decreased, but this is where the ear is least sensitive to noise.

Virtually all C-type circuits are built under license.

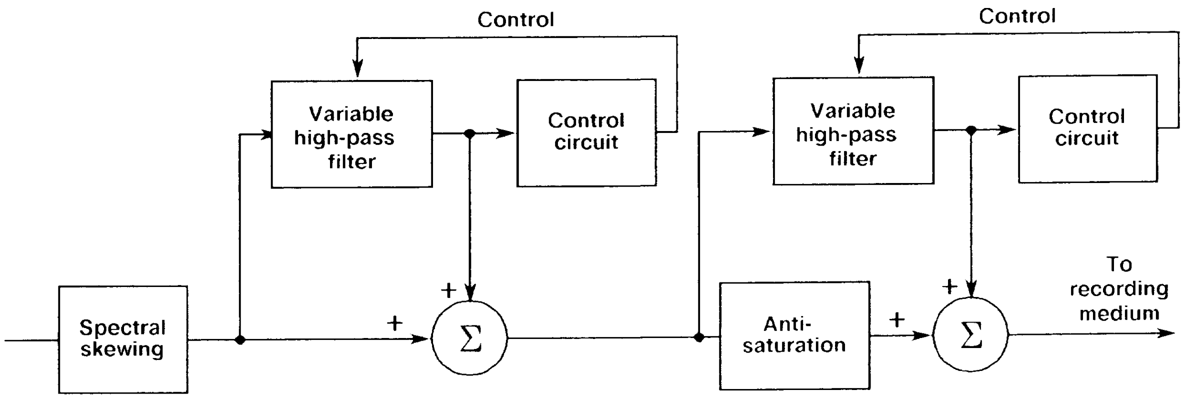


Figure 5.8 Block diagram of C-type encoder

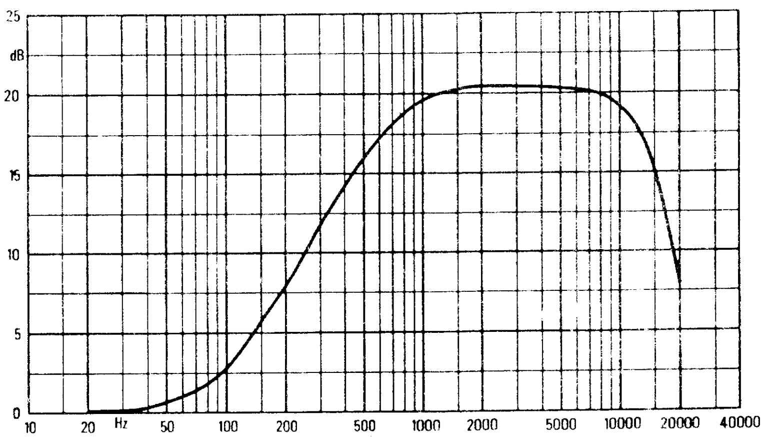


Figure 5.9 Low-level C-type encoder response

* Betacam and U-matic are trademarks of Sony Corporation, and MII is a trademark of Matsushita Electric Industrial Co. Ltd.

5.6 Dolby Spectral Recording

Introduced in 1986, Dolby Spectral Recording (SR) is a professional system which combines all the advantages of fixed bands (as in A-type) with the spectral adaptation of sliding bands (as in B-type and C-type). It also employs spectral skewing and anti-saturation, but applied at low frequencies as well as high.

SR uses three high frequency and two low frequency stages in tandem, with a crossover at 800 Hz (see Figure 5.10). Together they result in a reduction in audible noise of 24 dB (see Figure 5.11), taking into account the frequency dependent sensitivity of the ear.

The SR processor response adapts to the input spectrum to obtain full unchanged boost except in the immediate neighborhood of dominant frequencies. In this way, the system reduces not only noise but other unwanted signals such as tape modulation noise and distortion products introduced between the encoder and decoder.

The spectral skewing and multi-stage processing give greater tolerance to level and frequency response errors compared with A-type nr.

Dolby SR circuits are built exclusively by Dolby Laboratories.

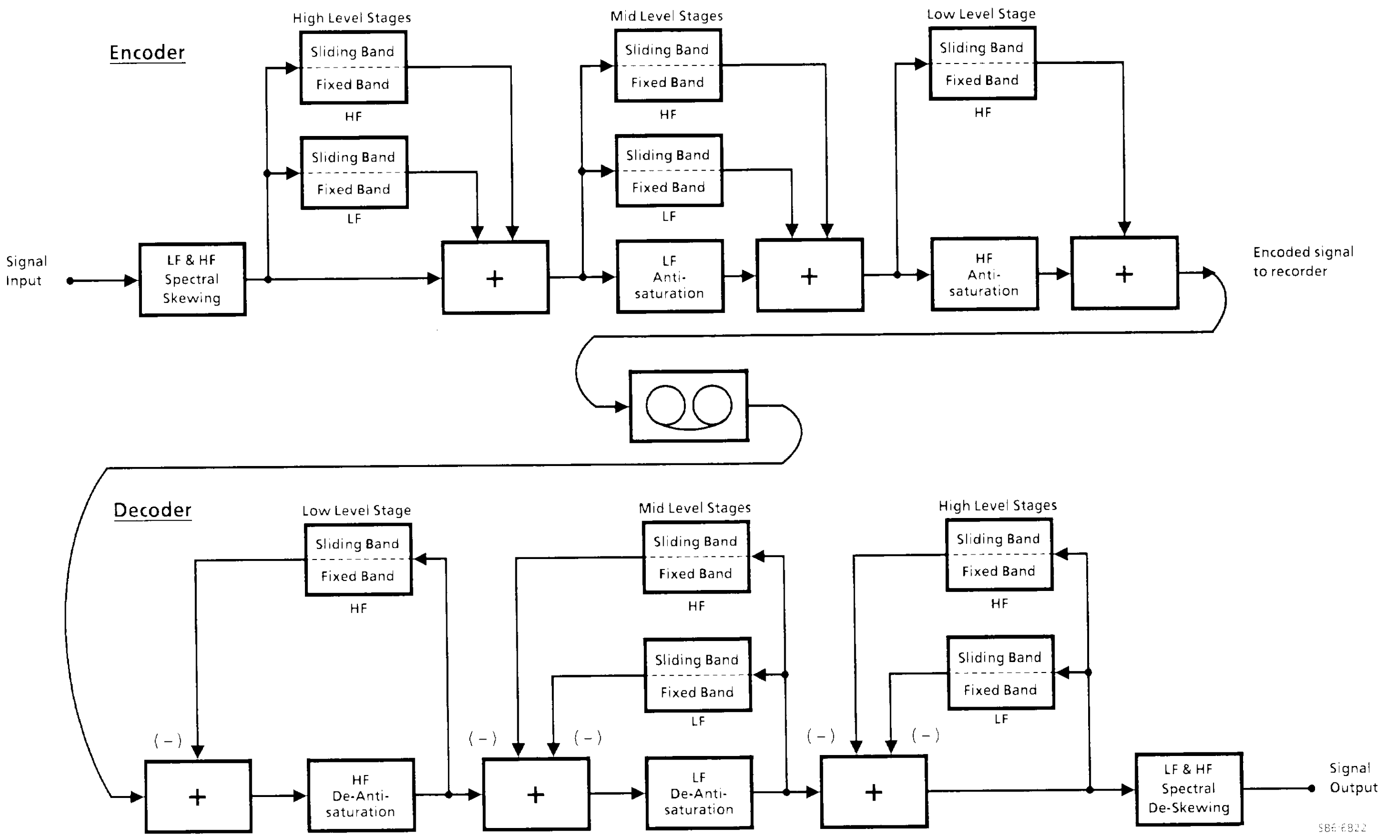


Figure 5.10 Block Diagram of Dolby SR

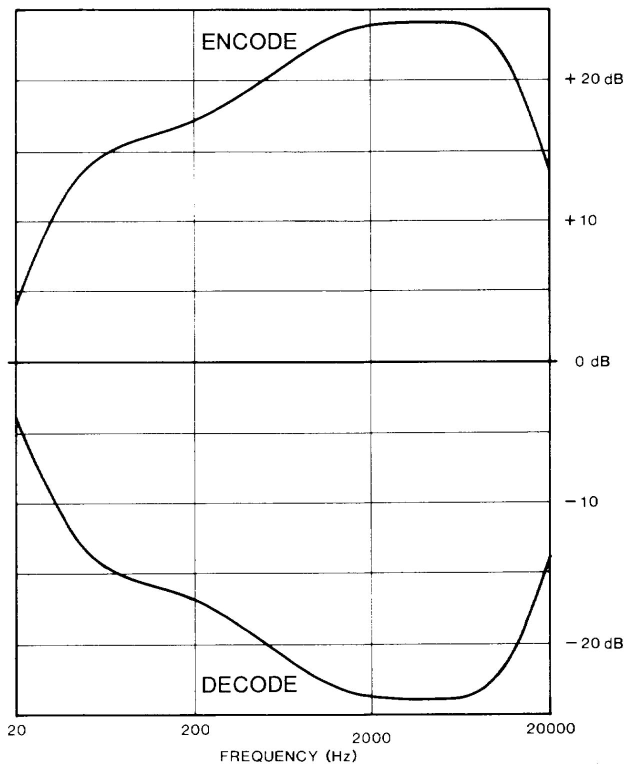


Figure 5.11 Low level response of Dolby SR processor

SECTION 6
CIRCUIT DETAILS

6.1 General

The unit is functionally split into various parts. Each audio channel comprises input and output stages together with various switches (monitoring, mode selection, etc.), and a separate signal processing module. In addition, there is a common power supply and calibration generator section. Apart from the modules (and the power line transformer), all the circuits are contained on one printed circuit board (called the motherboard), which may be removed for servicing.

The circuits on this main board are relatively simple in operation; in the event of any failure, it would be possible to repair them in the field using identical components as replacements. On the other hand, the modules are extremely complicated, and contain large numbers of components specially selected during the manufacturing process. If there is a module failure, the module must be returned to Dolby Laboratories or its agents for repair.

More notes on servicing are found later in Section 7.

6.2 Motherboard Audio Circuits

Refer to the block diagram drawing Figure 6.1, and detailed circuit drawings Figure 6.2 (for the channel circuits) and Figure 6.3 (for the common circuits). All diagrams are at rear of this section, following page 6.4.

Input signals from the console and the tape recorder enter the unit via separate XLR connectors, and pass to identical balanced input amplifiers. Some component values are 0.1% tolerance in this section to provide for excellent performance figures without the need to adjust values during manufacture. Each input amplifier has a 10-turn potentiometer at its output to allow calibration of input levels to that required in the Dolby signal processing modules.

A hard bypass is provided by relays at the outputs, which connect these outputs directly to the appropriate inputs when either the bypass function is selected or the unit is not powered. Note that on powering the unit, there is an initial one second delay (provided by IC905B and IC1305B) in the operation of these relays to prevent any thumps or clicks reaching the monitoring circuits.

Each input amplifier has two FETs across the input line; under normal operation, these are held off (by a −30 V supply generated from the Dolby tone oscillator circuit, IC 1202c). When the unit is not powered, they act as a short-circuit to prevent the non-linear distortion which would arise if large signals are applied to the input operational amplifiers in their unpowered state. In this mode, the input impedance drops to about 10 kohm.

The output amplifiers are also balanced, and provide the same output level even if one side is grounded (ie in unbalanced use) — although of course the maximum output level will be halved. Separate 10-turn potentiometers are provided for each output.

6.3 Monitoring and Mode Switching

The operation of the various switches is best understood from the block diagram, where the switches are drawn in such a manner as to emphasis their function (rather than their circuit). The simpler block diagrams in Section 3 are also useful as they show the signal paths individually for each mode of the Model 363. The actual realization of these switches is a combination of logic circuits and FET switches, and is shown on the detailed circuit diagram. Examination of one switch will allow the remainder to be understood.

The switching of the signal to the **To rec** output is a good example. Under normal operation, the **To rec** output is fed from the processing module **Output 2** signal in the record mode, and is muted in the play mode. This prevents a positive feedback loop being formed in the recorder under certain combination of Model 363 and tape recorder monitor switch settings.

In the play mode, another path is made available **if** the Model 363 is in the **Set-up** mode **and** the signal processing has been turned **off**. With these conditions met, there is a calibrated path from the **From console** input to the **To recorder** output, which allows a simple calibration procedure using console tones to be used (see Sections 3 and 4).

The switching for the **To console** output is inside the processing module, and will be described later.

If the unit is being used with a three-head tape recorder, it is possible to monitor the reproduced signal in either the record or play modes. The **check tape** switches operate the play bypass relays (K401, K801) connecting the **To console** outputs directly to the **From recorder** inputs. Note that if SR or A-type processing is being used, the signal heard will be an encoded signal, and so cannot be used for critical comparison with the input signal, but only as a simple check of the recording process.

6.5 Metering

Each module provides a metering signal of 388 mV (−6 dBu) when the internal signal is at Dolby calibration level. This signal is applied to a comparator circuit which at Dolby level gives equal illumination of the two green LEDs. As the signal moves away from this level, one of these LEDs fades down while the other gets a little brighter; eventually only the one green LED is illuminated. As the signal continues to change, this green LED in turn cross-fades into a red. The "low" red LED goes out when the signal level is about 11 dB below Dolby level. Exact levels and cross-fade appearance are given on the detailed circuit diagram.

6.6 Logic

The logic circuits accept simple control voltages from the internal and external control switches, and apply logical rules to provide appropriate control signals to the various FET switches (note that in addition there are logic circuits in the processing modules which also enter into the circuit functional design).

Signals from the modules are also used to indicate which module is in place, and to prevent false indications being given. For example, if a Cat. No. 350 module is being used, which only has SR capability, there will be no ground at pin 32 of the module. This information is sensed (by IC905c) so that if A-type is selected by the front panel switch, the A-type light will not be illuminated, and no processing will be selected in the module.

6.7 Calibration Generators

Circuits provide Dolby tone, Dolby noise and pink noise for use in the calibration of the Model 363 to other equipment and signal routing systems. The level of these signals is 388 mV (note that the value read for pink and Dolby noise is critically dependent on the type of meter used).

Dolby tone (used for calibration and identification of A-type recordings) is generated by applying a square wave to the input of a filter. IC1204c switches between precisely defined +6 V and −6 V supplies; the switching rate is 850 Hz. IC1202a/Q1204 modify this frequency to 935 Hz for 30 ms every 750 ms, producing the characteristic warble sound; frequency (rather than amplitude) modulation is chosen so that meters do not indicate any level change during the time of the modulation.

Pink noise is obtained by taking the output of a white noise generator (IC1205) and passing it through a "pinking" filter.

The pink noise also passes to Q1203 which inserts a 20 ms nick every 2 seconds; this is Dolby noise.

6.8 Power Supplies

Incoming AC power passes through a voltage selector switch and to the transformer, giving nominal operating voltages of 100 V, 120 V, 220 V and 240 V. Full wave rectification is used; the transformer secondary is centre-tapped.

The power supply is conventional; separate positive and negative IC regulators are used to provide +15 V and −15 V, used for all circuits. In addition, a simple circuit supplies +22 V for the relays.

The supply to these relays is arranged to provide delays in both switch-on and switch-off to prevent large transients in the outputs. On switch-on, the bypass relays remain open (in the bypass mode) for about one second (provided by C902/R914 and C1302/R1314). On switch-off, the relay is opened when the ±15 V supplies fall to about ±7 V.

A −30 V supply is generated by a simple voltage doubler circuit fed with the square wave of the Dolby tone generator (IC1202c, D1204, etc.); this voltage is used to hold the input FETs (QF101/102, QF301/302, etc.) in the non-conducting state.

A 2-phase ac fan is fitted at the rear of the unit for cooling, and is supplied from the secondary winding of the power line transformer.

6.9 Signal Processing Modules

Refer to the block diagram Figure 6.4 and detailed circuit diagram Figure 6.5.

Three types of modules may be used in the Model 363. Cat. No. 450 provides A-type noise reduction, Cat. No. 350 gives SR processing, and Cat. No. 300 is a switchable module providing both SR and A-type. Modules may be interchanged with no adjustments; changes in the logic are automatically made with the circuits acting on identification signals provided by the modules.

6.9.1 Normal operation

Figure 6.4 is a block diagram of the Cat. No. 300, and serves as the basis for describing all the modules. (Note however that the Cat. No. 450, containing only A-type noise reduction circuits (Figure 6.6), is considerably simpler and in fact differs significantly in its block diagram. Nevertheless the description given below gives a good understanding of how it functions.) Audio signals (selected by circuits outside the module) pass through a bandpass filter to remove out-of-band components and via IC702 to the heart of the module. At this point, the level for Dolby level is exactly 388 mV. From here, the main path continues to the Output 2 pin.

The signal also passes to a side-chain which comprises the signal processing circuits. A switch at the output of these allows the output of the selected processor to continue to the mixing stages; in modules with only one type of processing, this switch is not present.

In the record mode, the side-chain signal is added to the main signal to form the encoded signal which passes to Output 2. In the play mode, the side-chain signal is subtracted from the main signal. The monitor output of the module is Output 1; in the record mode this is the input (unencoded) signal, and in the play mode it is the decoded signal; thus in all cases the signal at Output 1 is a normal signal.

6.9.2 Set-up

In Set-up, Dolby noise and Dolby tone calibration signals are present at the appropriate inputs of the module. The appropriate calibration signal is selected by IC802a. (The Cat. No. 350 has no connection to the Dolby tone input on pin 18. In the Cat. No. 450, Dolby tone on pin 18 is connected directly to the normal/set-up switch, shown here as IC803a, and there is no connection to the Dolby noise input on pin 20.) Switch IC803a sends the selected calibration signal to Output 2, and then to the **To recorder** output of the unit. Note that while the A-type Dolby tone is recorded at normal level (at or near 0 VU), the SR Dolby noise is reached about 15 dB below to avoid any tape saturation. Circuits in the meter amplifier automatically compensate for this low level and raise the signal to the same level as Dolby tone.

In both record and playback modes the signal from the **From recorder** input socket is selected by switches in the frame and enters the module at pin 26.

When A-type processing is selected this input signal is sent to the monitor output via the normal position of IC803B and to the Meter output via the normal position of IC802B. The return signal from the tape, normally Dolby tone in this case, can therefore be heard and displayed on the calibration display.

When SR is selected, the input signal is normally Dolby noise. It is therefore fed to the meter output via a bandpass filter and amplifier selected by IC802B, allowing an accurate and steady display. Output 1 receives the Dolby noise and reference pink noise alternately, with four seconds of each selected by IC803B.

This Auto Compare check is an extremely critical check for small errors in gain or frequency response of the overall record/playback process. Identification of the two signals is simple, as the pink noise is uninterrupted while the tape Dolby noise has the characteristic "nicks" occurring every 2 seconds. In addition, there are indicators on the front of the module to help identification; these indicators can be duplicated at a remote location through a D-connector on the rear of the Model 363.

The sync output from the module (pin 16) is not used in the Model 363; its purpose is to allow synchronization of a master Dolby noise generator in multi-track applications of these modules.

6.9.3 Power supplies

The modules contain their own regulators, and provide stabilized outputs of +9 V and −9 V for the module circuits.

MAIN

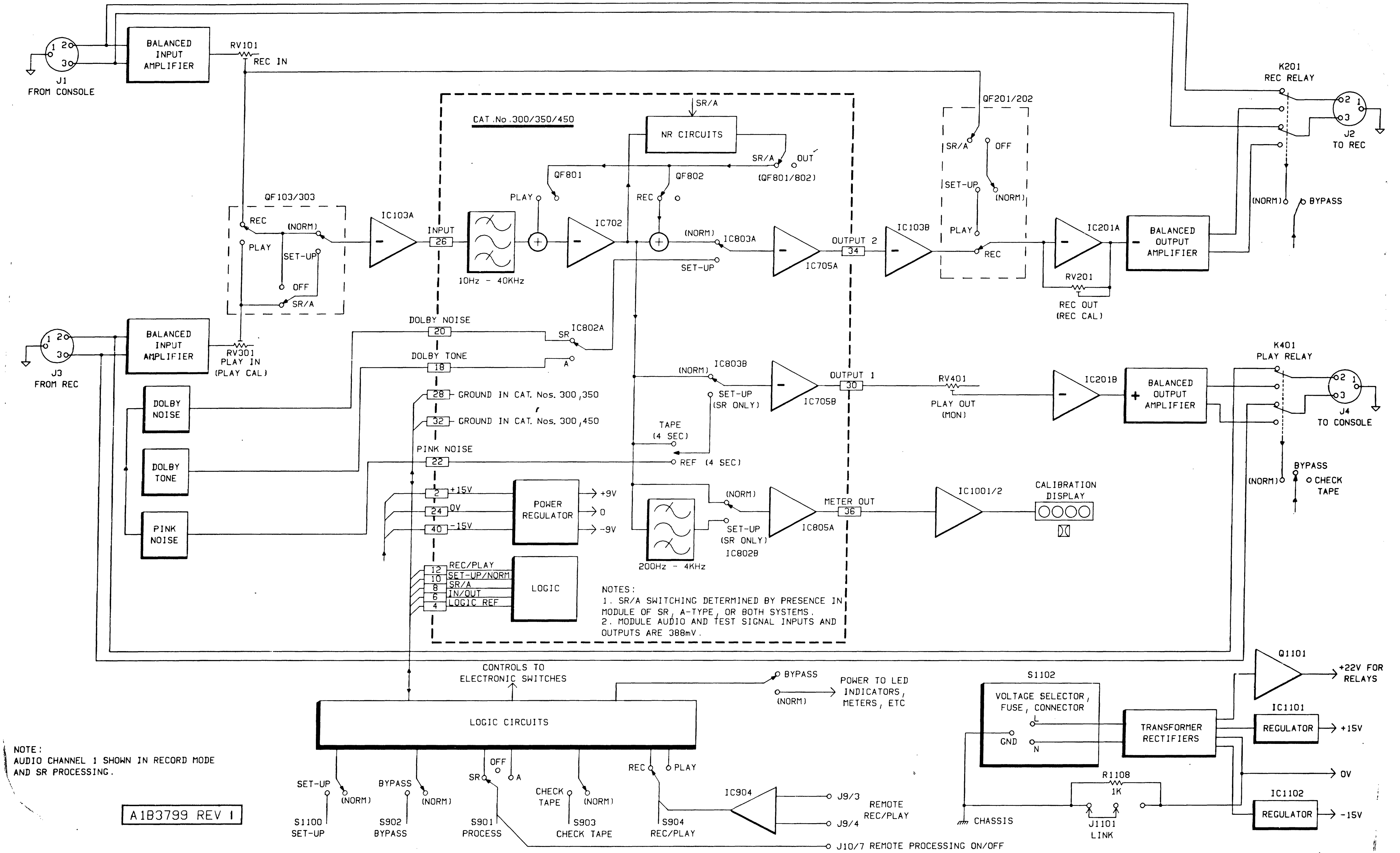
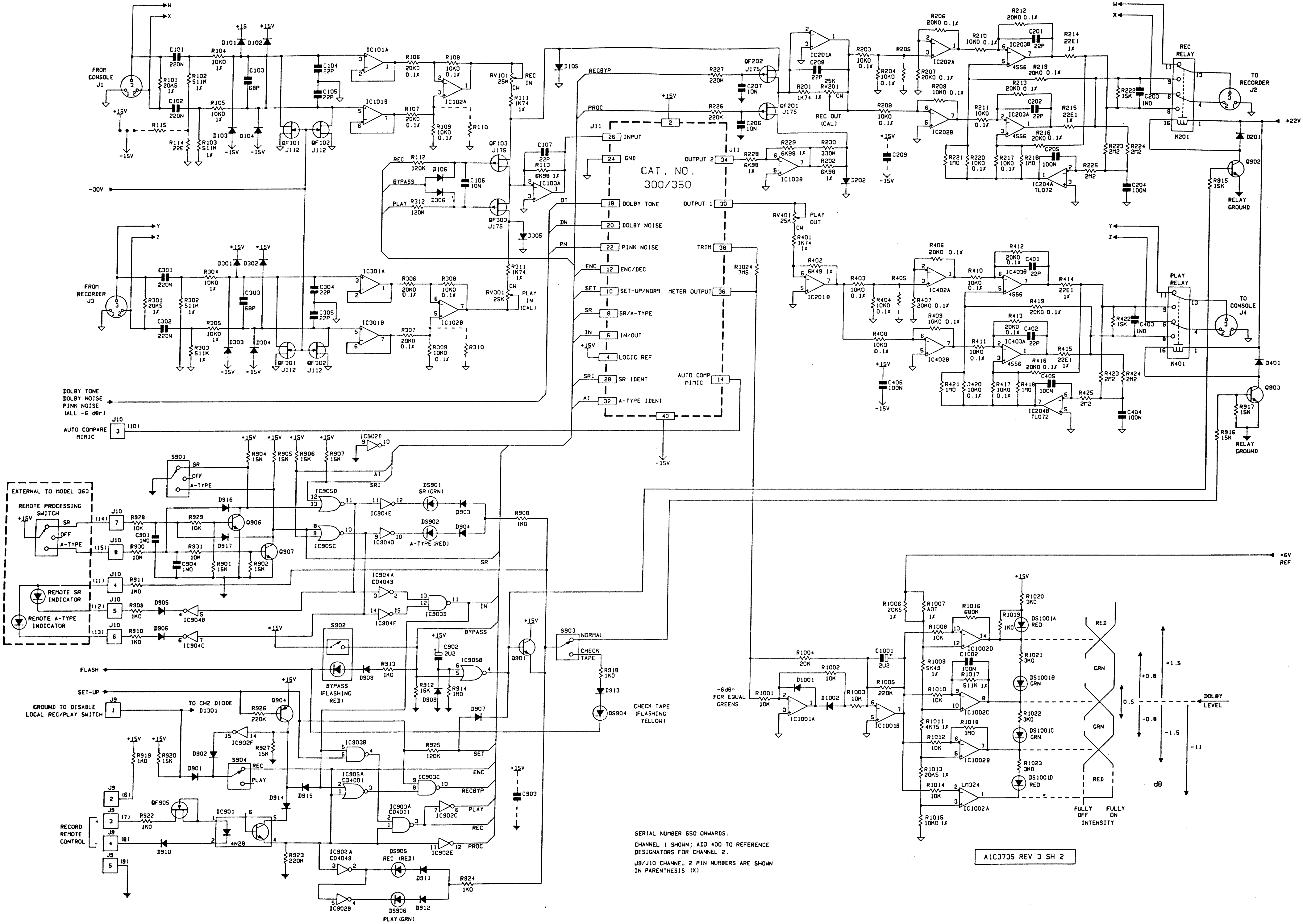
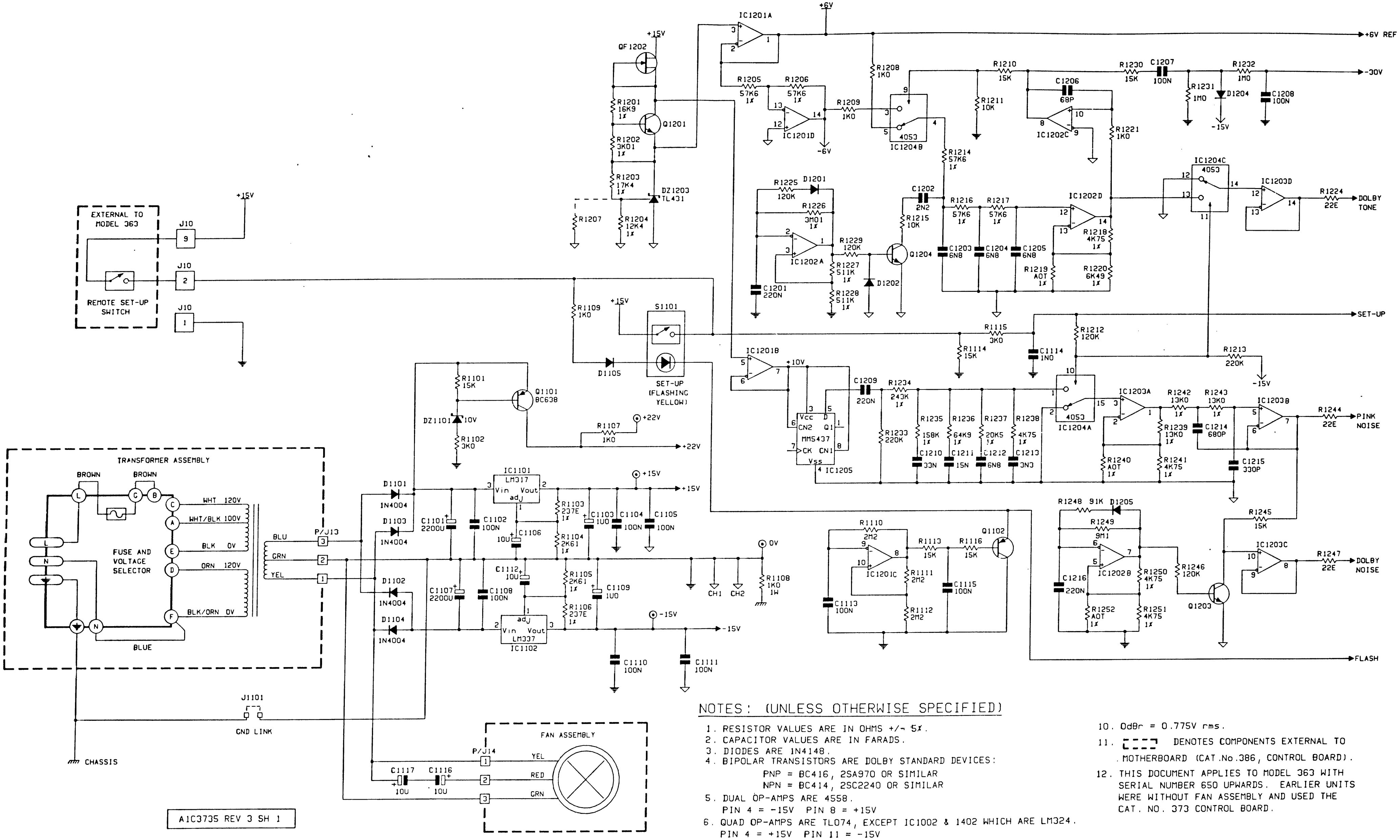


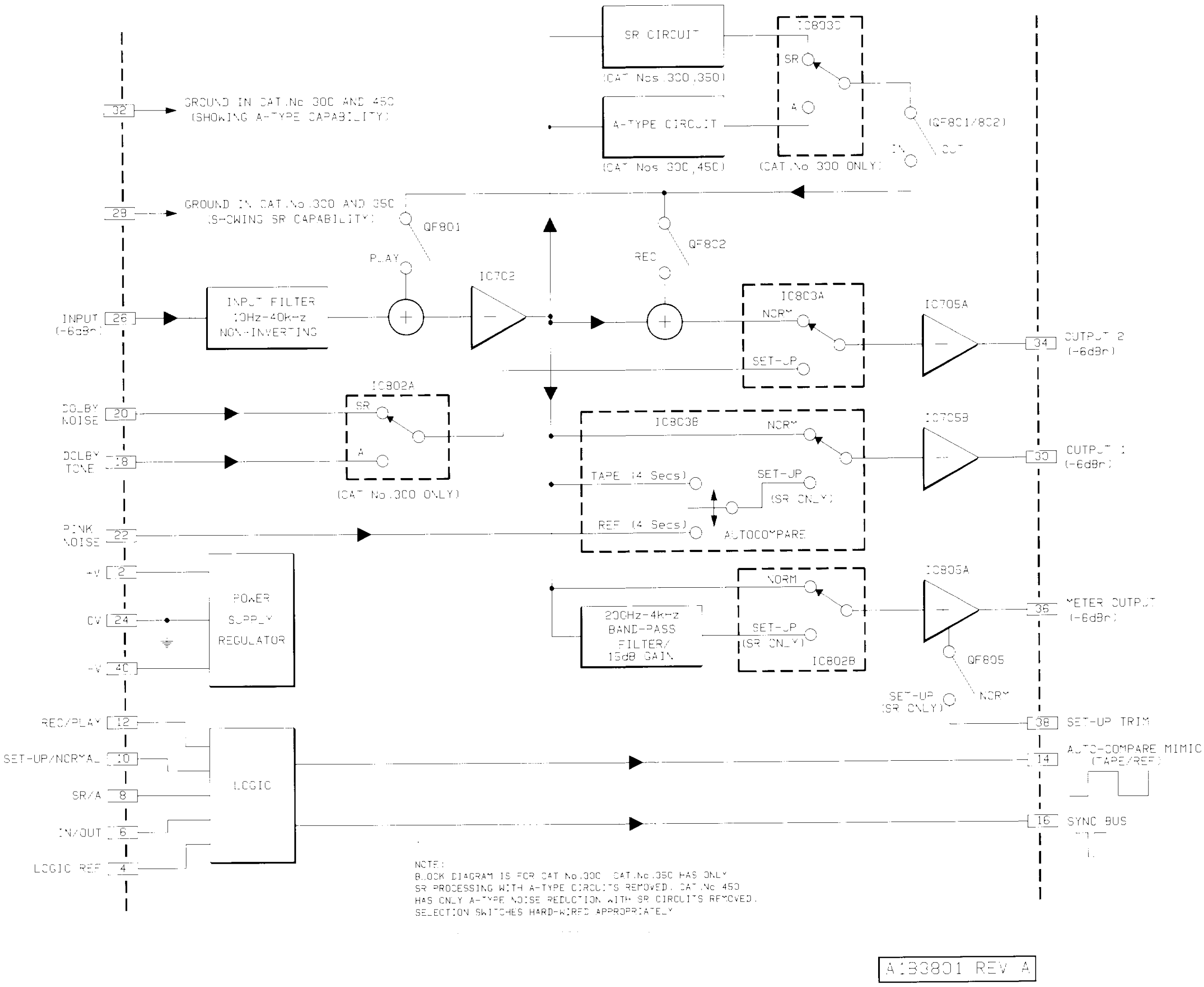
Figure 6.1 Model 363 block diagram


FROM CONSOLE
J1
TO RECORDER
J2
FROM RECORDER
J3
TO CONSOLE
J4
REC RELAY
K201
PLAY RELAY
K401
RELAY GROUND
+22V
-30V
CAT. NO.
300/350
J11
INPUT
GND
DOLBY TONE
DOLBY NOISE
PINK NOISE
ENC/DEC
SET-UP/NORM
SR/A-TYPE
IN/OUT
LOGIC REF
SR IDENT
A-TYPE IDENT
OUTPUT 2
OUTPUT 1
TRIM
METER OUTPUT
AUTO COMP MIMIC
REC OUT (CAL)
PLAY OUT
PLAY IN (CAL)
REC IN
RECBYP
PROC
REC
BYPASS
PLAY
DOLBY TONE
DOLBY NOISE
PINK NOISE
(ALL -6 dBr)
J10
AUTO COMPARE MIMIC
EXTERNAL TO MODEL 363
REMOTE PROCESSING SWITCH
SR
OFF
A-TYPE
REMOTE SR INDICATOR
REMOTE A-TYPE INDICATOR
S901
DS901 SR (GRN)
DS902 A-TYPE (RED)
S902
BYPASS (FLASHING RED)
S903
NORMAL
CHECK TAPE
DS904
CHECK TAPE (FLASHING YELLOW)
FLASH
SET-UP
GROUND TO DISABLE LOCAL REC/PLAY SWITCH
TO CH2 DIODE D1301
J9
S904
REC
PLAY
RECORD REMOTE CONTROL
4N28
DS905 REC (RED)
DS906 PLAY (GRN)
SET
ENC
RECBYP
PLAY
REC
PROC
+6V REF
-6dBr FOR EQUAL GREENS
DOLBY LEVEL
RED
GRN
GRN
RED
FULLY OFF
FULLY ON
INTENSITY
dB
+1.5
+0.8
0.5
-0.8
-1.5
-11
SERIAL NUMBER 650 ONWARDS.
CHANNEL 1 SHOWN; ADD 400 TO REFERENCE DESIGNATORS FOR CHANNEL 2.
J9/J10 CHANNEL 2 PIN NUMBERS ARE SHOWN IN PARENTHESIS (X).
A1C3735 REV 3 SH 2



NOTES: (UNLESS OTHERWISE SPECIFIED)

1. RESISTOR VALUES ARE IN OHMS +/- 5%.
2. CAPACITOR VALUES ARE IN FARADS.
3. DIODES ARE 1N4148.
4. BIPOLAR TRANSISTORS ARE DOLBY STANDARD DEVICES:
   PNP = BC416, 2SA970 OR SIMILAR
   NPN = BC414, 2SC2240 OR SIMILAR
5. DUAL OP-AMPS ARE 4558.
   PIN 4 = -15V PIN 8 = +15V
6. QUAD OP-AMPS ARE TL074, EXCEPT IC1002 & 1402 WHICH ARE LM324.
   PIN 4 = +15V PIN 11 = -15V
7. IC902, 904, 1302, 1304 ARE CD4049.
   PIN 1 = +15V PIN 8 = LOGIC GND
8. IC903, 1303 ARE CD4011. IC905, 1305 ARE CD4001.
   PIN 7 = LOGIC GND PIN 14 = +15V
9. IC1204 IS CD4053.
   PINS 6,7,8 = -6V PIN 16 = +6V
10. 0dBr = 0.775V rms.
11. DENOTES COMPONENTS EXTERNAL TO MOTHERBOARD (CAT.No.386, CONTROL BOARD).
12. THIS DOCUMENT APPLIES TO MODEL 363 WITH SERIAL NUMBER 650 UPWARDS. EARLIER UNITS WERE WITHOUT FAN ASSEMBLY AND USED THE CAT. NO. 373 CONTROL BOARD.



MAIN

Figure 6.4 Cat. No. 300 block diagram

NOTES: (UNLESS OTHERWISE SPECIFIED)

1. RESISTOR VALUES ARE IN OHMS
2. CAPACITOR VALUES ARE IN FARADS
3. IC's ARE LF442
4. P-CH FET's ARE J175, N-CH FET's ARE J210
5. DIODES ARE 1N4148
6. SINGLE OP AMPS TL070
   +9V TO PIN 7
   -9V TO PIN 4
7. DUAL OP AMPS TL062, TL072, LF442, NE5512, NE5532, LM358
   +9V TO PIN 8
   -9V TO PIN 4
8. CMOS SWITCHES ARE CD4053B (RCA ONLY) (STATIC SENSITIVE)
   SWITCHES SHOWN WITH LOGIC IN LOW STATE.
   +9V TO PIN 16
   -9V TO PIN 7
   GND TO PINS 6,8
9. [X] INDICATES CAT.No.300A EDGE CONNECTOR PIN.
   (JX-X) INDICATES INTERBOARD CONNECTORS J1,J2.

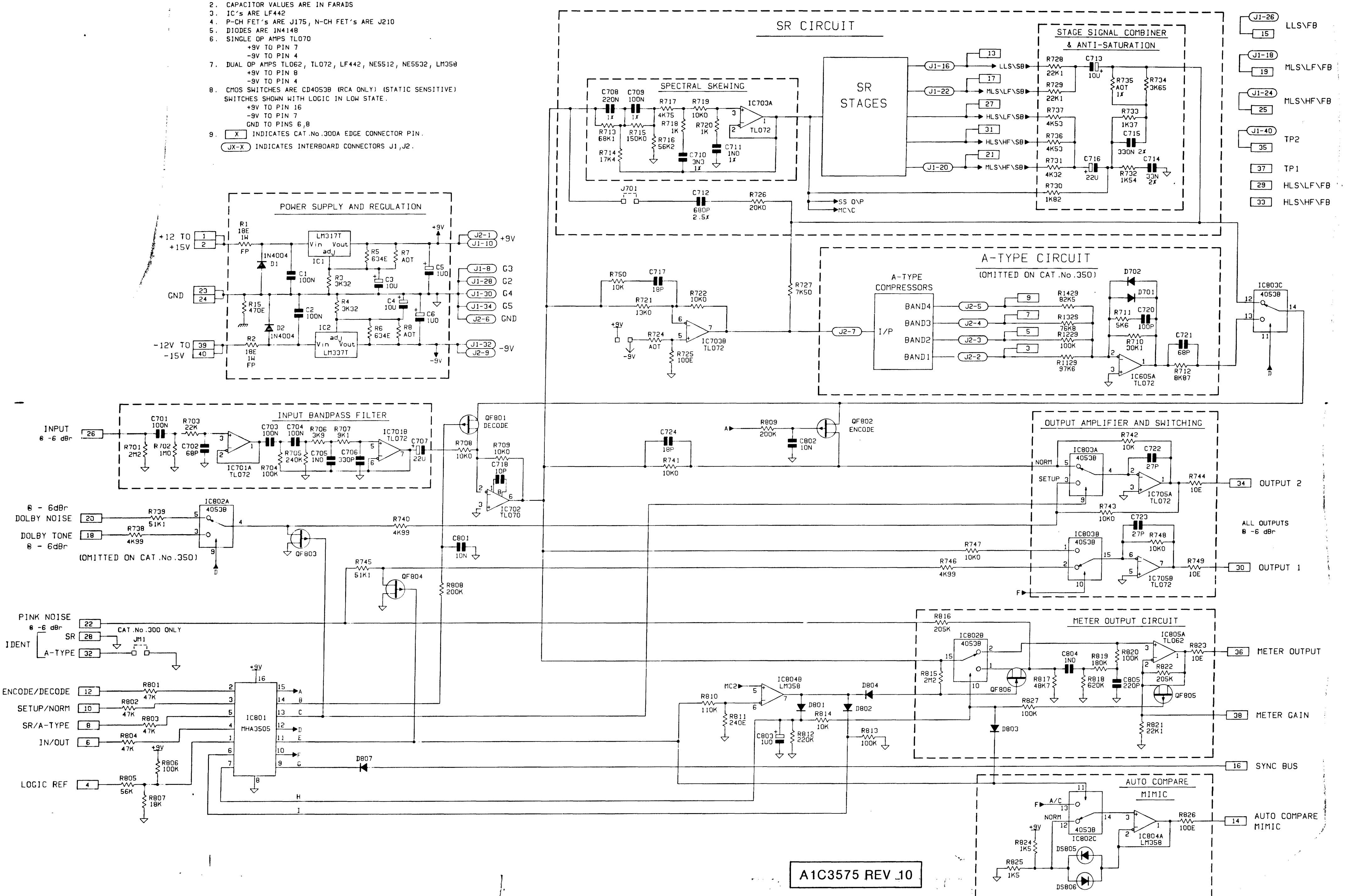
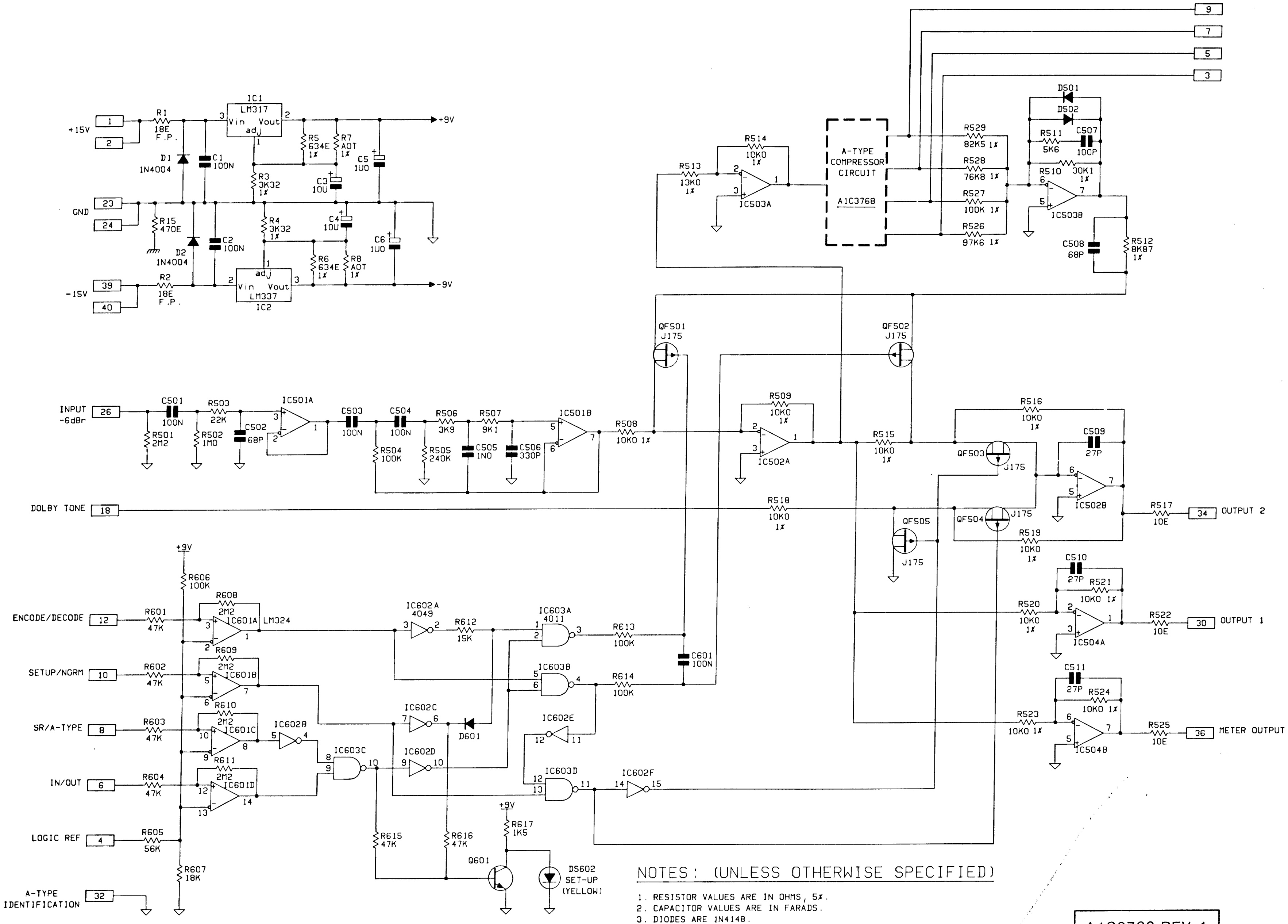


Figure 6.5 Cat. No. 300 main path circuit



NOTES: (UNLESS OTHERWISE SPECIFIED)

1. RESISTOR VALUES ARE IN OHMS, 5%.
2. CAPACITOR VALUES ARE IN FARADS.
3. DIODES ARE 1N4148.
4. BIPOLAR TRANSISTORS ARE DOLBY STANDARDS DEVICES:
   PNP=BC416, 2SA970 OR SIMILAR
   NPN=BC414, 2SC2240 OR SIMILAR
5. DUAL OP-AMPS ARE TL072. PIN8=+9V, PIN4=-9V.
6. IC601 IS LM324.
   PIN 4 = +9V PIN 11 = GND
7. IC602 IS CD4049.
   PIN 1 = +9V PIN 8 = GND
8. IC603 IS CD4011.
   PIN 7 = GND PIN 14 = +9V
9. F.P. DENOTES FLAME PROOF.

Figure 6.6 Cat. No. 450 main path circuit

SECTION 7
UNIT SERVICING/IN CASE OF DIFFICULTY

7.1 Isolation of the Problem

As a first step, switch the unit into **Bypass**. If the problem remains, it is very unlikely that the Model 363 is at fault. As a final confirmation that the unit is not at fault, the XLR connectors at the rear can be unplugged and connected together, thus removing the Model 363 totally from the signal path.

If the problem disappears with the unit bypassed, the fault is either in the Dolby processing modules or the motherboard inside the unit. In this case, exchange the two modules; if the fault changes channel, then the problem is likely to be in the processing module.

If the fault is present on both channels, it is possible that there is a fault in the power supplies.

Note that several functions are dependent on the type of module installed. For example, if the Cat. No. 450 (A-type only) is inserted, switching to SR will not illuminate the red LED and the module will remain in the "off" mode.

If the module is suspect, it should be checked. Testing equipment is available from Dolby Laboratories to verify the performance (see Section 7.3 below); alternatively refer to Section 7.2 "Quick Check" for a simple field test.

If a test extender (Dolby Cat. No. 387) is available, then a further check on the interface can be made by connecting together temporarily pins 26, 30, 34, and 36. Then with the module removed and the extender inserted, a signal at either the **From console** or the **From rec** input connectors (as determined by the **rec/play** switch) should appear simultaneously at the **To rec** and **To console** outputs, and also (if at or near Dolby level) on the calibration display. If the signal does not appear at any of these places in the normal record/play modes, then there is a problem in the motherboard.

The processor modules have been designed for accuracy, reliability and long life. There is no need for adjustment; the critical parts of the circuit are pre-adjusted during manufacture with selected fixed-value components, using custom-designed test equipment and procedures. These selected components determine the standard Dolby A-type and SR processor characteristics and any attempted repair by the user may result in degraded performance.

The modules should be returned to either Dolby Laboratories or the local agent for exchange and/or repair.

7.2 Quick Check for Dolby A-type NR and SR Processing

A quick test of the Dolby processing may be carried out in the field to check the outline characteristics of both the record (encode) and play (decode) modes. Tones at various levels and frequencies are sent to the Model 363 and the outputs at the **To recorder** and **To console** are measured. Alternatively, if (as is usually the case) two channels are available simple back-to-back checks can be made by putting one channel into the record mode and the other in the play mode. Test signals at various levels and frequencies and program material can be sent through the two channels in series, and a comparison of input to final output will very soon show up any problems.

In the field, it is sometimes difficult to measure levels precisely due to the lack of accurate instruments. This procedure, therefore, has been chosen to allow a quick operational "rule-of-thumb" check to be made. Accurate tests on the modules must be made using specialized test instruments and are of necessity too time consuming for an operational procedure.

MAIN

Measurements given and their expected ranges assume that the input signal level is only accurate to within ± 0.5 dB. Thus the range of values given does not represent the tolerance of the Dolby system; with a defined input, the accuracy is much better.

Tests are made by first switching out the signal processing, and selecting the record mode. Next send a tone into the **From console** input at Dolby level (so that the two green LEDs on the calibration display are equally illuminated). Measure the output at the **To recorder** output, which in most installations should be at the same level as the input. Reduce the level from the console by 24 dB; note the reading at the output, and select either A-type or SR. The measured signal should increase as listed in the table below.

To check the play mode, transfer the input test signal to the **From recorder** input and the meter to the **To console** output. Repeat the test above, making sure to switch to **play** and to check the calibration display has equally illuminated green LEDs (with processing switched out) before reducing the input by 24 dB.

FREQUENCY	SR		A-TYPE	
	ENCODE (±0.3dB)	DECODE (±0.9dB)	ENCODE (±0.3dB)	DECODE (±0.9dB)
80 Hz	+5.0	–9.9	+5.2	–8.6
800 Hz	+9.1	–17.1	+5.0	–8.9
8 kHz	+5.5	–10.1	+6.9	–11.3

Table 1. Approximate encode/decode level changes at 24 dB below Dolby level

7.3 Testers

Dolby Laboratories manufactures a range of simple go no-go test instruments for these modules. These testers (called Cat. No. 379 and Cat. No. 381) check two modules in a back-to-back connection, and perform simple tests to give a good indication if the module is operating correctly. While the test is not sophisticated enough to detect minor malfunctions, it is very useful to confirm that no major problems exist.

7.4 Motherboard Fault Diagnosis

If the fault is isolated in the motherboard, then either the unit can be returned to Dolby Laboratories offices or to the local Dolby agent. Alternatively, the unit may be repaired In-house. Circuit diagrams are provided in Section 6, and should allow simple fault-finding and repair.

> **CAUTION: The service instructions are for use qualified personnel only. To avoid electric shock do not perform any servicing unless you are qualified to do so!**

First, remove the unit from the rack. and remove the top cover (four screws at the sides—see Figure 7.1. at the end of the section for an exploded view of the unit). This allows access to much of the circuitry with the modules in place, and to all with them removed. The module edge connectors are also accessible and are labeled, and temporary connections can be made at the connector to simulate the module connections to assist in tracing signals.

If this analysis reveals the faulty components, then the complete interface assembly can also be removed to allow access to both sides of the board. Alternatively, since the board may be safely operated out of the case, removal may help in fault-finding.

7.5 **Disassembly** (numbers as (16) refer to Fig. 7.1)

1. Disconnect power from the unit, and remove cover (2) .

2. Unplug the three-pin connector (P13) linking the power transformer to the printed circuit board.

3. Turn the unit over, and undo the two screws (16) holding the heat sink to the chassis. (Do not undo the larger single screw (14) holding the transformer.)

4. Undo the three screws (18) holding the front panel extrusion to the main chassis.

5. Turn the unit right-side up and remove the two counterbored screws (17) holding the extreme left and right hand top side of the front panel extrusion to the chassis sides.

6. Remove the front panel extrusion (3) from the chassis.

7. Unlock each XLR body from its shell. Three different types of locking system are employed in different styles of connectors, two made by Switchcraft and one by Neutrik. Study the diagrams in Figure 7.2 to identify which variety is used in your unit (there will only be one style in any particular unit). In two of the styles (Figure 7.2 a and b), the bodies are clamped by means of a cam-lock screw which turns at the most through an eighth of a turn. These styles are made by made by either Switchcraft or Neutrik. These connectors , while using the same principles, have the locking screws in slightly different positions and operate in opposite senses. The screws <u>must</u> be unlocked from the rear of the unit (even though there appears to be access from the inside of the unit for the Switchcraft version).

   A more recent Switchcraft design uses a simple countersunk locking screw on the top of the body of the connector, towards the rear (Figure 7.2 c).

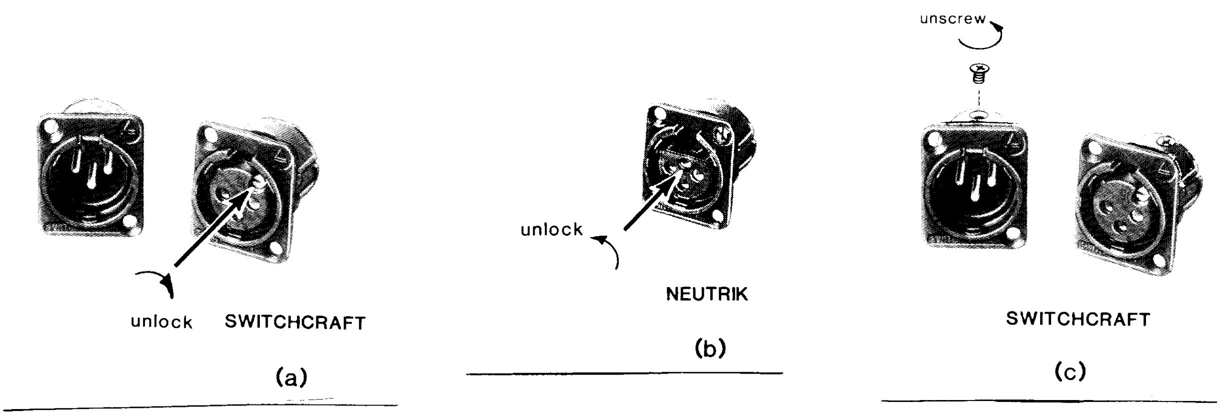


Figure 7.2

8a. In the case of the designs using a cam-lock system, identify the connector manufacturer by reading the name at the bottom outside of the connector. If a Switchcraft component, the screws will be towards the top right of the connector, and are silver coloured to stand out against the black colour of the body and insulator. Unlock the shells using a thin flat-bladed screwdriver by turning the screws <u>clockwise</u>.

8b. If a Neutrik connector, the screws are on the centre-line of the connector just above the centre. A very thin-bladed screwdriver must be used (tip width about 1.8 mm, around half that of the screw hole (alternatively, the special Neutrik tool can be used). Turn the screw <u>anti-clockwise</u> to unlock the shells.

8c. If the connector is in the style of Figure 7.1 c, simply undo all of the top-mounted captive countersunk screws a few turns to allow the connector bodies to slide out of the housing.

9. The motherboard (5) can then be withdrawn through the front of the tray. Once out, it can be reconnected to the transformer.

> **CAUTION:** Do not leave the mother board assembly complete with processing modules powered outside the chassis for extended periods of time. The heatsink will become very hot and the regulators may shut down, preventing further troubleshooting.

10. The transformer assembly can be removed by removing the green/yellow ground wire from the transformer to the ground lug (29), undoing the bottom fixing screw (14), then sliding the power input module upwards out of the chassis.

> **CARE!!** Live terminals are present on the rear of the power input module when the rubber boot is removed!!

11. Re-assemble unit by reversing the above steps.

7.6 Parts List

Most parts are available locally; however, the following lists Dolby part numbers for items which are either special or may be difficult to obtain locally. They can be ordered from local agents or Dolby Laboratories offices in London or San Francisco.

Component	**Circuit Reference** (only channel 1 listed)	**Dolby Part Number**
capacitors, 2.5%		
330p	C1215	25003
680p	C1214	25008
1n0	C203,403,901,1114	25010
2n2	C1202	25014
3n3	C1213	25017
6n8	C1203,1204,1205,1212	25021
capacitors, 5%		
22p	C104,105,201,202,304,305,401,402	24082
68p	C103,303,1206	24123
10n	C106,206,207	21071
15n	C1211	21073
33n	C1210	21077
220n	C101,102,301,302,1201,1209,1216	21135
capacitor 10u/50v	C1117,1118	22072
card guide		62281
connector, edge	J11,12	71133
connector, XLR female	J1,3,5,7	71145 plus 71146
connector, XLR male	J2,4,6,8	71144 plus 71146
connector, fan, 3-pin	PL14	70121 plus 3 x 75004
connector,9-pin male D	J9	71141
connector, 9-pin female D, cable mounting		70113 plus 9 x 75050
connector,15-pin female D	J10	71142
connector, power		70106
connector, transformer	PL13	70111 plus 3 x 75014
cover, screws M 5x5		60199
cover, unit top, up to S/N 649		63277
cover, unit top, after S/N 650		63280
diode, reference	DZ1203	44054
display, calibration	DS100	32043
duct for fan		63283
extrusion, front panel		63274
escutcheon, control panel		63275
fan and wiring harness		83099
fuse 250 mA 1.25"		56033
fuse 250 mA 20 mm		56035
fuse 500 mA 1.25"		56003
fuse 500 mA 20 mm		56036
fuse holder IEC		70108
fuse holder US		70109
insulator for guide		63284
integrated circuit	IC1205	45014
integrated circuit 4556	IC203, 403	44099
integrated circuit LM317T	IC1101	44025
integrated circuit LM337T	IC1102	44042
integrated circuit TL431	DZ1203	44054
jumper, grounding	J1101	74087

Component	Circuit Reference (only channel 1 listed)	Dolby Part Number
motherboard (up to S/N 649)		Cat. No. 373
motherboard (after S/N 650)		Cat. No. 386
opto isolator 4N28	IC901	33003
potentiometer, multi-turn	RV101,201,301,401	12014
power input module		70106
power input rubber boot		70112
regulator, +ve, LM317T	IC1101	44025
regulator, −ve, LM337T	IC1102	44042
relay	K201,401	51015
resistors, 0.1%, 0.25 watt:		
10k0	R108,109,203,204,208,209,210, 211,217,220,308,309,403,404, 408,409,410,411,417,420	13022
20k0	R106,107,206,207,212,213,216,219, 306,307,406,407,412,413,416,419	13023
resistors, 1%, 0.4 watt:		
22E1	R214,215,414,415	10350
237	R1103,1106	10871
1k74	R111,201,311,401	10455
2k61	R1104,1105	10473
3k01	R1202	10479
4k75	R1011,1218,1238,1241,1250,1251	10499
5k49	R1009	10505
6k49	R402,1220	10511
6k98	R113,202,228,229	10515
10k0	R140,105,304,305	10530
12k4	R1015,1204	10539
13K0	R1239,1242,1243	10541
16K9	R1201	10553
17k4	R1203	10554
20k5	R101,301,1006,1013,1237	10561
57k6	R1205,1206,1214,1216,1217	10604
64k9	R1236	10609
158k	R1235	10648
243k	R1234	10720
511k	R102,103,302,303,1227,1228	10891
3M01	R1226	10798
switch, SR/off/A	S901	52064
switch, check, rec/play	S903,904	52073
switch, bypass	S902,1302	52098
switch, set-up	S1101	52099
transformer, power line	T1101	54033
transistor, BC638	Q1101	41018
transistor, n-fet	QF101,102,301,302,901,1202	42007
transistor, p-fet	QF103,201,202,303	42003
transformer	T1101	54033
tray (up to S/N 649)		63276
tray (after S/N 650)		63282
voltage selector wheel		70107
white noise IC, MM5437	IC1205	45014

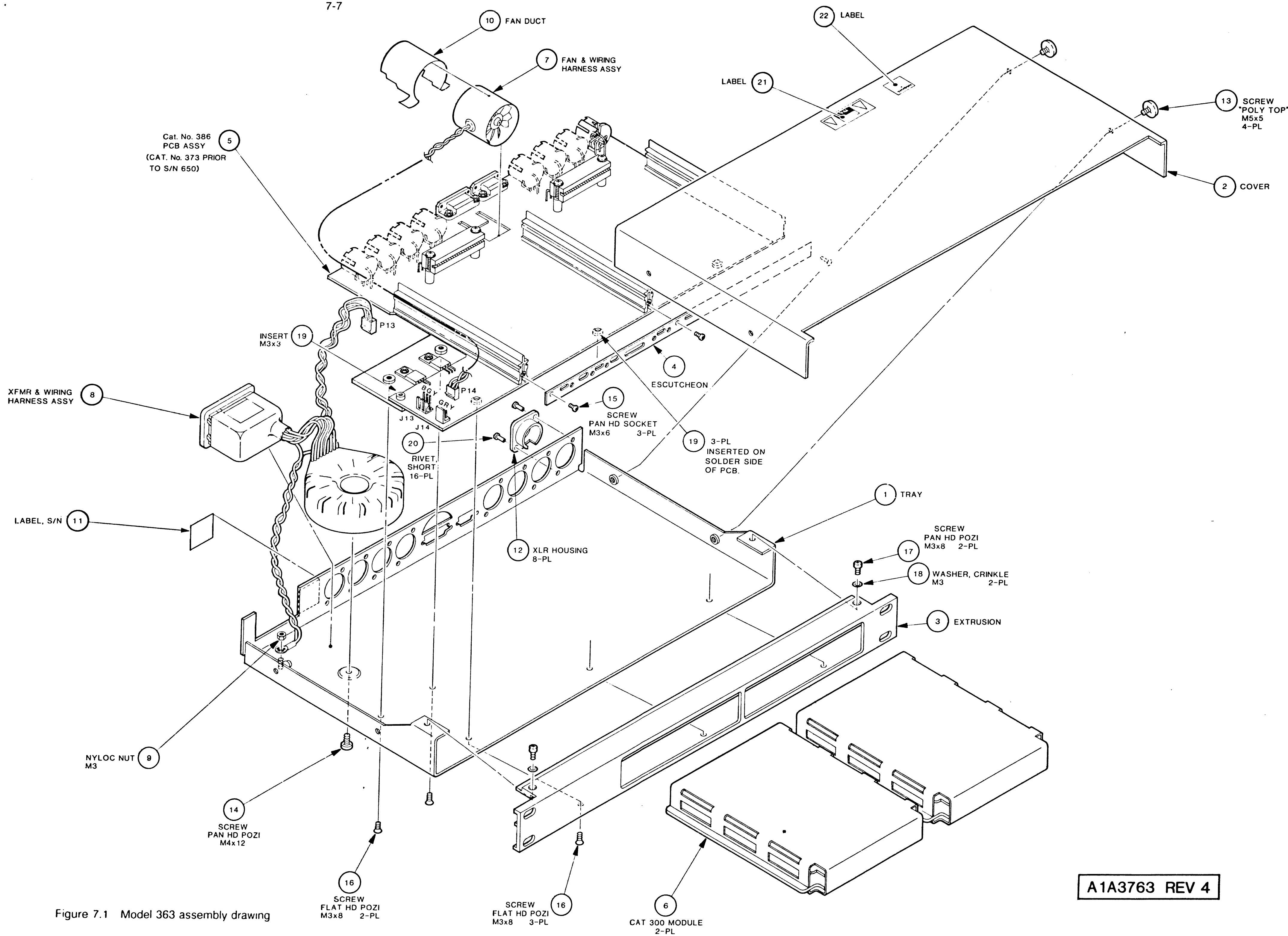


Figure 7.1 Model 363 assembly drawing

SECTION 8
APPENDICES

Part 1 Measurement of Noise and NR Effect
Part 2 Dolby SR; What It Is, and What It Does
Part 3 The Spectral Recording Process
Part 4 An Audio Noise Reduction System

Part 2 is a reprint of a semi-technical description of Dolby SR, which forms a good introduction to the concepts before reading the following two papers.

These next two papers in parts 3 and 4 were written and presented by Ray M. Dolby; the first, on Dolby SR, was given at the 81st Convention of the Audio Engineering Society, Los Angeles, November 1986. The second, on Dolby A-type noise reduction, was given at the 32nd Convention in Los Angeles, in April 1967. Both papers are reprinted from the Journal of the Society.

Measurement of noise and noise reduction effect of Dolby SR

A. Measurement of Noise

Human hearing is not uniformly sensitive with frequency. At low levels, those in which unwanted background noises exist, it is most sensitive over a middle range of frequencies, and progressively less so at frequencies further from the middle. Beyond about 20 Hz and 20 kHz it is "infinitely insensitive," that is, these frequencies cannot be heard irrespective of level by the large majority of people. However, the spectrum of noise delivered by most audio systems is roughly uniform with frequency ("white"), containing significant noise contributions at the extremes of the audio spectrum and usually well beyond those extremes.

Dolby SR has been designed to reduce noise in accordance with its audibility. Operating on signals at the extremes of the audio spectrum or beyond, where noise is not a problem, is counter-productive; it increases the possibility of audible side-effects and obviously has no audible benefit in reducing noise. SR therefore reduces noise most over the middle of the spectrum, has only a modest effect at the extremes, and indeed slightly degrades noise at subsonic frequencies (less than 20 Hz). The design of SR in accordance with the audibility of noise means that, for sensible results, noise measurements must also take into account the audibility of noise. A simple wide-band (unweighted) noise measurement will yield a disappointing result which does not agree with the listening experience. Most of the noise being measured will be at and beyond the extremes of the audible range; this can be confirmed by examination of the noise on a spectrum analyzer. On an oscilloscope the unreduced very low frequency noise will yield an unsteady base-line, which has in the past mistakenly been interpreted as low frequency instability; it actually is the original noise waveform with the audible frequencies removed, leaving the inaudible very low frequencies.

Weighted noise measurements yield answers which correlate more closely with audibility. However, depending on the amount of ultrasonic noise, the A- weighting curve may provide inadequate attenuation of noise above 20 kHz, and therefore must be supplemented by an audio bandpass filter (say 20 kHz, preferably using a 4-pole design) to eliminate inaudible frequencies. CCIR 468 and CCIR/ARM give results which agree with listening provided the test equipment works properly. Unfortunately, many integrated instruments (meters containing switchable weighting filters) have inadequate headroom in their amplifiers and/or high frequency crosstalk across their switches, and as a result give grossly inaccurate results. Such problems can be identified while measuring noise reduced by SR by switching the meter from wide-band (not audio band, but 100 kHz or more) to weighted measurement. If it is necessary to increase the meter gain by 20 dB or more to restore the deflection of the meter, the meter amplifier is probably overloading, and a quiet audio low- or bandpass filter should be connected ahead of the meter. In many cases a passive single-pole low-pass at 20 kHz will be adequate.

Assuming that noise sources ahead of the SR encoder and after the SR decoder are negligible, Dolby SR will give an audible and a weighted measured noise reduction in the range 20 to 24 dB. If less noise reduction is observed, it is likely that the measurement is at fault, as described above.

B. Measurement of Noise Reduction Effect

In many installations the extended dynamic range offered by Dolby SR means that noise sources which can normally be neglected (such as input and output amplifiers) may become significant. Any measurement of the effectiveness of the noise reduction action must also measure the contribution of noise from these sources. Simply replaying a biased (no signal) tape and turning the noise reduction system on and off will not give a figure for the practical efficiency of the system; it will only measure the playback effect.

Unfortunately, this playback-only technique does not always give results which relate well to the effect produced in the full record/replay process. The Dolby system is a complementary system: recordings to be played with noise reduction switched on are not made with noise reduction switched off. Therefore it is not sufficient to use a piece of blank tape for the tape noise source

The tape to be played back with noise reduction switched on must be recorded with noise reduction switched on. Similarly, the tape to be played back with noise reduction switched off must be recorded with noise reduction switched off. Only by doing this can the effect of extraneous noise sources on the total noise be properly assessed.

The correct test procedure is as follows:

a. Feed a signal at normal reference level from the console, record a test length, play back the tape, and confirm the output level is also at reference.

b. Remove the source signal (if the test oscillator is merely turned down, ensure there is really no signal at its output--a small leakage of tone would invalidate the noise reduction measurement).

c. Record a length of tape with the noise reduction system off, and at a convenient timing point turn the system on.

d. Rewind the tape, and play back the section recorded without noise reduction with the noise reduction switched off, and measure the output noise.

e. At the point at which the system was switched on in the recording, switch on the system and measure the reduced output noise.

f. The noise reduction effect is the difference between the measurements in d. and e. above. The exact figure will depend on the weighting method used, but should be over 10 dB for Dolby A-type noise reduction and over 20 dB for Dolby SR.

Dolby® SR

Dolby® spectral recording

What It Is and What It Does

Dolby Spectral Recording
Dolby spectral recording (Dolby SR) is a new professional studio mastering system that yields recordings with exceptional purity of sound. Several important technical advancements are introduced in the new system. One is a substantial extension of available headroom, which allows the use of a uniformly high maximum recording level at all audio frequencies. Another is the practical elimination of the influence of noise and non-linearity on the reproduced sound. These advancements are achieved by new circuit functions, adaptive to the signal spectrum, and by the consistent application of minimum processing to the signal: *the principle of least treatment.*

Recording with Dolby SR is extremely simple. It can be used with any modern professional recorder; installation is rapid, operation is simple and reliable, and routine maintenance is unnecessary. Its unique Auto Compare feature allows immediate, dependable validation of recording system performance without additional instruments or delays in the studio work schedule. Plug-in module construction and compatibility with existing Dolby mainframes provide convenience and economy for studios that use Dolby SR.

MAIN

The principle of least treatment

The ear is the final destination of all audio signals, and the most sensitive instrument for their analysis. An ideal audio device or system would impose no audible limitation on the signal passing through it. The design of Dolby SR has been carried out with close attention to the properties of human hearing, especially the need to prevent any audible artifacts of signal processing.

At the lowest signal levels, or in the absence of a signal, Dolby SR applies a fixed gain/frequency characteristic that reduces noise and other low-level disturbances by as much as 25 dB. Only when the level of part of the signal spectrum increases significantly does the circuit adaptively change its own spectral characteristics. When this happens, Dolby SR changes gain only at frequencies where change is needed, and only by the amount required. Adherence to this principle, the principle of least treatment, is critical to maintaining the extreme purity of sound audible in Dolby SR recordings.

Listening is the most demanding test

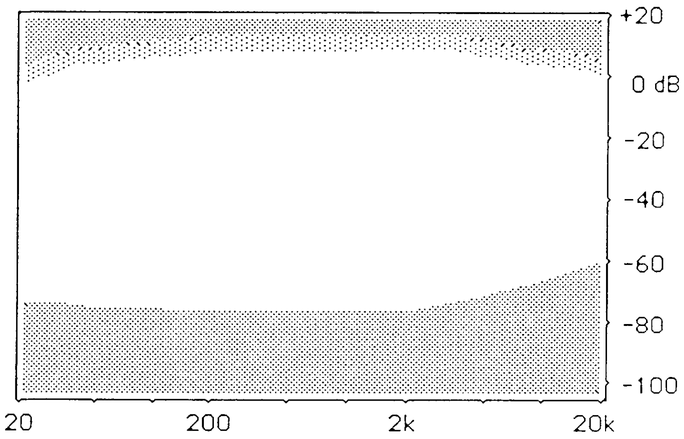
Laboratory measurements using test signals show that Dolby SR recordings contain very little noise, distortion and other impurities. However, meters cannot tell us how good Dolby SR tapes sound, because equipment does not respond to a recorded signal in as complicated a way as the ear and brain. The most important and reliable test of any signal processor is a careful comparison of line-in and line-out signals while a live recording is made in a quiet studio. We urge engineers, producers and recording artists to carry out such tests with Dolby SR, and to compare their own Dolby SR recordings to those based on any other technology.

Why Dolby SR is needed

All recordings and communication systems have definite dynamic range limits. However, a simple measurement of maximum level and noise level does not reliably indicate how recordings made with such a system will sound. Such a test says nothing about noise that appears only in the presence of a signal, or about system behavior when the signal is at the overload level. Analog tape, for example, saturates gently; digital recordings, on the other hand, clip fully if maximum level is reached even for less than a millisecond. Because Dolby SR increases recording headroom considerably, there is less risk of under- or over-recording. The engineer's working space is increased, and there is greater freedom for creative effort. Dolby SR provides effective protection during original recording, during mixdown, when equalization or specialized signal processing requires the lowest possible noise level, and when multi-generation copies are needed.

The analog tape recording window

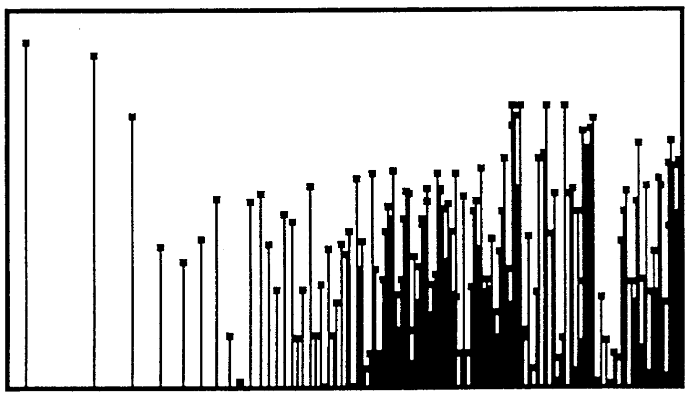
1



The limits of unassisted analog recording, using a standard professional recorder and tape at 15 ips, are sketched in Figure 1. The limit at high signal levels, actually a gradual overload, is at the top of the clear area. The noise level is the bottom of the clear area. Both the overload level and the noise vary with frequency. The central, open part of the sketch can be thought of as a window through which the signal must fit if it is to be recorded.

Recording a signal

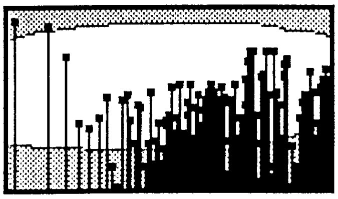
2



A music program constantly changes in level and frequency content. A moment in such a program might have the spectrum shown in Figure 2. The signal level varies with frequency. The gain setting that would give the best recording at middle frequencies would cause the high frequencies to overload. One objective of Dolby spectral recording is to achieve as nearly optimum a level as is possible at all audio frequencies.

Finding the right level

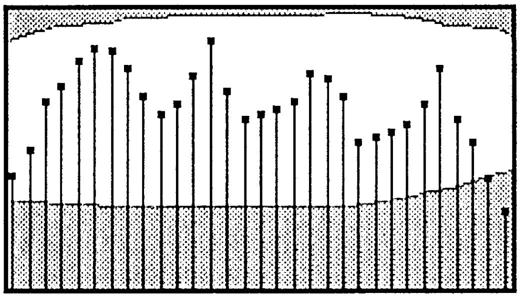
3



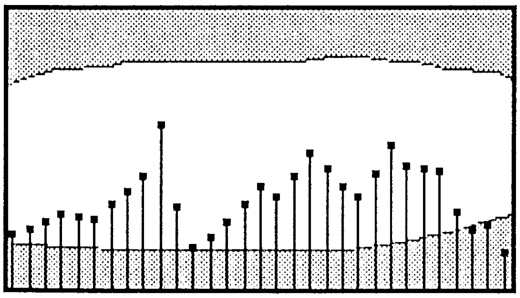
4

Figures 3 and 4 show what happens when the recording gain is adjusted; the signal spectrum moves up or down in the window. Even when the recording gain is set at the highest safe value, as in Figure 3, much of the capacity of the recording system, that is, the *spectral* space above the tops of signal components and inside the window, is unused. Dolby SR makes use of this capacity.

The simple compander



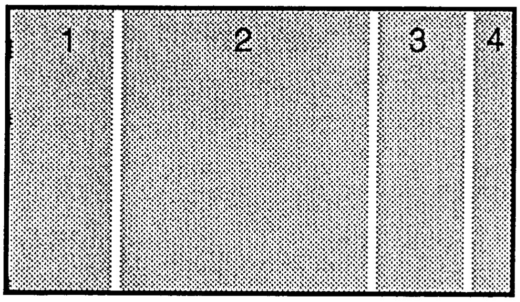
5a



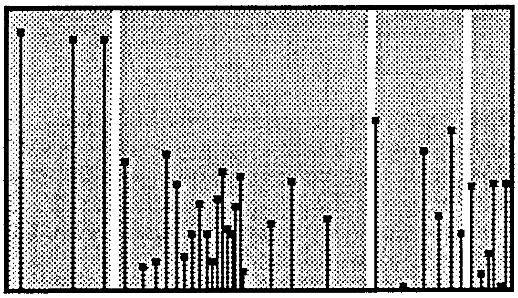
5b

The simple or broadband compander (compressor and expander) was first used in attempts to increase the dynamic range of recording systems. During recording, such a compander increases gain when the overall signal level is low; some companders also reduce gain when signal level is high. During playback, the compander's action is reversed; high-level signals, regardless of their frequency, cause all frequencies to be played back at a high level (Figure 5a), while low signal level, or no signal, causes playback gain to drop (Figure 5b). The *measured* dynamic range of a recording system may seem to be increased greatly. However, a simple compander meets its dynamic range specifications only when no signal is recorded. When a real program is recorded, the compander is often at rest when it is needed most. When it works, on the other hand, critical listeners can hear artifacts, like "pumping" modulation of the background noise or signal, as well as limited transient performance.

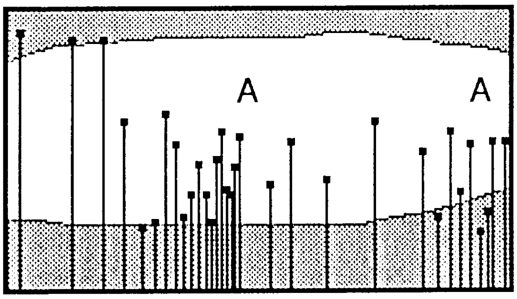
Dolby A-type noise reduction



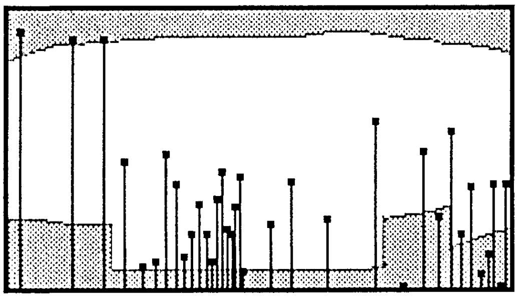
6a



6b



6c



6d

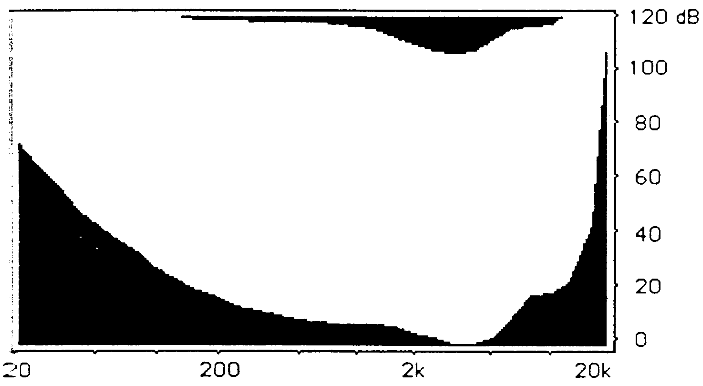
No professional signal processor has ever been as widely used as Dolby A-type noise reduction. This is still true now, 25 years after its introduction. Dolby A-type processing substantially improves the efficiency of magnetic and optical recording media and audio communication systems. Dolby A-type signal processing relies on compression and expansion, but only at low signal levels, and separately in four frequency bands (Figure 6a). The signal components in each band (Figure 6b) are integrated; if this level is below a fixed threshold, it is boosted during recording (locations marked "A" in Figure 6c), and attenuated during playback (Figure 6d).

The boost used in Dolby A-type noise reduction is 10 dB across most of the audio band, increasing to 15 dB at very high frequencies. To improve recording further, it is not enough simply to increase these figures; the boost must conform more closely to the signal spectrum than is possible in a four-band system.

MAIN

The auditory window

7



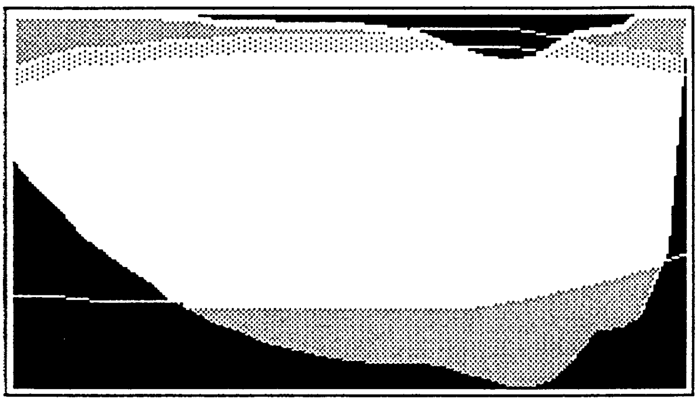
One way to define ideal sound reproduction is to show the limits of the human hearing system as a window, as we did for analog tape recording in Figure 1. Such an auditory window is sketched in Figure 7. The top of the graph corresponds to 120 dB, a continuous sound pressure level that is unbearably loud; at 5 kHz, sound changes to pain at about 110 dB. It is safest, of course, to leave a margin between such a level and the highest continuous sound level one aims at reproducing.

The boundary of the window near the top of the plot is 6 dB below the threshold of pain at each frequency.

The bottom of the window is the *threshold of hearing*; sounds at this level are just audible to a listener with sensitive hearing. The level of the background noise in a *very* quiet recording studio may be 10 to 15 dB. A recording system with a window like that shown in the figure could be played back without audible noise or overload, even if the highest-level signals were literally at deafening levels.

Playback of an analog recording

8

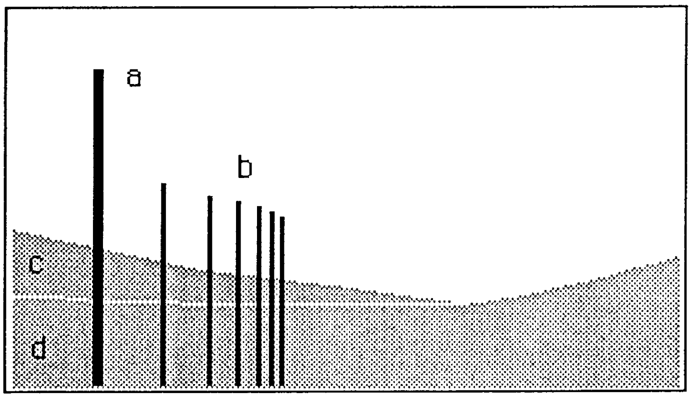


We can learn more about tape recording by superimposing the analog tape window on top of the auditory window. Sliding the analog window up or down corresponds to playing a tape at higher or lower level. In Figure 8 we have set the playback gain so that the maximum level signal that can be put on the tape lines up with a continuous sound pressure level of 110 dB. We will stay with that setting as we look at various recording system windows.

Several interesting facts are visible in this figure. One is that the noise of the tape will be audible only in a restricted range of middle frequencies, that is, where the auditory threshold is lowest. This is because noise or distortion components at higher and lower frequencies, even if only slightly below the threshold, are totally inaudible. Another observation is that if the audible noise in the mid-range frequency band could be reduced by 20-25 dB, no noise would be heard at all.

Disturbances produced by the presence of a signal

9



When no signal is present, the only low-level defect that can be measured is tape hiss. However, in the presence of a signal, the analog tape recording window closes further as other artifacts are added to the signal, layer by layer. Figure 9 shows several components of noise and non-linearity that can appear in the presence of a signal. The signal is shown as the vertical bar at (a), and is at a level that causes 3% harmonic distortion. These harmonics are shown in correct scale at (b). Modulation noise, which appears only when a signal is present, is spread over a wide range in the spectrum (c). The bottom layer of noise, tape hiss (d), is caused by statistical fluctuations in magnetic domain orientation in the tape coating.

Available dynamic range

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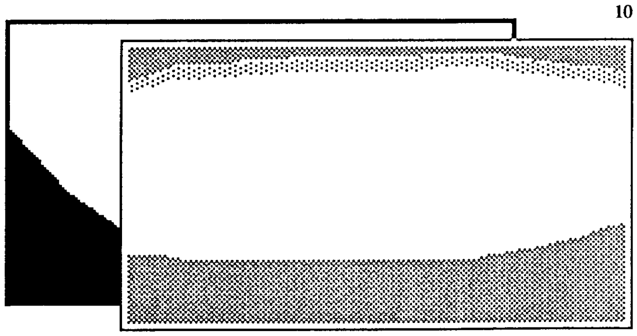
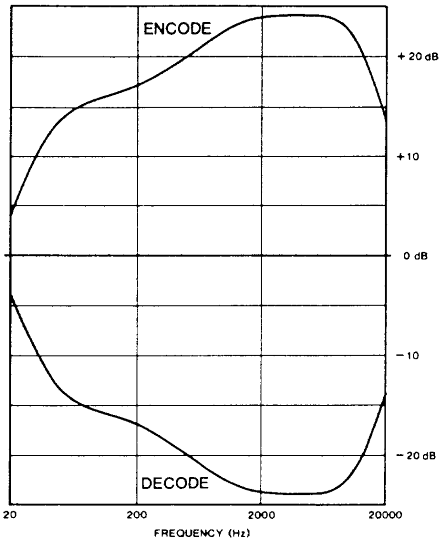


Another interesting fact is shown in Figure 10. Analog recording might be thought to be deficient in dynamic range at very low frequencies because saturation occurs at lower tape flux levels in that part of the spectrum. However, the opening in the analog window at low frequencies is actually *larger* than the opening in the auditory window. The same is true at high frequencies. If signal components at different frequencies were simply recorded at different gain settings, the effective dynamic range of analog recording could be extended considerably.

What Dolby Spectral Recording Does

Dolby SR at low levels

11a



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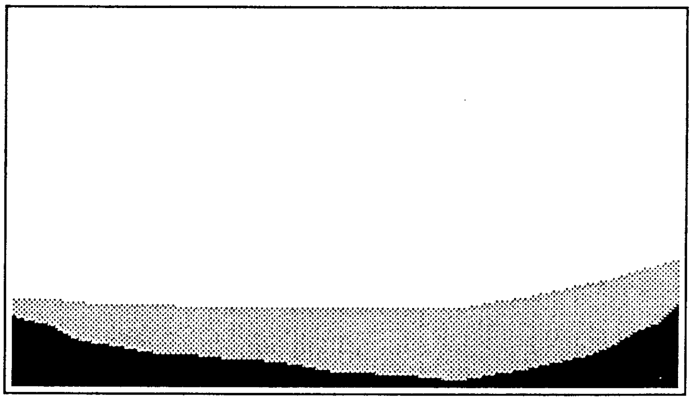

The simplest way to suppress noise and other low-level recording defects is to use as high a recording gain as possible. From what we have already seen, an even better idea would be to use the optimum gain in each part of the spectrum. This is what Dolby SR does at very low signal levels. The result is a form of fixed equalization that does not change as long as the signal level stays below a certain threshold. When the recording is played back, the same equalization is applied in reverse, and any background noise is lowered by the same amount (Figure 11a). The upper curve in Figure 11b is the noise level of a typical professional tape at 15 ips. The lower curve shows the change in this level that results from the use of Dolby SR fixed low-level equalization. Since there are no dynamic changes taking place – the equalization is fixed – no dynamic side effects, audible or inaudible, are possible.

Dynamic action of Dolby SR at moderate levels

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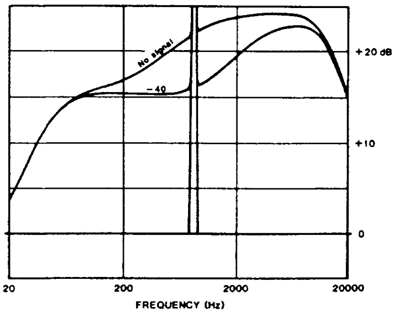

If the signal increases in any part of the spectrum, some adjustment of gain must be made to ensure that overload will not take place. This is done in a very gradual way by Dolby SR, so gradual that there is no danger of producing audible modulation of the signal or any other audible effect.

The principal mechanism of Dolby SR is a group of ten fixed- and sliding-band filters with gentle slopes. Those with fixed bandwidth are electronically controlled to vary their gain; those with fixed gain can be adjusted to cover different frequency ranges. By selecting and combining from the group, the Dolby SR control circuit can create an infinite number of filters through which the signal must pass before it is recorded. During playback, filters are automatically created that are the exact opposite of those used during recording.

Filter selection and adjustment is controlled by a continuous analysis of the signal spectrum and a process called *action substitution.* Action substitution determines which of the two types of filters will predominate and how each must be adjusted to produce the optimum composite filter (Figure 12). Even when the signal level increases substantially, the system is designed to deviate as little as possible from the fixed characteristic shown in Figure 11, in accordance with the principle of least treatment.

Dolby SR action at high levels

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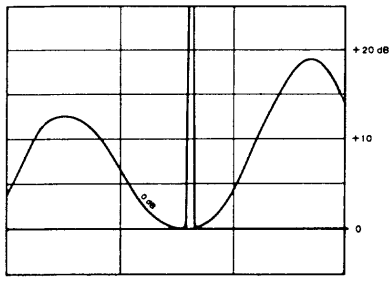

When a high-level component appears in the signal spectrum, Dolby SR assumes the kind of characteristic shown in Figure 13. In this example, a single tone at 800 Hz and at a level of 0 dB has been applied to the system input. Dolby SR reduces recording gain, but only at and near the frequency of the tone, and only by the amount needed to prevent overload. Above and below this part of the spectrum, the Dolby SR curve returns to the fixed, low-level characteristic. This action has results that are especially impressive when listening to a Dolby SR recording *without* decoding. Although bright in sound, the program does not appear to have been subjected to any dynamic processing. Since the only other system action consists of fixed equalization at very low levels, it is not surprising that during decoded playback, no trace of processing can be heard, except for remarkable clarity of reproduction.

Masking and Dolby SR

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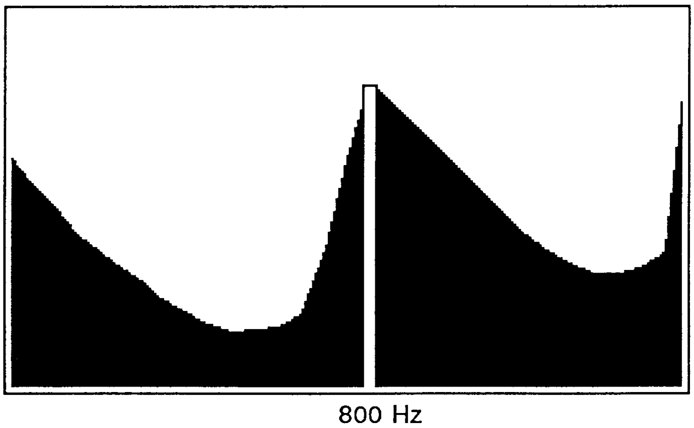

The changes that take place in the Dolby SR circuit are adaptive; that is, the system filter always adjusts itself to maintain the highest practical gain at every frequency as the signal *spectrum* changes. The human ear and brain also respond to these changes in the signal spectrum; one such response is a form of signal processing known as *masking*, one of the most extensively studied aspects of hearing.

Masking is the concealment of a low-level sound by a sound higher in level. A similar effect takes place in vision, when the daylight sky makes the stars disappear.

Most audio signal processing systems operate quite independently of the behavior of the ear and brain, and always take the same action that they would if nobody were listening to the output of the system. The extraordinary purity of sound of Dolby SR recordings is due in large measure to an elegant cooperation of adaptive signal processing and auditory masking.

In masking, a high level signal component raises the auditory threshold above and below the signal frequency. Sounds lower in level and near it in frequency disappear completely in this psychoacoustic "shadow" (Figure 14). In the Dolby SR circuit, feedback of the signal characteristics determines how each filter in the Dolby SR circuit must change to most closely envelop the masking "shadow." This is the way that Dolby SR applies as much gain as possible everywhere in the spectrum. The only region of the spectrum that is not boosted in gain is the region that is controlled by masking, where audible low-level information does not exist. Dolby SR electronic signal processing is silently traded for auditory signal processing in that part of the spectrum. It could fairly be said that although most of the Dolby SR system is on the circuit board, some of it is in the human brain.

Immunizing the system to errors

Another feature of Dolby SR is *spectral skewing*, which reduces the level of the incoming signal at extremely low and extremely high frequencies. Spectral skewing desensitizes Dolby SR to minor aberrations in tape-to-head contact and azimuth alignment, which might cause fluctuations in high-frequency response, and to head bumps or low-frequency variations in alignment tape levels. Although these effects are often inaudible, they can disturb the operation of other signal processing systems.

Recording complex signals at high levels

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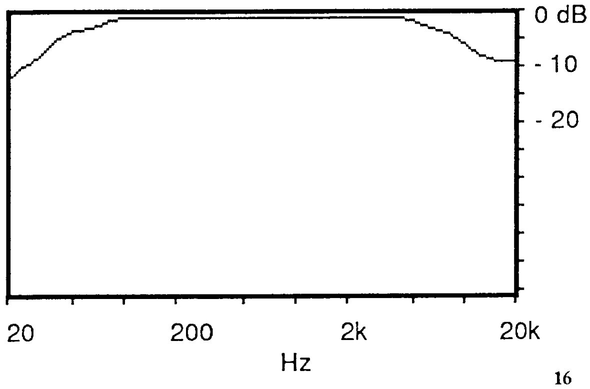

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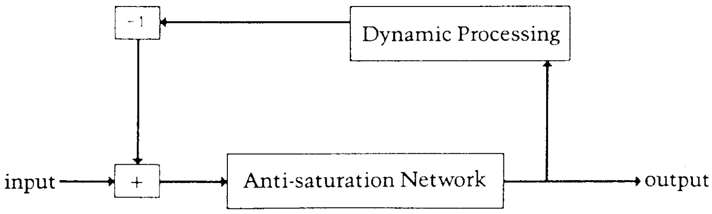


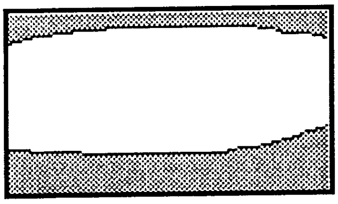
Headroom is as important as any other property of a recording system. Analog tape recording, as engineers know, has a gradual or "soft" clipping characteristic. In digital recording, there is no saturation region at all; there is simply clipping, in which the same digital "word" is recorded over and over as long as the signal remains above the limit. The Dolby SR circuit contains a feature designed specifically to deal with extremely high levels at low and high frequencies. The anti-saturation characteristic is sketched in Figure 15, and the circuit configuration shown schematically in Figure 16. Low-level signals pass through the side chain for processing; as signal level increases, an increasing proportion of the signal follows the direct path, which applies no dynamic signal processing. By introducing attenuation of high and low frequencies in this path, a significant increase in headroom is provided, further maintaining signal purity, with negligible effect on low-level operation.

Comparing Dolby spectral recording to other methods
We can compare the *static* performance of different recording systems by superimposing their windows and the auditory window (Figure 7). Any limitation that might be audible will appear as an obstruction that reduces the size of the opening in the auditory window. It is important to remember that this method of comparison does not show audible *dynamic* effects, such as modulation noise of analog systems; nor does it show low-level non-linearities, non-monotonicity, or effects of d.c. asymmetry, all of which may occur to varying degrees in digital recording systems. These effects all close the corresponding system windows from the bottom when a signal is present. In the figures that follow, for unassisted analog tape and Dolby SR, zero level is 320 nWb/m; for digital recording, zero level is 10 dB below the absolute clipping level. In all three examples, the relative vertical positions of the windows have been chosen so that the maximum recorded level will be presented at an acoustic level of 110 dB during playback.

Analog recording (no signal processing) [Figures 17,18]
Unassisted analog tape recording shows characteristic limitations in available headroom at low and high frequencies and a substantial level of noise at mid-frequencies.

Digital recording [Figures 19,20]
A typical digital recording provides performance that is better than unassisted analog tape in several obvious ways. The main drawbacks are the hard clipping barrier of digital recording and the disadvantageous spectral distribution of noise. Although the measured dynamic range of a digital recording system may exceed 90 dB, the noise level is not uniform with frequency. The noise level is extremely low at very low frequencies, much lower than the noise of analog tape, which is already more than adequate at low frequencies. However, digital system noise crosses the auditory threshold precisely in the spectral region where the ear is most sensitive. The usable improvement in noise level, especially in the presence of a signal, is not as great as theory predicts. Because the noise generated by a digital system is not random, and is therefore especially noticeable to the ear, it is normally masked by the addition of "dither" noise, elevating the final noise level.

Dolby SR [Figures 21,22]
This data was obtained using standard tape and a widely used professional recorder operating at 15 ips. The noise at the very bottom of the window could not be heard in a recording studio or control room unless the playback gain were increased considerably; under those conditions, maximum peak levels would approach or surpass the auditory threshold of pain. Played back at very high levels for test purposes, the audible noise floor of a Dolby SR recording is normally the noise of the microphone amplifiers, console electronics, or electronic instrument amplifiers. In a studio, with playback gain set as shown in the figure, and no signal present, the background noise of a Dolby SR recording is below the threshold of human hearing, and cannot be operationally improved.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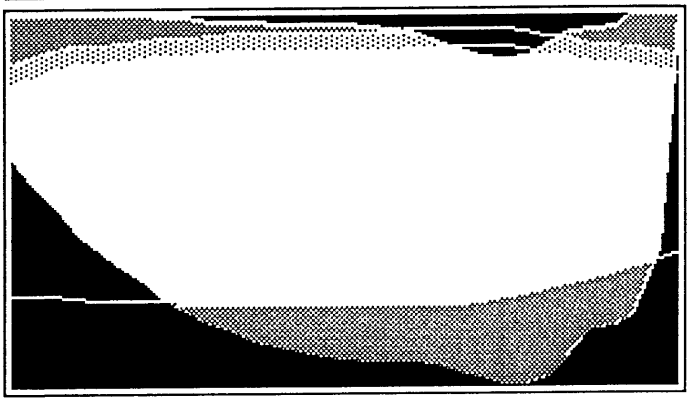
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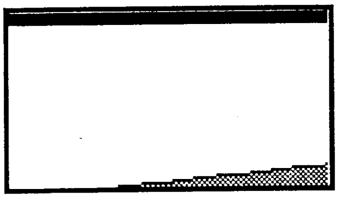
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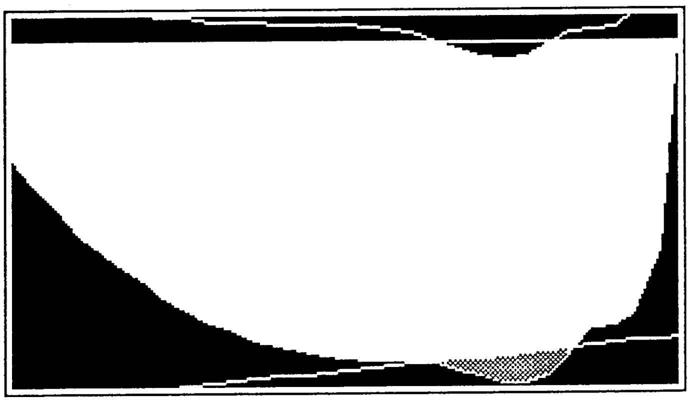
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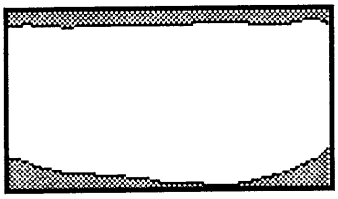
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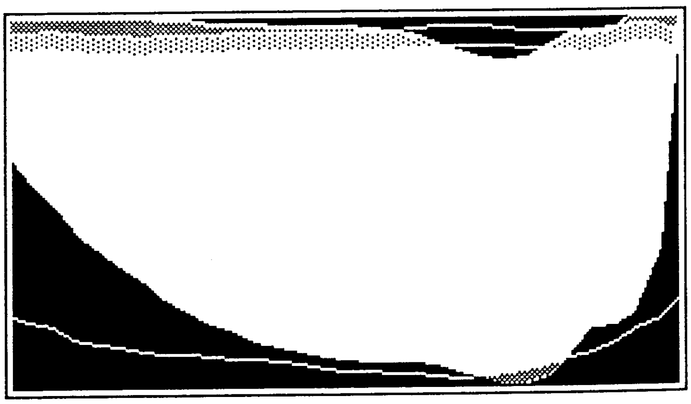
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MAIN

Dolby SR works with every modern recorder

Dolby SR processing can be used with any modern professional analog tape recorder or high quality audio communication system. This means that nearly every recording studio and communications facility in the world is already equipped to install and use Dolby SR. Often, changing over to Dolby SR will only require the removal of the Dolby A-type noise reduction modules already installed and their replacement by Dolby SR modules of the appropriate type. Its practical design makes recording with Dolby SR easier than recording without it; and editing, mixdown, copying, maintenance and other studio procedures are generally simpler because of the features of the new system.

Full information on Dolby spectral recording equipment is available on request from Dolby Laboratories, Inc.

Technical note

The following list contains sources used to prepare the illustrations in this booklet and suggestions for further reading. Copies of publications marked (*) are available on request from Dolby Laboratories, Inc.

Beranek, L. *Acoustics* (1954), McGraw-Hill, New York.

Dolby, R. M. "A 20 dB Audio Noise Reduction System for Consumer Applications" (1983), J. Audio Eng. Soc., **31**, 3, pp. 96-113.*

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Fielder, L. D. "Assessing the Low Level Sound Quality of Digital Audio Systems" Presented at the 79th Convention of the Audio Engineering Society, November, 1985.*

Furrer, W. and Lauber, A. *Raum- und Bau-Akustik Lärm-abwehr* (1972), Birkhäuser-Verlag, Basel.

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Warren, R. M. *Auditory Perception: A New Synthesis* (1982), Pergamon Press, New York.

Zwicker, E. *Psychoakustik.* (1982), Springer-Verlag, Berlin.

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MAIN

The Spectral Recording Process*

RAY DOLBY

Dolby Laboratories Inc., San Francisco and London

A complementary audio signal encoding and decoding format, called spectral recording (SR), for use in professional magnetic recording and similar applications is described. The processing algorithm is highly responsive to the spectral properties of the signal. A further characteristic used during encoding deemphasizes high-level signal components in the frequency regions usually subject to channel overload. The process results in a significant reduction of audible noise and distortion arising in the channel.

0 INTRODUCTION

In 1980, some 14 years after the introduction of A-type noise reduction [1], the author began work on the development of the next generation system for general-purpose professional recording and transmission. A configuration that would employ the A-type characteristics as part of the new system, with switchable compatibility, was considered initially. However, this would not take full advantage of the new technology embodied in the C-type system [2], nor would it readily allow the incorporation of some further new concepts. Therefore, the particular parameters of the A-type system were abandoned as a starting point for the new development. However, the basic principles, which appear to be as valid as when they were first introduced, were retained: the use of a main signal path without any dynamic processing to pass high-level signals, coupled with a low-level side chain compressor to provide dynamic action.

The design goals of the new system were set high. The new technology, called spectral recording (SR), should provide master recordings of the very highest quality, especially with regard to audible signal purity. Yet the system should be practical and economical for routine applications, being suitable for easy and troublefree use in a wide variety of professional recording and transmission environments. Certain new techniques, to be described, provide the required signal quality and practicality but result in circuit complexity. Reliance has been placed on improved circuit implementation and manufacturing techniques to overcome the problems of complexity and to ensure economical production of the new system.

* Presented at the 81st Convention of the Audio Engineering Society, Los Angeles, 1986 November 12–16.

1 BRIEF OUTLINE

The goal of the spectral recording process is to modify the various components of the incoming signal in such a way as to load an imperfect recording or transmission medium in the most rational way. Generally, high-level signal components at both ends of the spectrum are attenuated, whereby a better match with the overload characteristic of the medium is provided. At the same time, low-level components of the signal are amplified substantially, in a highly frequency-selective way. These effects are reversed during reproduction, restoring the original signal. The result is a significant reduction of distortion and noise, in both the absence and the presence of signals.

The process has a number of layout and operating characteristics in common with the A-type ([1], B-type [3], [4], and C-type [2], noise reduction systems. The SR process takes these developments considerably further in the same general direction. An understanding of the new system requires reference to the technical papers on these earlier systems, the C-type paper [2] being particularly relevant.

Referring to Fig. 1, which will later be described in detail, a main signal path is primarily responsible for

MAIN

conveying high-level signals. A side chain signal with the SR process characteristic is additively combined with the main signal in the encoding mode and subtractively in the decoding mode, whereby an overall complementary action is obtained.

The SR stage layout resembles that of the C-type system, except that three levels or stages of action staggering are used: high level, mid level, and low level (HLS, MLS, LLS). There are various advantages arising from the use of multilevel stages, including accuracy and reproducibility, low distortion, low overshoot, and action compounding for good spectral discrimination. For the high-level and mid-level stages both high-frequency and low-frequency circuits are used, with a crossover frequency of 800 Hz. The low-level stage is high frequency only, with an 800-Hz high-pass characteristic.

Each stage above has a low-level gain of somewhat over 8 dB, whereby a total dynamic effect of about 16 dB is obtained at low frequencies, 24 dB at high frequencies. A further dynamic action of about 1 dB takes place above the reference level.

The spectral skewing network has the same purpose and function as in the C-type system, except that a spectral skewing action is provided at low frequencies as well. The spectral skewing networks desensitize the SR process to the influence of signal components at the extreme ends of the audio frequency band. This effect is particularly helpful if the recording or transmission system has an uncertain frequency response in these regions. The networks are also important in attenuating subsonic and supersonic interferences of all kinds. The spectral skewing action is compensated in the decoder, resulting in an overall flat frequency response.

Both high-frequency and low-frequency antisaturation networks are provided in the main signal path, again operating in substantially the same way as in the C-type system. There is an effective compounding of the antisaturation effects produced by the antisaturation networks and the spectral skewing networks. In this way the SR process achieves a significant increase in high- and low-frequency headroom.

2 GENERAL PRINCIPLES

2.1 Least Treatment Principle

A design philosophy used in the development of the new system is that the best treatment of the signal is the least treatment. The operating goal of the encoder is to provide fixed, predetermined gains for all frequency components of the signal, with corresponding attenuations in the decoder. If a large signal component appears at a particular frequency or frequencies, then the gains should be reduced at those frequencies only, in accordance with predetermined compression laws for restoration of the signal during decoding. In other words, the compressor should try to keep all signal components fully boosted at all times. When the boosting must be cut back at a particular frequency, the effect should not be extended to low-level signal components at other frequencies.

The audible effect of this type of compression is that the signal appears to be enhanced and brighter but without any apparent dynamic compression effects. (The ear detects dynamic action primarily by the effect of a gain change due to a signal component at one frequency on a signal component at some other frequency, somewhat removed.) If the ear cannot detect dynamic effects in the compressed signal, then 1) it is unlikely that noise modulation effects will be evident in the decoded signal, and 2) it is unlikely that signal modulation effects will be evident in the decoded signal if there should be a gain or frequency response error in the recording or transmission channel.

In the SR process two new methods are used that greatly reduce the circuitry required to achieve the design goal of a full spectrally responsive system. In particular, both fixed and sliding bands are used in a unique combination, called action substitution, that draws on the best features of both types of circuits. A further technique, called modulation control, greatly improves the performance of both the fixed and the sliding bands in resisting any modulation of signal components unless necessary.

The use of the new methods reduces the basic encoder to two frequency bands only (high frequency and low frequency), each with a fixed-band circuit and a sliding-band circuit (this combination being referred to as a stage). When the three-level action-staggering layout is taken into account, five fixed bands and five sliding bands are employed in the spectral recording process.

2.2 Action Substitution

A new type of compression and expansion action that is highly responsive to spectral changes can be achieved by superposing or overlaying the individual characteristics of different types of dynamic action circuits. One circuit may provide a quiescent characteristic or defining umbrella. A further characteristic is hidden until signal components appear that cause the hidden characteristic to be revealed and become active.

For discussion purposes let the gains in a compressor system be arranged such that subthreshold signals pass without attenuation. That is, the maximum possible action is that of providing a certain gain, unity, for instance. Somehow to achieve this gain over as broad a range of frequencies as possible, in the presence of higher level (dominant) signals, is the task of the system.

Thus, in a superposed action compressor circuit, represented in Fig. 2, a signal is fed into a first compressor circuit. The output from this circuit represents the completed part of the total potential action. The input signal minus the completed part is, therefore, the uncompleted part. This is so derived and fed into the next compressor circuit, which has some different characteristic. The output of the second circuit is then added to that of the first, augmenting the action of the first. In an extreme condition, in which the output of the first circuit may be negligible at a particular fre-

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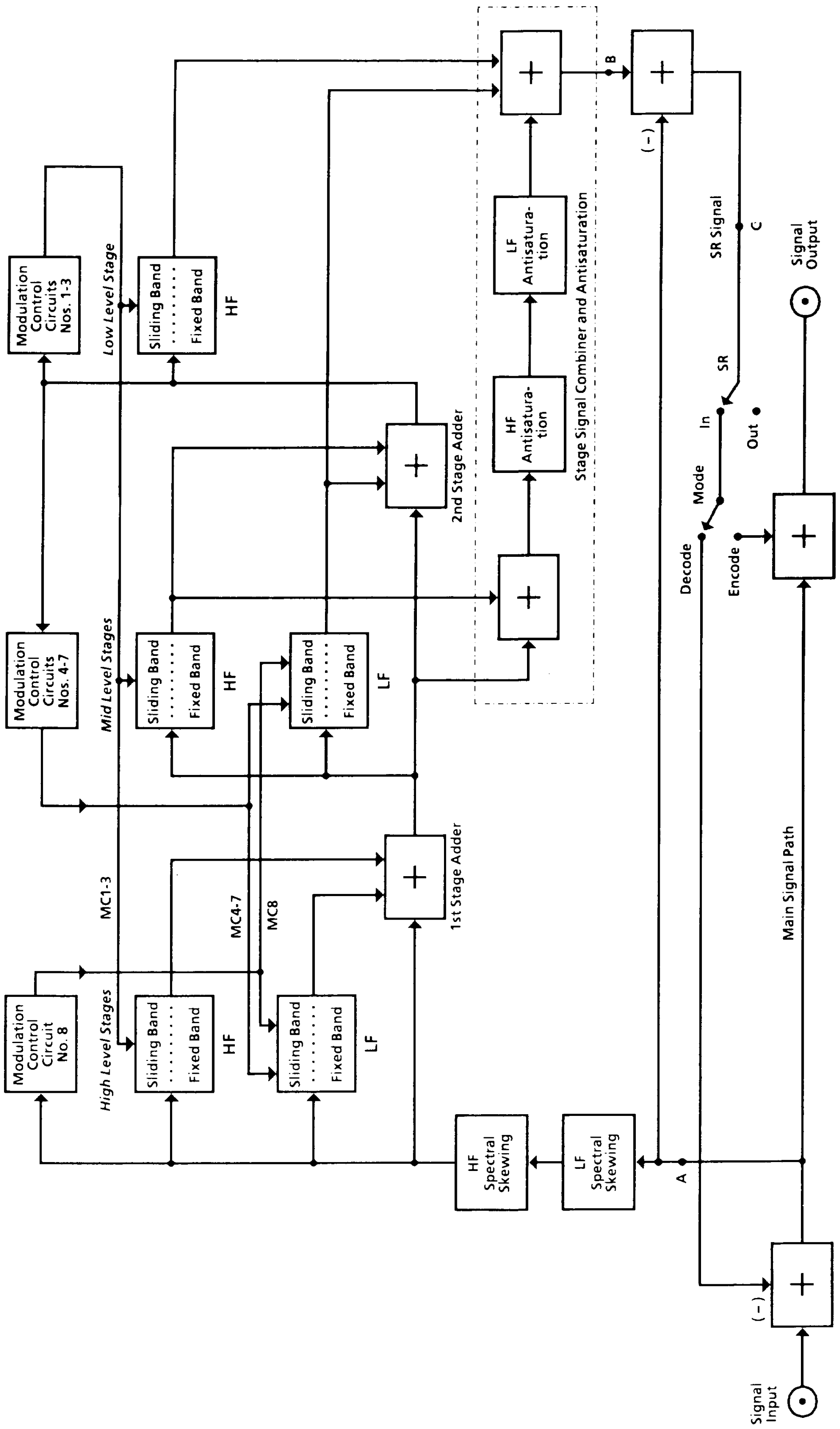


Fig. 1. Basic block diagram—spectral recording process. Diagram shows main features of process: multistage high-frequency and low-frequency dynamic action staggering, action substitution (fixed bands and sliding bands), modulation control, high-frequency and low-frequency spectral skewing, and high-frequency and low-frequency antisaturation.

quency, the action of the second circuit is effectively substituted for that of the first.

The operation of the action substitution compressor can be characterized directly from the above description. With an input signal V_i and an output signal V_o, a first compressor transfer function $F_1(s)$ and a second compressor transfer function $F_2(s)$, each being responsive to the respective applied signal, we have

$$V_o = V_i\,[F_1(s) + F_2(s) - F_1(s)\,F_2(s)] \quad . \tag{1}$$

This equation shows that the overall transfer function is the sum of the individual transfer functions minus their product. In other words, to the extent that the transfer functions may overlap, a factor is subtracted from the sum of the transfer functions.

The above type of action can be achieved with various circuit topologies, the one used in the initial implementation of the SR system being shown in Fig. 3. In this arrangement, the compressor circuits are arranged in a stack. Both circuits are fed in parallel and the output is taken from the top circuit, which is configured as a three-terminal network with terminals a, b, and c. The output of the bottom circuit is fed to the reference terminal c of the top circuit. It can be shown that the signal components at the output terminal b are those specified by Eq. (1).

The usefulness of the superposition technique can be appreciated by consideration of Fig. 4(a) and (b). The advantages of fixed-band compressor circuits [Fig. 4(a)] arise from the fact that all signal frequencies within the band are treated equally, in contrast with sliding-band action [Fig. 4(b)]. Thus the appearance of a dominant signal component actuating the compressor results in a loss of noise reduction effect that manifests itself in a uniform manner throughout the band, 2 dB in the example shown. The loss is not concentrated in any particular frequency region as it is in sliding-band circuits; note the 5-dB loss shown in the example of Fig. 4(b). The main significance of this is that if the recorder or transmission channel has an error in gain and/or frequency response, there is no undue exaggeration of the error at other, nondominant signal frequencies. In sliding-band circuits the amplification effect may be significant (the midband modulation effect, discussed in [2]).

In contrast, the advantages of sliding-band compression and expansion circuits derive from the fact that all signal frequencies are *not* treated equally. In particular, compression, expansion, and noise reduction action are well maintained above the frequency of the dominant signal component in high-frequency circuits, and below the frequency of the dominant signal component in low-frequency circuits. This action maintenance effect, except on a one-to-one basis, is absent in fixed-band circuits

Clearly it would be desirable to have the benefit of fixed-band operation on the stop-band side of the dominant signal frequency and sliding-band operation on the pass-band side. The action substitution technique provides this useful combination. In Fig. 4(c) the response of an action substitution compressor to the signal conditions of the two previous figures is shown. As is seen, the output is primarily from the fixed band for frequencies up to the dominant signal component and from the sliding band above that frequency. Conversely, for a low-frequency stage the output is from the fixed band for frequencies down to a low-frequency dominant component and from the sliding band below that frequency. This cooperative effect is particularly useful in the level region from the circuit threshold up to some 20 dB thereabove.

In the SR process, action substitution operation is used in both the high- and the low-frequency circuits. Thus both fixed-band and sliding-band dynamic actions are used in each of the five stages, a total of ten compressor circuits. While there is an effective interaction of the fixed and sliding bands in any particular stage, all of the stages operate independently. Depending on the levels and spectral conditions in each stage, fixed-band operation is used whenever it provides best performance; sliding-band operation is substituted whenever it has an advantage. The substitution is effective on a continuous and frequency-by-frequency basis.

Even though the frequency division of the stages is nominally 800 Hz, the use of what are effectively single-pole band-defining filters results in a significant overlap region between the high- and low-frequency stages. The high-frequency stages extend their effects down to about 200 Hz, the low-frequency stages extending their effects up to about 3 kHz. This overlap, together with the use of action substitution, contributes to the achievement of a very good spectral tracking effect

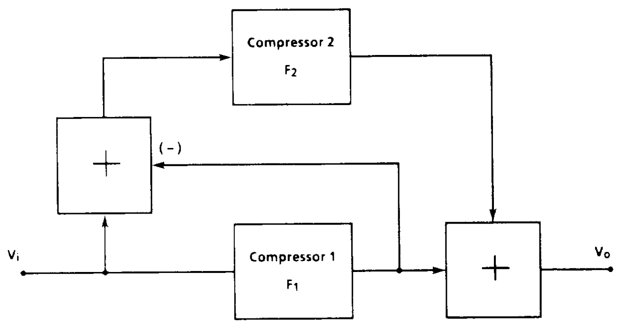


Fig. 2. Basic layout of action substitution compressor.

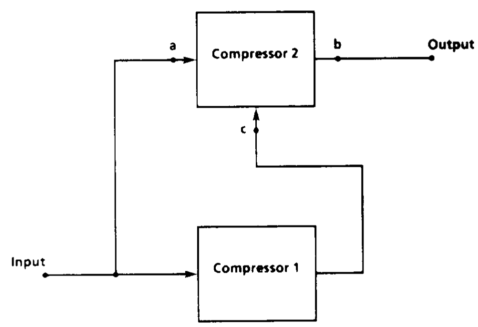
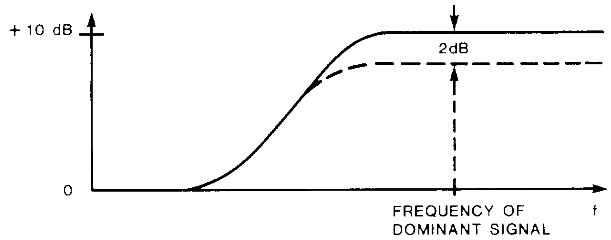


Fig. 3. Another action substitution compressor configuration.

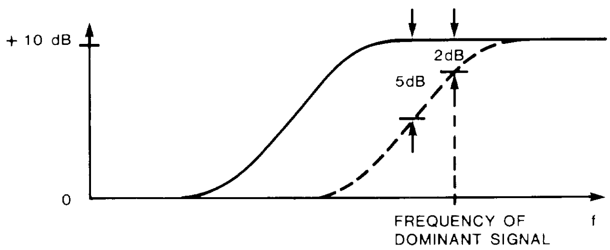
under all frequency and level conditions. The practical significance is that an excellent noise reduction effect is obtained in the presence of signals, and that the system has a remarkable tolerance to gain and frequency response errors in the signal channel.

A further aspect of action substitution relates to the transient recovery characteristics of the system. A fixed-band compressor circuit has a recovery time that is essentially independent of frequency, at least in the pass band. A sliding-band circuit has a fast recovery time for nondominant signals at the pass-band end of the spectrum, and a slow recovery time for nondominant signals at the stop-band end of the spectrum. The choice of integrator recovery times is therefore a matter of compromise between this recovery time situation and the amount of steady-state and modulation distortion obtained. The compromise is made much easier by the use of the action substitution technique. In particular, the fixed band provides a definite and rapid recovery time for the overall system, so that the sliding band can employ longer time constants than would otherwise be desirable. This results both in low modulation distortion and a fast recovery time.

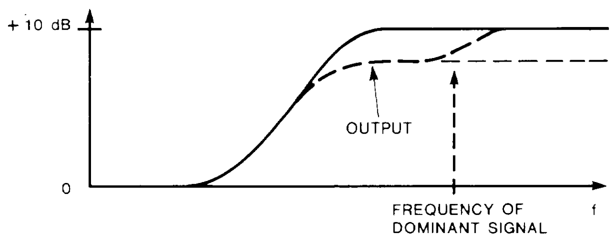
Thus the action substitution technique provides the advantages of fixed and sliding-band circuits while avoiding their disadvantages. In other words, there is a significantly improved adherence to the ideal of least



(a) Fixed-band compressor characteristic.



(b) Sliding-band compressor characteristic.



(c) Action substitution compressor characteristic.

Fig. 4. Types of compressor characteristics.

signal treatment. In the level region somewhat above the circuit threshold the signal more closely approaches fully boosted conditions in the encoding mode, with a consequently improved noise reduction effect in the decoding mode. For signals at higher levels, the technique of modulation control, described below, is employed.

2.3 Modulation Control

In the A-type, B-type, and C-type systems the signal from the side chain is highly limited under high-level signal conditions. This high degree of limiting, beginning at a low-level threshold, is responsible for the low distortion, low overshoot, and low modulation distortion which characterize these systems.

A closer examination shows that it is unnecessary to utilize such a low threshold and such a strong limiting characteristic under certain signal conditions. In particular, whenever the side chain signal departs from an in-phase condition with respect to the main path signal, then the threshold can be raised. Moreover, after an appropriate degree of limiting has taken place at a given frequency (in order to create the desired overall compression law), then it is unnecessary to continue the limiting when the signal level rises even further. Rather, the level of the side chain signal can be allowed to rise as a function of a further increase in signal level, whereby it stabilizes at some significant fraction of the main path signal level.

In the fixed-band portions of the spectral recording process the above arrangement results in conventional performance in the pass-band (in-phase) frequency region. However, in the stop-band region the modulation control scheme causes the limiting threshold to rise and the degree of limiting to be reduced. The possibility of doing this can be appreciated by consideration of the phasor diagrams of the two conditions shown in Fig. 5. In the pass-band (in-phase) condition the side chain signal and the main path signal add directly. Therefore a relatively low threshold must be maintained at all pass-band frequencies [Fig. 5(a)]. However, in the stop band the effective amplitude contribution of the side chain signal may be minimal due to the phase difference between it and the main path signal. Because of this it is possible to raise the threshold significantly and to reduce the limiting strength once the desired amount of attenuation has been obtained at a given frequency [Fig. 5(b)]. The result is that large signals in the stop band do not cause signal modulation in the pass band and consequently create an impairment of the noise reduction effect achieved during decoding.

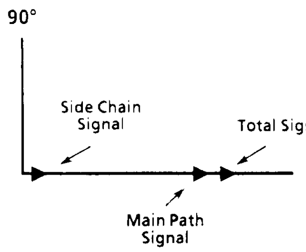
Similar considerations apply in the SR sliding-band circuits. By way of introduction, in the B-type and C-type sliding-band circuits a variable filter follows a fixed filter, which has proved to be an efficient and reproducible arrangement. At frequencies outside the pass band a pure two-pole filter would result in overall amplitude subtraction from the main path signal because of the large phase angles created. Therefore the type of filter which has been employed is only quasi-two pole (a single-pole fixed filter plus a variable-shelf characteristic).

The same type of arrangement is used in the spectral recording process, with a one-octave difference (in the stop-band direction) between the variable filter turnover frequency (under quiescent conditions) and the fixed filter cutoff frequency. Above the threshold at a particular frequency the variable filter slides to the turnover frequency needed to create the overall (main path plus side chain signal) compression law. As the input level rises, and once an overall gain of about unity is obtained—when the variable filter cutoff frequency is about two to three octaves above the dominant signal frequency—there is no reason for further sliding of the variable filter. At this point the modulation control arrangement counteracts the sliding. As with the fixed-band circuits, this technique prevents unnecessary modulation of the signal.

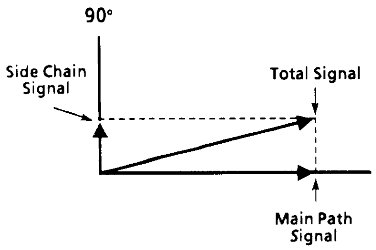
The above effects in both the fixed and the sliding bands are created by circuits called modulation control circuits. Suitably filtered or frequency-weighted signals from the main signal path are rectified, and in some cases smoothed, and are fed in opposition to the control signals generated by the control circuits of the various stages. The result at higher signal levels, relatively (beginning at about 20 dB above the threshold of the relevant compressor circuit), is to tend to create a balance or equilibrium between the compressor circuit control signals and the modulation control signals. Under these conditions there is a significantly reduced gain reduction or sliding of the variable filters as a function of increasing input signal levels.

Fig. 6 illustrates the action of modulation control with a high-frequency fixed-band compressor circuit. The circuit has a low-level gain of about 8 dB and an 800-Hz high-pass characteristic. Fig. 6(a) shows the response of the circuit in the absence of modulation control. Ideally there should be no attenuation in response to a 100-Hz signal because the overall shape of the envelope is such that there is negligible signal boosting at 100 Hz. Nevertheless, with a conventional compressor circuit as shown here, when the 100-Hz signal increases in level, there is a reduction of low-level signal boosting over the whole frequency band. The unnecessary attenuation has two effects: 1) substantial noise reduction action is lost during expansion, and 2) when the amplitude of the 100-Hz signal varies, it can modulate low-level signal components at higher frequencies, resulting in possible incorrect restoration of the signal by the expander if the recording channel has an irregular frequency response in the vicinity of 100 Hz.

Fig. 6(b) shows the operation of the same circuit with modulation control. A greatly reduced attenuation occurs when the 100-Hz signal is varied over the same

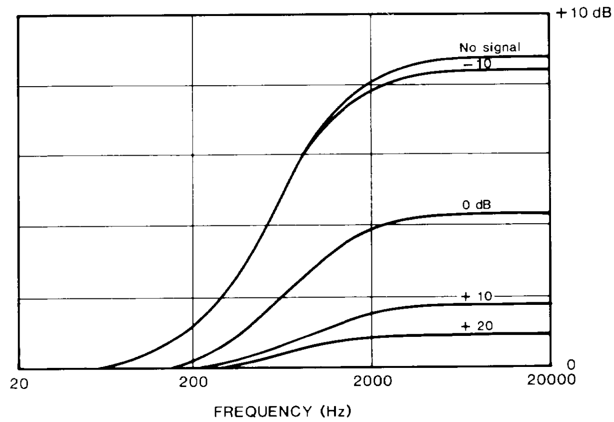


(a) In the pass-band, a low threshold and strong limiting characteristic are required.

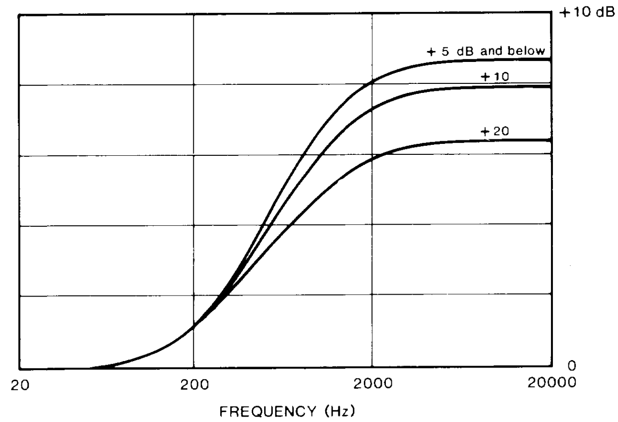


(b) In the stop-band, phase shift results in the side chain signal having negligible influence on the total signal amplitude; therefore a higher threshold and weaker limiting can be used.

Fig. 5. Phasor diagrams, dual path compressor.



(a) Frequency response curves with 100-Hz signal at the levels indicated, no modulation control.



(b) Same as 6(a), with modulation control.

Fig. 6. Effect of modulation control on fixed-band compressor circuit.

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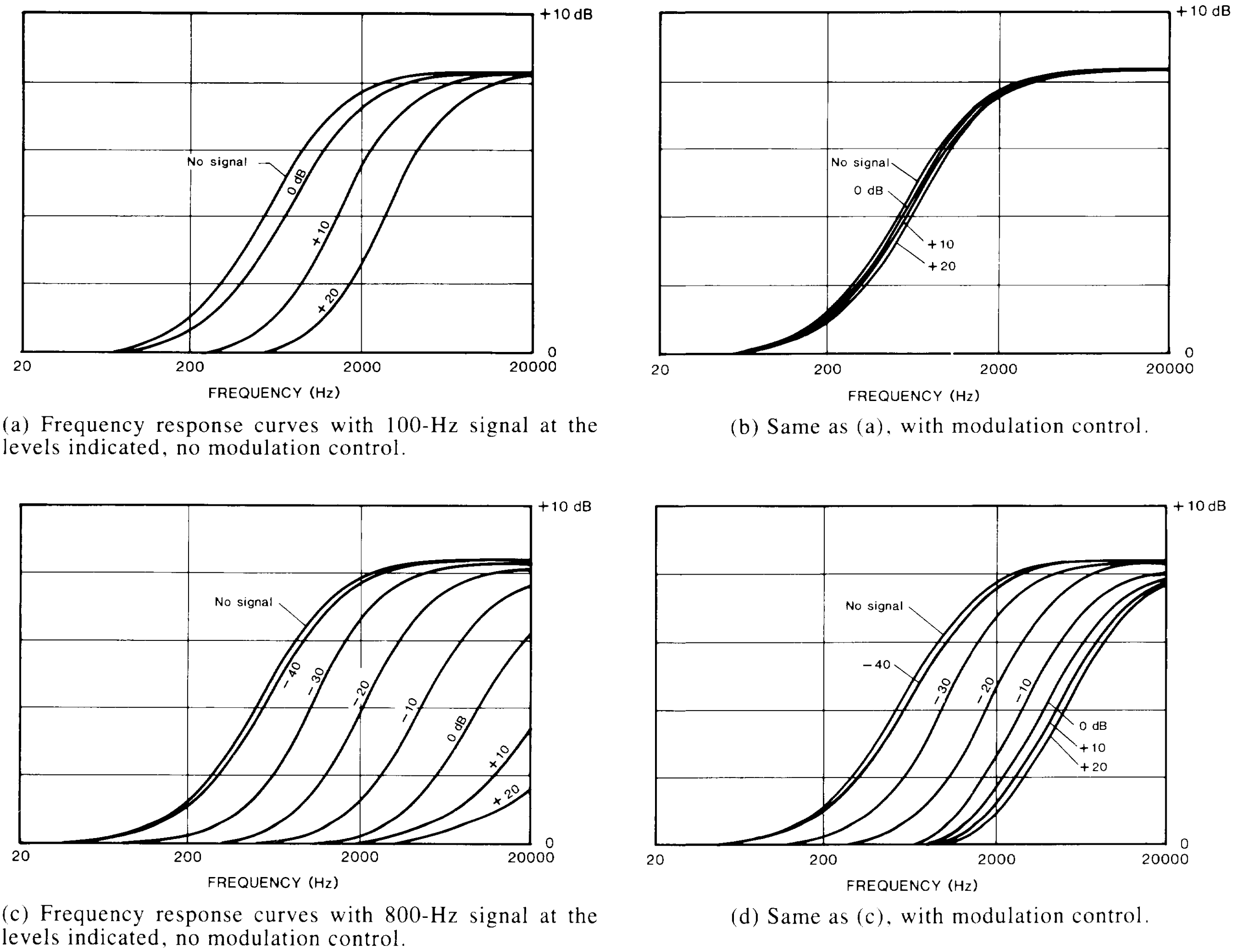
range of levels as in Fig. 6(a). Thus a significant immunity to strong signals in the stop-band frequency region is achieved, the effect decreasing as the dominant signal frequency approaches the pass-band frequency region of the circuit.

In Fig. 7(a) the operation of a sliding-band circuit under comparable conditions is shown. As with the fixed-band circuit, ideally there should be no sliding in response to a strong 100-Hz signal. Nonetheless, as the 100-Hz signal increases in level, the band slides upward. As with the fixed-band circuit, the unnecessary sliding results in a loss of noise reduction action and the modulation of signals at higher frequencies when the sliding band varies under the control of the 100-Hz signal.

Fig. 7(b) shows the operation of the same circuit with modulation control. Minimal sliding occurs when the 100-Hz signal is varied over the same range of levels as in Fig. 7(a). Thus the sliding-band compressor is also made essentially immune to strong signals outside its pass band.

The effect of modulation control is further illustrated by Fig. 7(c) and (d), except that the frequency of the dominant signal is changed to 800 Hz, a frequency within the pass band of the circuit. Ideally, sliding is required to go only so far as not to boost the 800-Hz signal above the 0-dB reference level. Thus in the Fig. 7(c) response, without modulation control, the sliding produced by the 800-Hz signal at levels above −10 dB is excessive. Fig. 7(d) shows the response of the circuit with modulation control: sliding at and above the 0-dB level is greatly reduced. The effect is progressively reduced for low signal levels, but is observable to some extent at the −10-dB level.

The use of modulation control techniques also has advantages under transient conditions, both from their use in the steady-state circuits and also because of their use in the overshoot suppression circuits. Modulation control generally prevents any further fixed-band attenuation or sliding of the variable filter than is required to respond to a given signal situation. Therefore 1) signal modulation is reduced, 2) the SR process is rendered very tolerant of channel errors, 3) subsequent noise modulation during decoding is reduced, and 4) recovery from transient signal conditions is faster. The electronic reality in both the steady-state circuits and transient control circuits is that the integrator capacitors are prevented from charging to voltages as high as they normally would in the absence of modulation control. With lower fully charged voltages, recovery is faster.



(a) Frequency response curves with 100-Hz signal at the levels indicated, no modulation control.

(b) Same as (a), with modulation control.

(c) Frequency response curves with 800-Hz signal at the levels indicated, no modulation control.

(d) Same as (c), with modulation control.

Fig. 7. Effect of modulation control on sliding-band compressor circuit, with a signal in the stop-band (a), (b), and a signal in the pass-band (c), (d).

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The modulation control aspects of the SR process result in an encoding action which is remarkably free of noticeable signal-related modulation effects. Working together with action substitution, modulation control contributes to the goal of least treatment, in providing a highly boosted, audibly stable encoded signal.

2.4 Overshoot Suppression

A side effect of the modulation control scheme is that at high signal levels the amplitudes of the signals from the several stages are relatively high in comparison with the situation in the A-type, B-type, and C-type systems. Because of this it is not possible to employ simple overshoot suppression diodes as in these previous systems. A more flexible but necessarily more complex scheme, operating directly upon the control signals, is used.

In common with the A-type, B-type, and C-type systems, the SR process features overshoot suppression thresholds that are significantly higher than the steady-state thresholds; this results in low modulation distortion. The overshoot suppression thresholds are set about 10 dB above the relevant steady-state thresholds. The net result is that for most musical signals the overshoot suppressors rarely operate; the compressors are controlled by well-smoothed signals. When the suppressors do operate, the effect is so controlled that modulation distortion is minimal.

Under extreme transient conditions, such as from a subthreshold signal situation, the overshoot suppression threshold is set at its lowest point. The overshoot suppression effects are then phased in gradually as a function of increasing impulse level.

Under relatively steady state but nonetheless changing signal conditions the overshoot suppression effects are gradually phased out as a function of increasing signal levels, this action further ensuring low overall modulation distortion from the system. The phasing out effect is achieved by increasing the overshoot suppression thresholds. The thresholds are controlled by signals that are the same as or derived from the modulation control signals used to control the steady-state characteristics, whereby a tracking action between the transient and steady-state behavior is obtained. This arrangement results in both well-controlled overshoots and low modulation distortion.

Both primary and secondary overshoot suppression circuits are employed, the latter acting as fall-back or long-stop suppressors. In the high-frequency circuits the secondary overshoot suppressors improve the performance just inside the stop band—that is, in the 400–800-Hz region. In the low-frequency circuits these additional suppressors improve the performance under extreme complex signal conditions (such as high-level low and mid-frequency transient signals in combination with high-level high-frequency signals). In the low-frequency circuits a further overshoot suppressor (LF O/S) is used for very-low-frequency signals. This is a very gentle, slow-acting circuit which reduces low-frequency transient distortion.

2.5 Staggered Action Multilevel Format

The principles discussed above are incorporated into each stage of the multilevel staggered action encoder and decoder. See [2] for a detailed discussion of staggered action circuits. In the SR system two stages are employed at low frequencies, three at high frequencies. The thresholds used are approximately −30 dB, −48 dB, and −62 dB below reference level (about 20 dB below SR peak signal level). In the series-connected staggered action format there is a compounding of the actions of the individual stages; the transfer functions of the several stages are multiplied, whereby the dB characteristics add. In this way a large total dynamic action can be achieved with low modulation distortion, low overshoot, and good manufacturing reproducibility. An important additional result is that there is an overall steepness enhancement of the frequency discrimination abilities of the circuit, further inhibiting signal modulation and noise modulation effects.

2.6 Spectral Skewing

The spectral skewing networks employed in the SR process comprise both high-frequency and low-frequency sections, with the same rationale and mode of operation as discussed in [2]. The spectral distributions of the signals processed by the encoder are altered or skewed, well within the pass band, such that the encoder action is significantly less susceptible to the influence of signals beyond the abrupt roll-off frequencies of the spectral skewing networks.

The high-frequency network is a low-pass filter with an attenuation characteristic similar to that of a 12-kHz two-pole Butterworth filter, but with a limiting attenuation of about 35 dB (that is, a shelf). The low-frequency network is a 40-Hz high-pass filter, connected in series with the high-frequency network, also with a two-pole Butterworth-like characteristic, but with about a 25-dB limiting attenuation. These shelves do not interfere with the attenuation within the audio band, but provide phase characteristics that are essential in the decoding mode.

2.7 Antisaturation

The general principle of antisaturation was described in [2]. Briefly, by placing a fixed attenuation network, usually a shelf, in the main path of a dual path compressor, it is possible to create an effective antisaturation characteristic at the extremes of the audio band without undue adverse effects on the noise reduction achieved during decoding. In the SR process high- and low-frequency networks are operative above about 5 kHz and below about 100 Hz, respectively. In addition, the spectral skewing networks have a secondary but very useful antisaturation effect, especially at very low and very high frequencies.

3 BLOCK DIAGRAMS

3.1 Basic Block Diagram

As mentioned previously, Fig. 1 shows the basic

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layout of an SR processor. While the whole system comprises an encoder and a complementary decoder, the figure as drawn shows a switchable configuration, which generally is the most useful one. The main signal path transmits high-level signal components. To this is added in the encoding mode, and subtracted in the decoding mode, the output of the side chain circuitry, called the SR signal, point C. The stage circuits, as well as the spectral skewing networks and antisaturation networks above, are driven from point A. See [1] for a mathematical explanation of these arrangements.

A secondary main path which does not include any antisaturation is employed as the basis of the side chain, to which the outputs of the high-level and mid-level stages are added in the first-stage and second-stage adders, respectively. The low-level stage and modulation control circuits 1–7 are driven directly from the output of the second-stage adder. Modulation control circuit 8 is driven from the output of the spectral skewing network, as will be discussed.

The antisaturation effects are created in the dashed block labeled stage signal combiner and antisaturation. The arrangement shown provides a high-frequency deemphasis effect on the secondary main path signal, which includes the output of the high-level stages, and on the high-frequency mid-level stage signal. This deemphasis is effective not only on the steady-state aspects of these signals, but on all transient effects as well. The output of the low-level stage is then added directly. For low-frequency antisaturation the low-frequency deemphasis is effective on the secondary main path signal, including the high-level stage outputs. The low-frequency mid-level signal is then added directly. The mathematical basis for these arrangements is provided in [2].

With the final combination of signals in the last adder, an SR encoded signal appears at point B. The encoded signal can be considered to comprise an unmodified component from the input plus an SR signal which carries all of the SR characteristics. Thus the SR component can be derived by subtracting the unmodified input signal at point A from the SR encoder output at point B. This provides an SR signal at point C that can be handled and switched the same way as in the A-type system. This simplifies practical use of the system.

3.2 Modulation Control Circuits

Fig. 1 shows how the inputs of the various modulation control circuits are connected and how the resultant signals are distributed. Modulation control signals MC1–MC7 are derived from the output of the second-stage adder. In this way the modulation control signals begin to have a significant influence at relatively low levels, such as at −30 dB (because of the contributions of the HLS and MLS stages); the phase relationships between the modulation control signals and the signals in the control circuits of the several stages are also optimized. In the generation of MC8, which is used for low-frequency stage overshoot-suppression inhibition under high-frequency transient signal conditions, the influences of the HLS and MLS stages are undesirable. MC8 is therefore derived from the first feed point of the stages, just following the spectral skewing networks.

Fig. 1 also shows the distribution scheme of the modulation control signals. MC1–MC3 are used for the high-frequency stages; MC5–MC8 are used for the low-frequency stages.

In Fig. 8 the basic layout of the modulation control circuits is shown. MC1 controls the high-frequency sliding-band circuits. The signal from the takeoff point is fed through a 3-kHz single-pole high-pass filter, full-wave rectified (all rectifiers in the system are full wave), and fed in opposition to the control signals generated by the high-frequency stages. An all-pass phase shift network is used to optimize the phase of the MC signal in relation to the stage control signal at low frequencies; this reduces control signal ripple. MC1 is also smoothed by a two-stage 1-ms integrator and is employed, as MC2, to oppose the operation of the high-frequency sliding-band overshoot suppression circuits; the overshoot suppression thresholds thereby track the steady-state thresholds. MC2 must be smoothed because the phase relationships of MC1 and the signals in the stages vary (because of the sliding-band action) throughout the audio band, being a function of frequency and level.

MC3 controls the high-frequency fixed-band circuits. The signal from the takeoff point is weighted by cascaded 400-Hz and 800-Hz single-pole low-pass filters, rectified, and fed in opposition to both the steady-state and the transient control circuits of the high-frequency fixed-band circuits. There is no need to provide a smoothed MC signal for the overshoot suppressors of the high-frequency fixed-band stages because a fixed phase relationship exists between the stage signals and the control signals throughout the audio band.

MC4 controls the sliding-band circuits of the low-frequency stages. The signal from the takeoff point is fed through a 200-Hz single-pole low-pass filter, rectified, and fed in opposition to the sliding-band control signals generated in the stages. The phase of the modulation control signal is optimized by the use of an all-pass phase shifter, as with MC1; low-frequency control signal ripple is thereby reduced. MC4 is smoothed by a two-stage 2-ms integrator to form MC5; this signal is used to control the low-frequency sliding-band overshoot suppressors.

MC6 controls the low-frequency fixed-band circuits. The signal from the takeoff point is weighted by cascaded 800-Hz and 1.6-kHz single-pole high-pass filters, rectified, and used to oppose the steady-state fixed-band control signals. MC6 is also smoothed in a two-stage 2-ms integrator, forming MC7, which is used to control the low-frequency fixed-band overshoot suppressors. This smoothing is necessary in the low-frequency fixed-band stages because, unlike the situation in the high-frequency fixed-band stages, there is no fixed phase relationship between the stage signals and the overshoot suppression signals. MC7 is also used in a supplemental way to control the low-frequency

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sliding-band overshoot suppressors.

MC8 is used to control the overshoot suppression circuits of both the fixed- and the sliding-band low-frequency circuits. MC8 compensates for the fact that no frequency weighting is used in the generation of the low-frequency primary overshoot suppression signals. High-frequency transient signal components are detected and used to oppose the operation of the LF primary overshoot suppression circuits. The signal from the MC8 takeoff point is fed through a 5-kHz high-pass filter, rectified, double differentiated with 15-μs time constants, and peak held with about a 30-ms time constant. The resultant high-frequency transient modulation-control signal MC8 is then employed to oppose the low-frequency overshoot suppression action.

3.3 High-Frequency Stage

Fig. 9 shows both the steady-state and the transient control aspects of the high-frequency stages. The diagram shows only the basic parameter determining elements; the practical circuits of course contain other details such as buffering, amplification, and attenuation. The high-level, mid-level, and low-level stages have the same basic block diagrams and schematics. The main distinctions are that the ac and dc circuit gains are increased for the mid- and low-level stages.

Referring to the block diagram, each stage comprises a fixed-band section on the bottom and a sliding-band section on the top, each with its own control circuits. The fixed- and sliding-band circuits are fed in parallel, and the output signal is taken from the sliding-band circuit. The sliding-band variable filter is referenced to the output of the fixed band; that is, the fixed-band output is fed directly to the bottom end of the sliding-band variable resistance RV_S. This connection results in the action substitution operation discussed previously. At any given frequency the overall output will be the larger of, or some combination of, the fixed- and sliding-band contributions. If there is a signal situation in which the fixed-band output is negligible, then the sliding band predominates. Conversely, if there is little or no sliding-band contribution, the output from the fixed band will still feed through to the output through RV_S. In this way the action of one circuit augments that of the other, and, as the occasion requires, may be substituted for that of the other.

The incoming signal is fed through an 800-Hz single-pole band-defining filter. This is followed by a 400-Hz single-pole filter which attenuates the low-frequency signal levels fed to both the fixed- and the sliding-band circuits; this reduces waveform distortion and complex-signal transient distortion at high signal levels. The filter also forms part of the fixed-band control signal weighting network. The output signal is taken from the sliding-band stage and is fed through a 400-Hz network having a reciprocal characteristic to that of

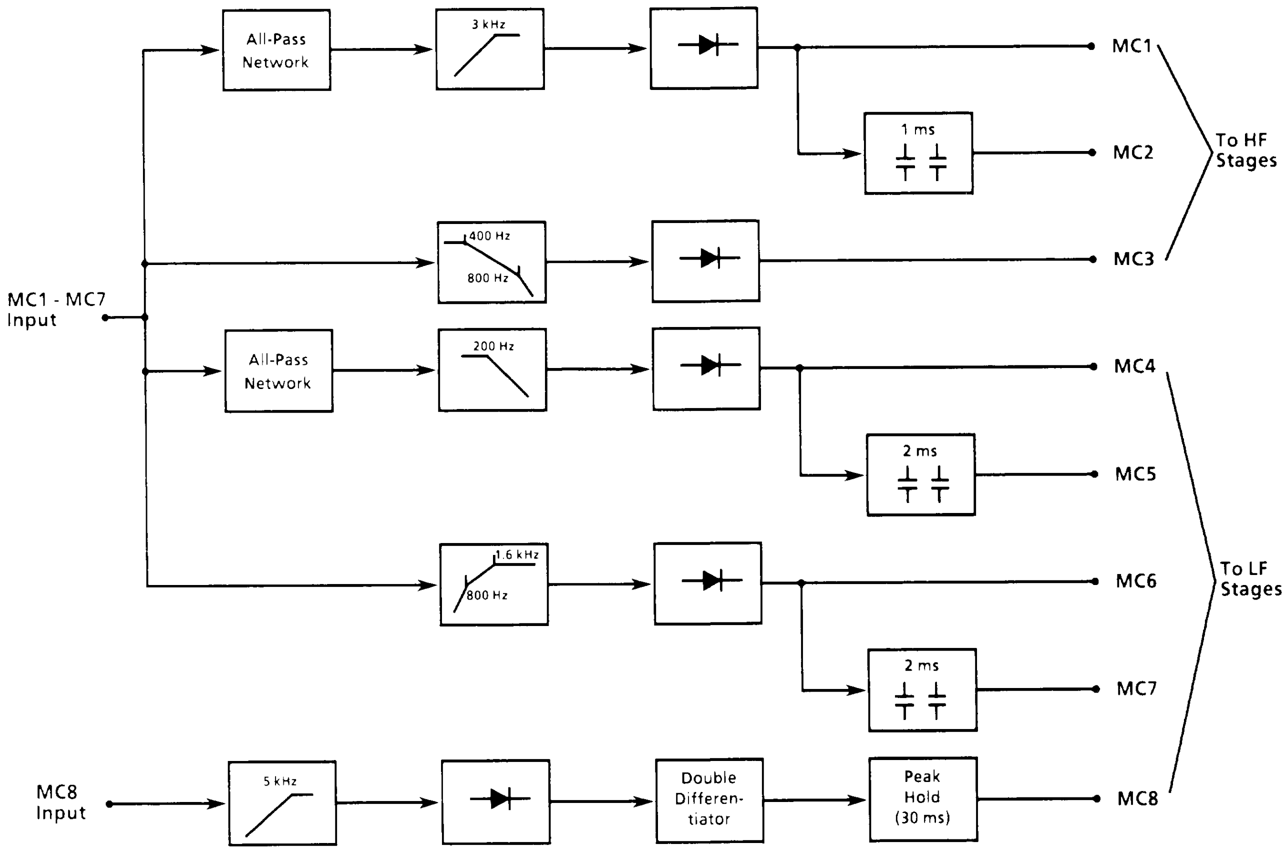

Fig. 8. Basic layout of modulation control circuits. These circuits reduce modulations of the gains and frequency response characteristics used in system, especially at signal levels significantly above the compressor thresholds.

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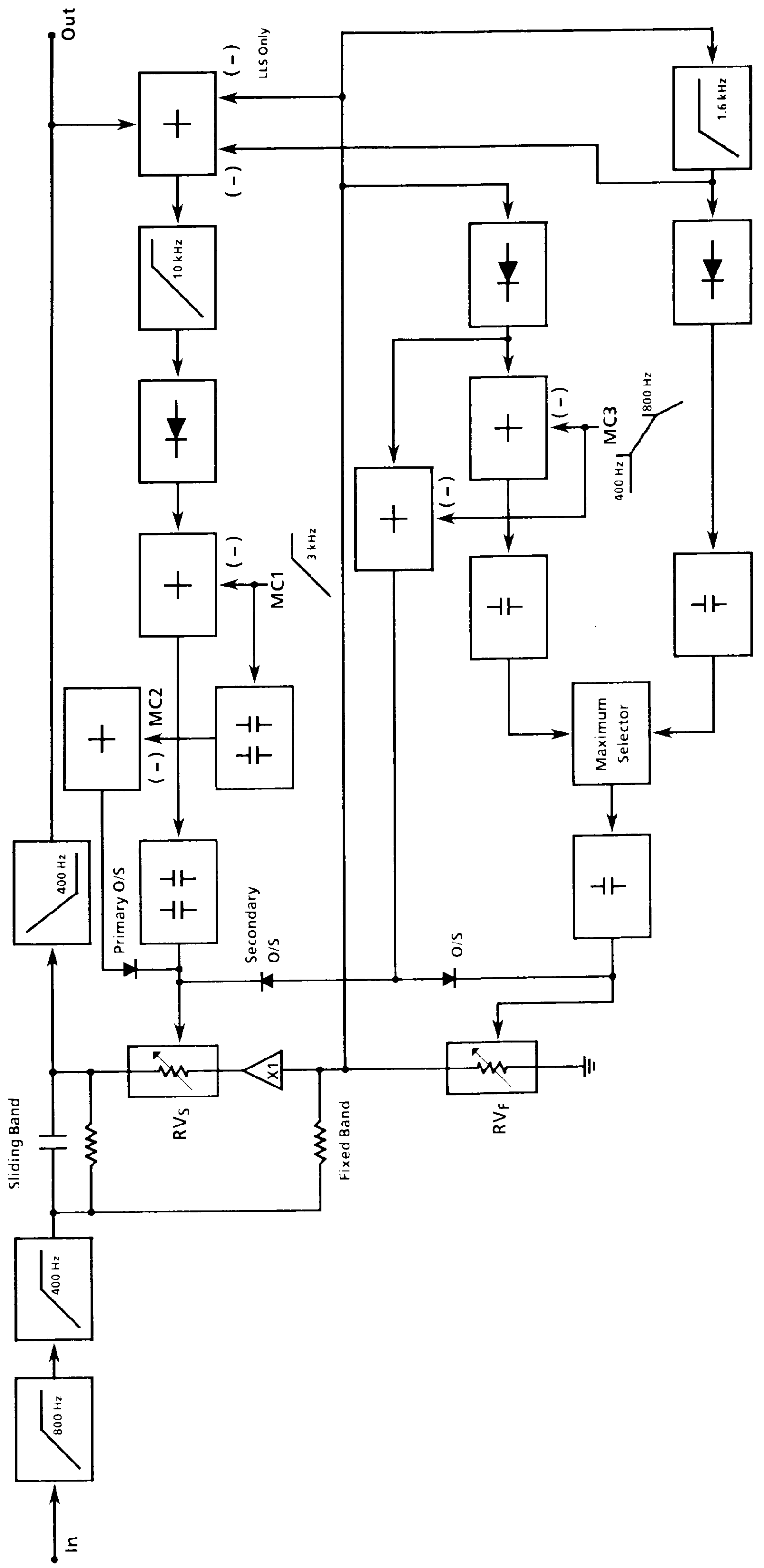

Fig. 9. High-frequency stage block diagram.

the 400-Hz high-pass filter at the input. Thus the overall quiescent (subthreshold) frequency response of the circuit is that of a single-pole 800-Hz high-pass network. The low-frequency stages have a complementary 800-Hz single-pole low-pass characteristic, which overall results in optimal combination of the signals from the high- and low-frequency stages.

The fixed-band output from RV_f (that is, variable gain circuit) is fed to two control circuits, the main control circuit (middle of Fig. 9) and the pass-band control circuit (bottom of Fig. 9). In the main control circuit the signal is rectified and opposed by the modulation control signal MC3. The resulting dc signal is smoothed by an integrator circuit with a 15-ms time constant (the overall steady-state control signal characteristic in this and all other stages is average responding). The control signal is then fed to one input of a maximum selector circuit, which passes to its output the larger of two signals applied to the input.

The fixed-band output is also fed to the pass-band control circuit (bottom of Fig. 9), which comprises a 1.6-kHz single-pole high-pass filter, a rectifier, and a smoothing circuit (15 ms). The pass-band control signal is applied to the other input of the maximum selector circuit. The output of the maximum selector circuit is further smoothed by a 160-ms time constant and is used to control the fixed-band variable resistance RV_f or other variable gain means.

The dual control circuit arrangement described above is employed to obtain optimal performance under both simple-signal (single dominant signal) and complex-signal (more than one dominant signal) situations. The modulation control signal MC3 is optimized in frequency weighting and amount for simple signal conditions, in which the modulation control action is most useful. Under complex signal conditions, however, the modulation control signal developed becomes larger, and the subsequent modulation control action is then greater than necessary; that is, the dc control signal output from the main control circuit is less than required. Under this condition the control signal from the pass-band circuit is phased in, via the maximum selector circuit, to control the overall action of the fixed-band compressor circuit.

The output of the fixed band is fed through a buffer with an overall gain of unity to provide the reference for the sliding band filter. This is the only signal output of the fixed-band compressor circuit.

The sliding-band control signal is derived from the stage output (top of Fig. 9). The signal is fed through a single-pole high-pass weighting network (about 10 kHz, different for each stage) and is rectified. The rectified signal is opposed by modulation control signal MC1. Since MC1 also has a single-pole high-pass characteristic, the ratio between the rectified control signal and MC1 monitors the signal attenuation (this ratio creates an end-stop effect on the sliding-band action). The result is smoothed first by a time constant of about 5 ms (different for each stage), and finally by a time constant of 80 ms. The smoothed control signal is then used to control the sliding-band variable resistance RV_s. A single control circuit suffices in the sliding-band circuit because the 10-kHz high-pass control weighting network tends to offset the effect of complex signals on the modulation control voltage developed (MC1).

A modification is made in the sliding-band control characteristic at low levels. Signals from the fixed-band circuit are combined in opposition with the sliding-band output signal (combining circuit at right of Fig. 9). The effect is in the direction of simulating the derivation of the sliding-band control voltage from the voltage across the sliding-band variable filter only (that is, from the voltage across RV_s). This tends to raise the sliding-band threshold at high frequencies, which reduces unnecessary sliding of the band. (The 10-kHz control weighting network provides the correct amount of control signal for the variable filter at medium and high levels, but it produces the undesirable side effect of lowering the threshold at high frequencies. The differential control signal derivation method counteracts this threshold-lowering effect.)

The overshoot suppression (O/S) arrangements are also shown in Fig. 9. In the high-frequency circuits a general feature is that unsmoothed rectified signals from the control circuit rectifiers are opposed by appropriate modulation control signals and are fed via diodes to the final integrator circuits. The low-frequency arrangements follow the same pattern, with some modifications.

Referring to the middle of the diagram, in the high-frequency fixed-band circuit the overshoot suppression signal is derived from the rectifier of the main control circuit. As with the steady-state control signal, the rectified signal is opposed by MC3, so that the overshoot suppression threshold is appropriate for conditions in the steady-state regime. The resultant overshoot suppression signal is coupled by a diode to the final integrator circuit.

In the sliding-band circuit (top of Fig. 9) two overshoot suppression signals are used, primary and secondary. The primary overshoot suppression signal is derived from the control circuit rectifier and opposed by MC2, a smoothed version of MC1 (MC1 controls the steady-state characteristics). The smoothing is necessary because, unlike the situation in the fixed-band circuit, there is no constant and favorable phase relationship between the signal in the control circuit and MC1 (because of the sliding band); the smoothing enables reliable bucking action to take place.

The secondary overshoot suppressor supplements the action of the primary overshoot suppressor under certain conditions. The primary overshoot suppression signal is derived from the same rectifier used in the steady-state control circuit, with the consequence not only of economy but of a favorable phase relationship between the overshoot suppression impulse and the signal to be controlled; this results in low transient distortion. However, the control circuit frequency weighting responsible for this situation also causes a reduction of

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control signal amplitude with falling frequency. A dc bias is used in the overshoot suppression circuit to create the required suppression threshold; when the signal amplitude in the overshoot suppression circuit decreases, the bias results in the overshoot suppression effect falling away faster than the signal amplitude. For frequencies below about 400 Hz a reduced overshoot suppression effect is appropriate because of the attenuation and phase shift of the stage input filter (see Fig. 5). However, in the 400–800-Hz region there is an overshoot suppression deficiency; this is compensated by feeding an appropriate amount of overshoot suppression signal from the fixed-band circuit into the sliding-band circuit. This supplemental signal is called the secondary overshoot suppression signal.

Regarding recovery times, the use of action substitution and modulation control both contribute to rapid action, as already mentioned. Nonetheless, reverse-biased recovery speed-up diodes are used in a fairly gentle way (series resistors) to provide a further increase in speed.

3.4 Low-Frequency Stage: Steady-State Aspects

Fig. 10 shows only the steady-state layout of the low-frequency stages. As with the high-frequency stages, only the basic parameter determining elements are shown. The high-level and mid-level low-frequency stages have the same block diagrams and circuits, but the ac and dc gains are increased for the mid-level stage; there are also some other minor differences.

Referring to the block diagram, certain similarities and differences may be noted with respect to the high-frequency diagrams. The dual-layer arrangement of the fixed band on the bottom and the sliding band on the top is similar. However, the sliding band acts downward, using a simulated inductance (gyrator circuit). As with the high-frequency stages, the fixed- and sliding-band circuits are fed in parallel, and the output signal is taken from the sliding-band circuit. The fixed-band output is coupled to the bottom of the sliding band to provide the action substitution operation described previously.

A notable difference from the high-frequency circuit is that the fixed 800-Hz band determining filter follows, rather than precedes, the variable filter. This arrangement has several advantages: 1) overshoot suppression signals can be generated without the delay inherent in a low-pass filter, resulting in lower transient distortion; 2) any transient distortion produced by the circuit is attenuated by the 800-Hz low-pass filter; and 3) noise generated by the gyrator circuit is attenuated by the filter. The price to be paid for these advantages is the resulting high signal levels that the variable resistances RV_f and RV_s must be capable of handling at high frequencies. (There is no active attenuation at all at very high frequencies, since sufficient passive attenuation and phase shift are provided by the 800-Hz low-pass filter.) Special control arrangements, called high mode, comprising complementary bootstrapping and control circuit gain boosting, enable the fixed- and sliding-

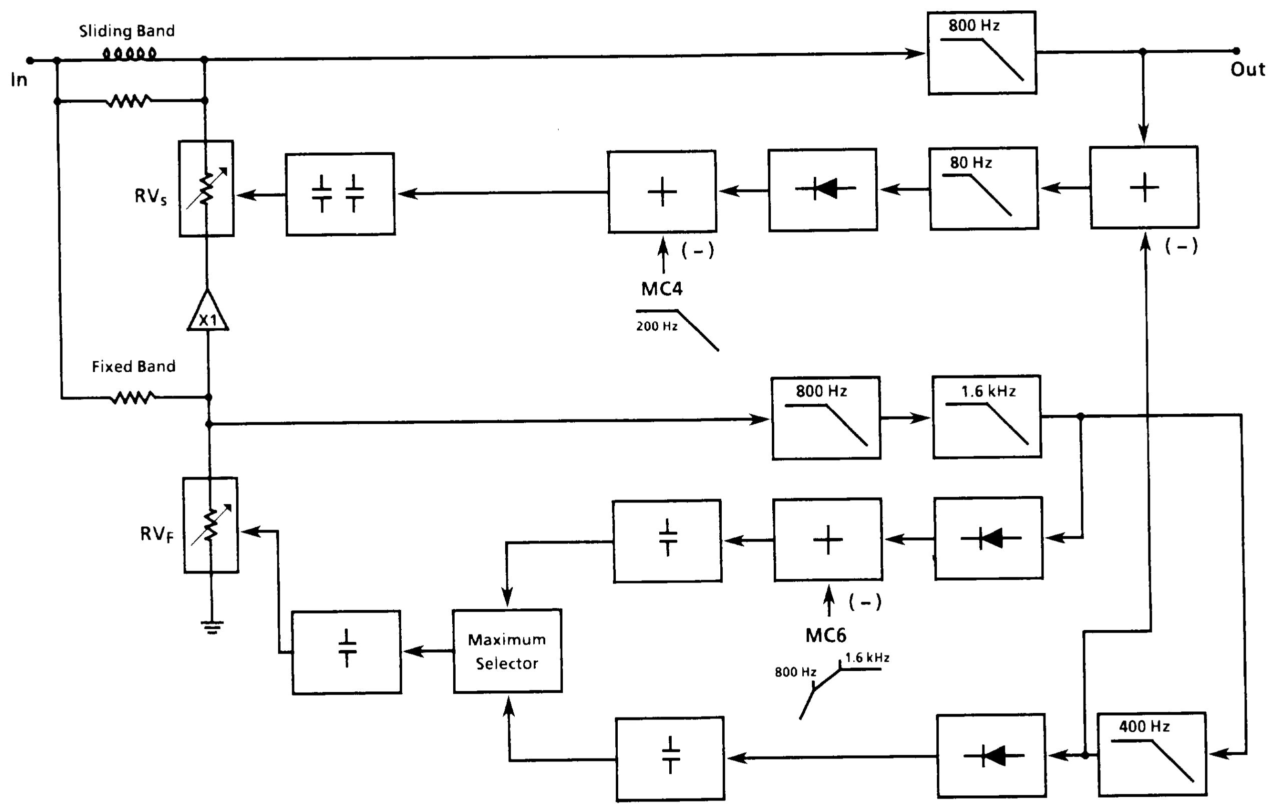

Fig. 10. Low-frequency stage block diagram—steady-state aspects.

MAIN

band signal circuits to handle the required levels with low distortion and low noise.

Referring to the fixed-band section (lower half of Fig. 10), the incoming signal is applied directly to the variable gain circuit. Control circuit frequency weighting is provided by cascaded single-pole 800-Hz and 1.6-kHz low-pass filters. (The corresponding filters in the high-frequency stages are the 800-Hz and 400-Hz filters at the inputs of the circuits.) The main control circuit rectifies the filtered signal; the resulting dc signal is bucked by modulation control signal MC6, smoothed by a 15-ms integrator, and fed to one input of the maximum selector circuit. The maximum selector circuit has the same purpose and mode of operation as in the high-frequency circuits.

The 800-Hz and 1.6-kHz frequency-weighted output of the fixed-band circuit is also fed to the pass-band control circuit (bottom of Fig. 10). Here the control signal is further weighted by a 400-Hz single-pole low-pass filter, rectified, smoothed by a 15-ms integrator, and fed to the other input of the maximum selector. As in the high-frequency stages, the larger of the two signals is passed to the final integrator (300 ms) to become the fixed-band control signal applied to RV_f. In this way, both simple and complex signals are accommodated.

The sliding-band control signal, as in the high-frequency circuits, is derived from the stage output—that is, from a point following both the fixed 800-Hz band determining filter and the variable filter. The signal is frequency weighted by an 80-Hz single-pole low-pass filter, rectified, and bucked by modulation control signal MC4 (which also has a single-pole low-pass characteristic, with the same type of sliding-band end-stop effect as in the high-frequency circuits). The result is smoothed by a 7.5-ms integrator and finally smoothed by a 150-ms integrator to become the sliding-band control signal applied to RV_s. As in the high-frequency stages, a single control circuit suffices for the sliding band.

The same type of low-level control characteristic modification is made in the low-frequency circuits as in the high-frequency circuits. Namely, a signal from the fixed band is combined in opposition with the sliding-band output signal (see combining circuit at right of Fig. 10). This differential control modification raises the sliding-band threshold at low frequencies.

3.5 Low-Frequency Stage: Transient Control Aspects

The transient control (and steady-state) aspects of the low-frequency stages are shown in Fig. 11. In a manner generally similar to that of the high-frequency circuits, unsmoothed rectified signals from the outputs of the variable elements are opposed by appropriate modulation control signals and are fed via diodes to the final integrator circuits.

Both the fixed and the sliding bands each have primary and secondary overshoot suppressors, which operate at frequencies above about 100 Hz. In addition, each has a gentle and slow-acting low-frequency overshoot suppressor, operating at frequencies below about 200 Hz; there is a crossover effect between the two types of overshoot suppression in the 100–200-Hz region. The primary overshoot suppressors provide the earliest and strongest suppression effect in simple transient situations. With more complex signals the primary overshoot suppression thresholds rise, and eventually the secondary overshoot suppression circuitry takes control.

In contrast with the high-frequency situation, the low-frequency general strategy is to derive the primary overshoot suppression signals from signal points that do not include any control circuit frequency weighting. This is because the required control circuit weighting networks of the low-frequency stages are low-pass in character, resulting in delays. (The high-pass networks used for control circuit weighting in the high-frequency stages do not introduce delays.) However, because of the lack of a weighting factor in the primary overshoot suppression signal, there is no inherent tracking between the steady-state and overshoot suppression thresholds of the circuits involved, particularly in the stop bands. Therefore further modulation control techniques are employed to obtain the required tracking. The secondary overshoot suppression signals are derived from a point in the fixed-band circuitry that provides adequate tracking in both the fixed and the sliding bands.

Referring to Fig. 11, the fixed-band primary overshoot suppression signal is generated by passing the variable attenuator output through a 200-Hz single-pole high-pass filter (middle of figure). This filter reduces the influence of the primary overshoot suppressor at low frequencies, allowing the more gentle low-frequency overshoot suppressor to take over the transient control function. The signal is rectified and then opposed by modulation control signal MC7, a 2-ms smoothed version of MC6 (the fixed-band steady-state modulation control signal); the effect is in the direction of improving the steady-state and overshoot suppression threshold tracking on a steady-state basis. However, the thresholds must also track on a transient basis. This is the function of the high-frequency transient modulation control signal MC8, which is a high-frequency-weighted, peak-detected signal that opposes the primary overshoot suppression signal in the time interval before MC7 becomes effective. The overshoot suppression signal is then diode coupled to the final integrator circuit of the fixed-band circuit.

In the generation of the sliding-band primary overshoot suppression signal the output of the variable filter is fed through a 200-Hz single-pole filter (top of Fig. 11) in order to reduce the effect of the circuit at low frequencies (as in the fixed-band circuit). The signal is rectified and then opposed by modulation control signals MC5 and MC7 to provide an adequate degree of tracking between the steady-state threshold and the overshoot suppression threshold on a steady-state basis. As in the fixed-band circuit, MC8 provides the required degree of tracking on a transient basis. The resultant overshoot suppression signal is diode coupled to the

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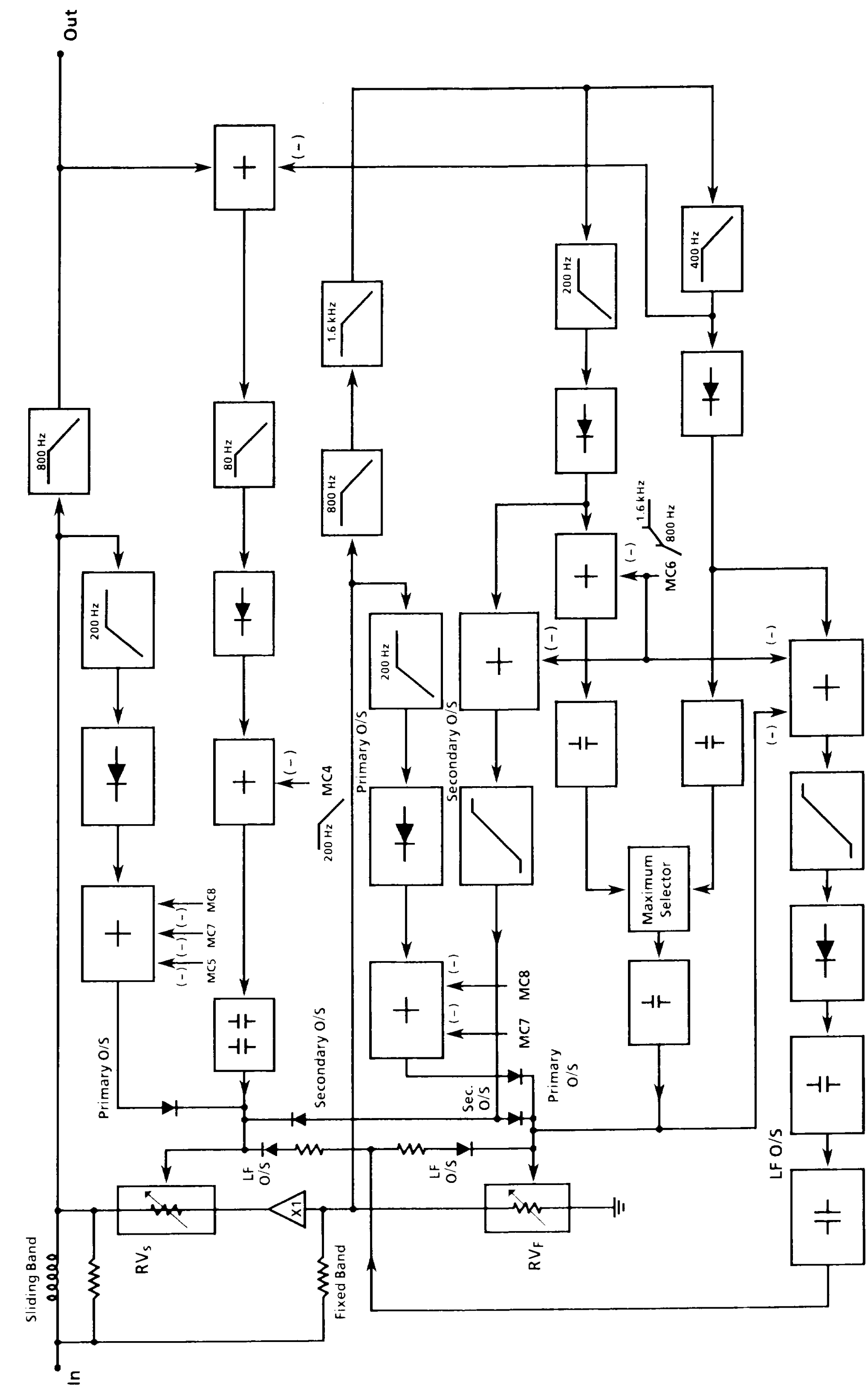

Fig. 11. Low-frequency stage block diagram—steady-state and transient control aspects.

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sliding-band final integrator circuit.

In both the fixed- and the sliding-band circuits, the effects of the primary overshoot suppression circuits are maximized for the most significant transient signal situation—that is, a single impulse or toneburst starting from a subthreshold signal level. A side effect of the use of smoothed MC5 and MC7 signals is that the overshoot suppression levels for low- and medium-frequency transient signals are raised under certain complex signal conditions, especially those in which relatively steady-state high-frequency signals at high levels are also present. To compensate for this effect, secondary overshoot suppression signals are derived from the fixed-band main control circuit rectifier and are diode coupled to the fixed-band and sliding-band final integrator circuits. The secondary overshoot suppressors have higher thresholds than the primary suppressors and operate only rarely because of the unusual circumstances for which they are designed.

The secondary overshoot suppression signals are generated from the frequency-weighted point (800-Hz and 1.6-kHz pass) in the fixed-band steady-state control circuit (lower middle of Fig. 11). To prevent interference with the low-frequency overshoot suppression circuit at low frequencies, the signal is further filtered by a 200-Hz single-pole high-pass network, as in the primary overshoot suppression circuits; the filtered signal is then rectified. (Note that in Fig. 10 this filter is not shown, for clarity. On a steady-state basis the pass-band control circuit controls the circuit at very low frequencies, via the maximum selector circuit; this arrangement allows the main control circuit rectifier to serve a double function.) The dc signal is opposed by MC6 in order to phase out the secondary overshoot suppression effect at high frequencies. An optimal phase relationship is obtained between the rectified signal and MC6, apart from the effect of the 200-Hz filter (which is negligible). An ideal tracking effect is achieved between the steady-state and secondary overshoot suppression thresholds.

The effect of the 800-Hz and 1.6-kHz frequency-weighting networks is to introduce a time delay into the secondary overshoot suppression signal. The effective delay is significantly reduced by increasing the gain used in the secondary overshoot suppressor circuit and applying limiting. The resultant overshoot suppression signal is more in the nature of a nearly fixed amplitude impulse, applied in the rare circumstances when necessary, than it is a proportional response. The signal is coupled through a diode to the fixed-band final integrator circuit and is also used, suitably biased, for secondary overshoot suppression in the sliding-band circuit, also coupled through a diode.

The low-frequency overshoot suppression signal (bottom of Fig. 11) is developed by tapping the rectified, but unsmoothed, output of the pass-band control circuit of the fixed-band circuit. The signal is opposed by MC6 to desensitize the circuit to high-level, high-frequency components. The signal is further opposed by the resulting fixed-band smoothed control signal, in a negative feedback fashion. (When the fixed-band control signal has risen to a sufficient level, there is no further need for any low-frequency overshoot suppressor action.) The signal is then highly amplified and limited, peak rectified, and smoothed by an integrator with about a 20-ms decay time constant. The resulting high-amplitude pulses are fed through a differentiating network, with a time constant of the same order as the integration time constant, to provide low-frequency overshoot suppression impulses of defined strength for distribution to the fixed-band and sliding-band final integrators, via high-value resistors and series diodes. The result is a decaying "constant current" charging of the capacitors of the final integrator circuits. This is in contrast with the higher peak currents and correspondingly more abrupt control voltage changes produced by the relatively low-impedance primary and secondary overshoot suppressors. The use of the low-frequency overshoot suppression method results in low waveform distortion of relatively slowly changing low-frequency signal impulses applied to the system.

The control signal recovery characteristics are similar to those of the high-frequency circuits, although being about half as fast because of the longer time constants employed. As in the high-frequency stages, the recovery time is favorably affected by the use of action substitution and modulation control, but is further augmented by the use of speed-up diodes.

4 OPERATING CHARACTERISTICS

The practical results of the various methods and circuits which have been described are given in Figs. 12–16, which show some of the main measurable operating characteristics of the SR process.

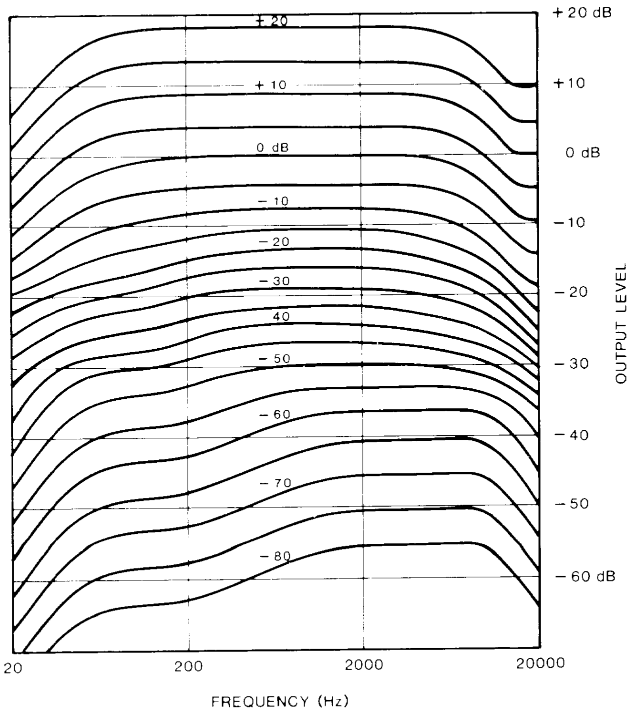

Fig. 12. Encoder characteristics—single tone.

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4.1 Dynamic (Compression) Action for Steady-State Dominant Signals

Referring to the single-tone encoder curves shown in Fig. 12, several features may be noted.

4.1.1 Low Frequencies

Dynamic action occurs in the range from about −48 dB to −5 dB (with respect to reference level). That is, there is no action (but full boosting) in the lower 35–40 dB of the dynamic range. Similarly there is no action in the top 25 dB of the total dynamic range. A linear dynamic characteristic prevails in these two regions (a bilinear characteristic).

4.1.2 High Frequencies

Dynamic action occurs in the range from about −62 dB to −5 dB. That is, there is no action in the lower 20–25 dB (but full boosting), or the top 25 dB of the dynamic range (a linear dynamic characteristic in these regions).

In the intermediate level regions of dynamic action the effects of the multilevel stages are joined together to create a compression ratio of about 2:1.

Referring to the high-level low- and high-frequency portions of the curves, the effective antisaturation of the system can be seen, with the combined effects of the spectral skewing and antisaturation networks, the SR stages, and about 1 dB of wide-band level compensation built into the coefficients of the stage signal combiner (Fig. 1). The overall result is an antisaturation effect of about 2 dB at 5 kHz, 6 dB at 10 kHz, and 10 dB at 25 Hz and 15 kHz. At high frequencies this amount of antisaturation significantly reduces distortion, reduces signal compression effects, and, with tape recording, improves the long-term stability of the recording. The high-frequency improvements are especially significant with 35-μs CCIR recordings. The antisaturation effect at low frequencies usefully counteracts tape overload, particularly with 3180-μs NAB recordings.

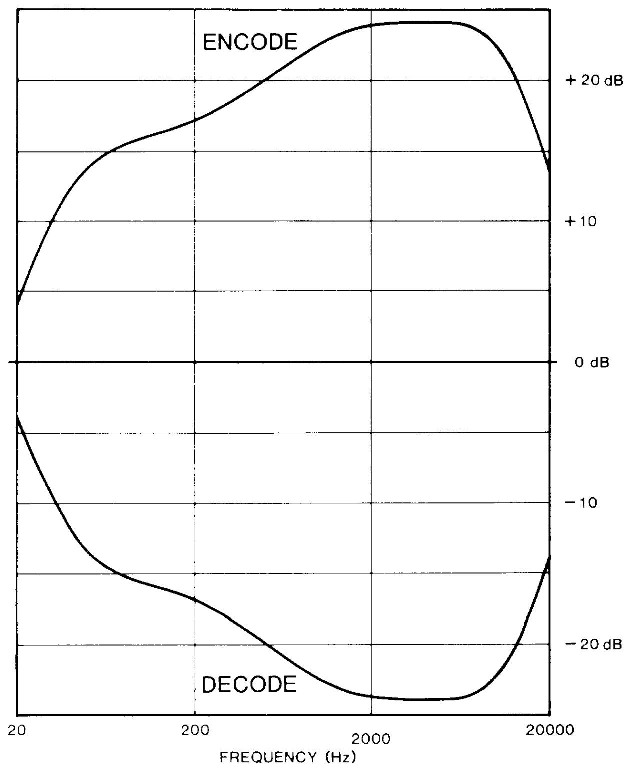

Fig. 13. Low-level (subthreshold) encoding and decoding characteristics.

4.2 Quiescent (Subthreshold) Signal Characteristic

The very low level, or subthreshold, characteristics of the SR process are shown in Fig. 13. The general shape of this characteristic was determined in a way that takes good advantage of the properties of human hearing. First, there is less of a noise generation and perception problem at moderately low frequencies (such as 200 Hz) than at moderately high frequencies (such as 3 kHz). Therefore two low-frequency stages are employed, but three high-frequency stages are used.

Second, at very low and very high frequencies even less noise reduction is needed (below 50 Hz and above 10 kHz). Strong spectral skewing actions can therefore be used in these regions, resulting in accurate decoding even when the recording medium has response irregularities. In addition the spectral skewing networks provide for good immunity to high- and low-frequency interference (supersonic audio components, tape recorder bias; subsonic noise components arising from wind, traffic, or other rumble sources).

Note that the overall shape of the low-level SR decode characteristic resembles the low-level Fletcher–Munson and Robinson–Dadson curves; the encode characteristic resembles the subsequently derived CCIR noise-weighting curve.

Thus the SR system is designed to reduce only those noises that can be heard. The prevention of action in inaudible signal regions promotes accuracy in the audible region.

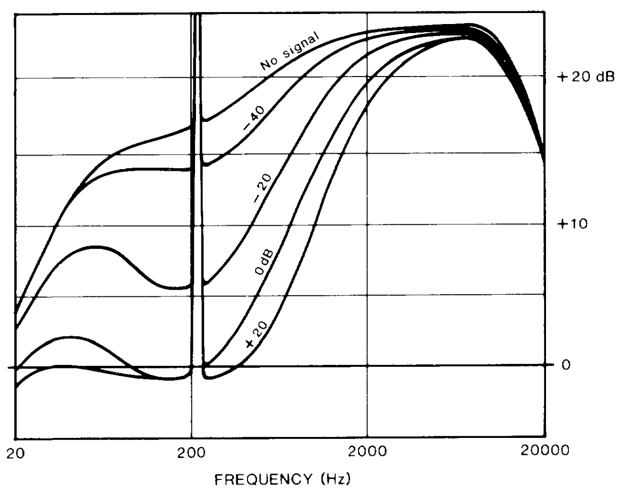

Fig. 14. Low-level encoding characteristic in the presence of a 200-Hz signal at the levels indicated.

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4.3 Treatment of Nondominant Signal Components

The behavior of the system with low-level nondominant signal components in the presence of higher level dominant signal components can be simulated by the use of probe tones. Such a representation is significant because it is an indicator of the noise reduction effect achieved with signals. Refer to the curves shown in Figs. 14–16, which were obtained by adding a swept frequency probe tone at levels between −60 dB and −80 dB into the encoder input signal and detecting the tone at the output with a tracking wave analyzer.

Toward the two spectrum ends, nondominant signal components are boosted more than the dominant signal by high- and low-frequency sliding-band actions. If there are two dominant signals, a fixed-band compression effect prevails for the nondominant signal components between the frequencies of the dominant signal components.

Thus nondominant signal components are boosted by an amount at least equal to the amount of boost of the dominant signal. The boosting of the nondominant signals is maintained toward the spectrum ends, even though the level of the dominant signal is relatively high (in the range −30 dB to +20 dB). This boosting action spectrally tracks the dominant signal frequency or frequencies.

It is advantageous to have a steeply rising boosting effect away from the frequency of the dominant signal component. In this connection the SR circuit profits from the steepness enhancing effect of cascaded stages. The low frequencies have two stages of steepness compounding; the use of three stages at high frequencies further improves the effect. These characteristics are particularly evident in the high-level areas of the probe tone curves.

The curves show that the encoder circuit tends toward keeping all low-level signal components boosted at all times. Only those components above the threshold are subject to a reduction of boosting. With regard to the overall encode–decode system, the advantages of this type of characteristic are 1) a powerful noise reduction effect in the presence of signals, and 2) a relative tolerance of level and frequency response errors in the channel between encoder and decoder.

4.4 Audible Results

As with the investigation of other psychoacoustic devices such as loudspeakers, measurements of the SR process can provide only a partial characterization. The rest must be obtained by detailed listening, using a wide range of source material and a variety of recording and transmission media.

Generally, the audible encoding effect of the system is to create a dense, bright sounding signal (as sent to the recording channel), but with little or no apparent dynamic action. Harmonics, overtones, and small-scale components of the sound, including noise, are all enhanced. At high signal levels the antisaturation characteristics cause a high- and low-frequency audible dulling of the encoded signal; when applied to the recording channel, this treatment results in a significant reduction of recording distortion.

The overall audible encoding and decoding effect of the SR process is simply to create a clean and accurate sounding replica of the input signal. Tape bias noise and modulation noise are significantly reduced. Also, a reduction of intermodulation distortion is achieved by the low-frequency noise reduction capabilities of the process, as well as by the effects of the antisaturation characteristics used during encoding.

Furthermore, the decoding portion of the system reduces harmonic distortion generated by the recording channel. Steady-state third-harmonic distortion is typically reduced to less than one-half, fifth-harmonic distortion to less than one-quarter; higher order harmonics are even further reduced. Thus, especially if the recording medium has a hard clipping characteristic, the subjective cleanliness of the signal at high recorded levels is significantly improved.

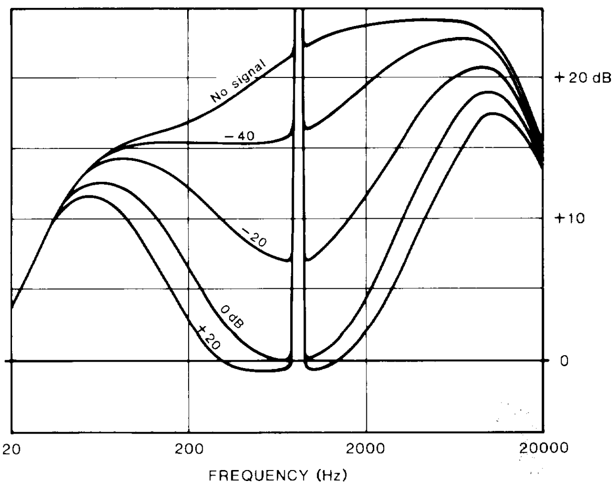

Fig. 15. Low-level encoding characteristic in the presence of an 800-Hz signal at the levels indicated.

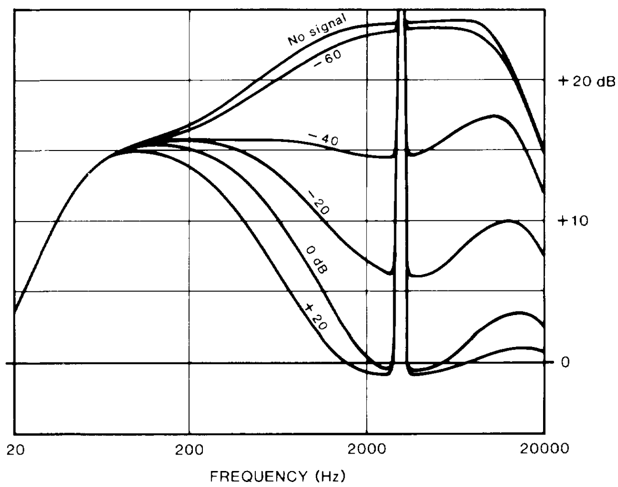

Fig. 16. Low-level encoding characteristic in the presence of a 3-kHz signal at the levels indicated.

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5 CALIBRATION

The spectral recording calibration procedures are conceptually similar to those of the A-type, B-type, and C-type systems. That is, signal levels in the decoder circuit ideally should match those in the encoder circuit, even though the SR process has been designed to be more tolerant of gain and frequency response errors than these previous systems. For tape interchange standardization it is also preferable if, at least within a given organization, the reference level of the encoder and decoder corresponds to a known and fixed flux level. Whether or not a standardized flux is used for this, the matching of the decoder to the encoder is accomplished by a calibration signal generated in the encoder and recorded on the tape; this allows the tape replay gain to be set correctly, using the meter in the decoder unit.

Most problems in the studio use of noise reduction, and indeed analog recording in general, can be traced to incorrect level settings and/or frequency response errors in the recorder. This may be because checking these factors is a time-consuming and boring process. A faster and more interesting method of accomplishing these checks would be more likely to produce reliable and consistent results. For this reason, practical embodiments of the SR process include pink-noise generators which are used for both level and frequency response calibration, instead of single-tone sine-wave oscillators. For identification, the pink noise is interrupted with 20-ms "nicks" every 2 s. During recording this signal is fed to the tape at a level of 15 dB below reference level, a level low enough not to cause saturation problems with low-speed tape recording or highly equalized transmission channels.

During playback the tape signal is automatically alternated with internally generated reference pink noise (uninterrupted) in 4-s segments (8-s total cycle time) and passed to the monitor output. An audible comparison can thus be made between the reference pink noise and the calibration noise coming from the tape. Any discrepancies in level and/or spectral balance are immediately noticeable and can be corrected or at least taken note of. If desired, the signal can also be fed to a spectrum analyzer.

In using the new calibration method it is important to be able to tell when the 4-s tape segments are being passed to the monitor and when the signal heard is from the reference pink-noise generator. Differentiation of the tape segments from the reference segments is accomplished in two ways. First, the reference segments are 4 s of continuous pink noise, and the tape segments begin with a nick, have a nick in the middle, and end with a nick; this time sequence is easily identified with a little practice. Second, colored lights identify the two different signals.

The new calibration facility gives the recording and production personnel a useful control of the recording process. At any time a check of the recorder can be made; the result can be heard immediately and conclusions drawn about whether adjustments might be necessary.

With tape and signal interchanges it is possible to tell quickly whether there is any error or misunderstanding about levels, equalization, azimuth, and the like. If the original recording of calibration noise stays with the tape, the quality of the ultimate playback, even after copying, can be retained. Thus the comparison function serves to ensure that the recorder and spectral recording process provide on a routine basis the signal quality and reliability of which they are capable.

6 CONCLUSION

A new professional recording format, designated spectral recording, has been described. The objective of the new encoding and decoding process is to record and reproduce audio signals with a high degree of audible signal purity.

The system employs a dual-path multilevel, staggered action arrangement of two low-frequency compressor stages and three high-frequency compressor stages, each with a fixed band and a sliding band. The outputs of the bands are combined in a unique way, called action substitution, which results in an unusually responsive treatment of the signal with respect to both frequency and level; a technique referred to as modulation control augments the spectral tracking abilities of the system. Spectral skewing contributes to a tolerance of channel errors, and the employment of both high- and low-frequency antisaturation techniques results in a significantly improved channel overload characteristic.

7 ACKNOWLEDGMENT

The author is indebted to Douglas Greenfield for the printed-circuit layouts, mechanical design, and general assistance with the first practical embodiments of the new system. Thanks are also due to many other individuals, particularly to Henry Bottino for the design of testers, to Brad Teague for preproduction testing, and to Michael Smith, Christopher James, and Martin Fried for production engineering.

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THE AUTHOR

Ray Dolby was born in Portland, Oregon, in 1933, and received a B.S. degree in electrical engineering from Stanford University in 1957. From 1949–52, he worked on various audio and instrumentation projects at Ampex Corporation, and from 1952–57 he was mainly responsible for the development of the electronic aspects of the Ampex video tape recording system. After he was awarded a Marshall Scholarship, followed by a National Science Foundation graduate fellowship, he left Ampex in 1957 for further study at Cambridge University in England where he received a Ph.D. degree in physics in 1961 and was elected a Fellow of Pembroke College (Honorary Fellow, 1983). During his last year at Cambridge, he was also a consultant to the United Kingdom Atomic Energy Authority.

In 1963, he took up a two-year appointment as a United Nations adviser in India and returned to England in 1965 to establish Dolby Laboratories in London. Since 1976 he has lived in San Francisco, where his company has established further offices, laboratories, and manufacturing facilities.

Dr. Dolby holds a number of patents and has written papers on video tape recording, long wavelength X-ray analysis, and noise reduction. He is a fellow and past-president of the AES and a recipient of its Silver Medal Award. He is also a fellow of the British Kinematograph, Sound and Television Society, and of the SMPTE, and a recipient of its Samuel L. Warner Memorial Award, the Alexander M. Poniatoff Gold Medal, and the Progress Medal. In 1979 he and his colleagues received the Scientific and Engineering Award of the Academy of Motion Picture Arts and Sciences. In 1986 he was awarded the British title of Officer of the Most Excellent Order of the British Empire (OBE).

MAIN

An Audio Noise Reduction System*

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A noise reduction system which is suitable for use with high-quality audio recording or transmission channels is described. A special signal component, derived from four band-splitting filters and low-level compressors, is combined with the incoming signal during recording or sending. During reproduction, the additional component is removed in a complementary way; any noises acquired in the channel are attenuated in the process. Practical features of the system include: 10 dB (unweighted) noise reduction; imperceptibility of signal-modulated noise effects; level frequency response (overall); accuracy of reproduced signal dynamics; low distortion; low internal noise level; and stability of characteristics.

INTRODUCTION In an audio recording or transmission channel, noises of varying degrees of avoidability arise in the channel itself. Apart from correction at the source, any scheme to reduce the audibility of such noises, which may include hum, crosstalk, print-through, hiss, and other undesired signals, can be classified broadly into one of two types (Fig. 1):

a) non-complementary, in which the signal is post-processed only, thereby producing an overall alteration of the signal while reducing noise, and

b) complementary, in which both pre- and post-processing are used, the attempt being to produce no overall alteration while reducing noise.

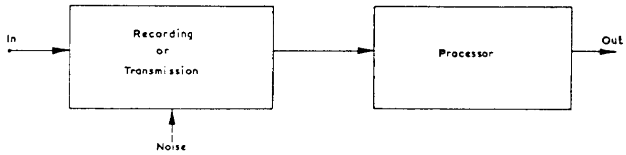
Examples of non-complementary noise reduction systems include simple tone controls and filters. In more sophisticated forms, the filtering action may be made dynamic, as in the Dynaural system of Scott[1] or the multiple-band diode expander system of Olson.[2] Automatic signal-controlled attenuators, which function similarly on a wideband basis, have also been described.[3,4]

The noise problem has also been attacked in complementary ways, the simplest method being the use of optimized equalization characteristics.[5] Various types of compressors and expanders have been devised, both of the instantaneous variety, employing nonlinear networks, and of the syllabic type, in which linear variable-gain devices are controlled in accordance with the signal envelope[6-9]; in some systems pilot tones are used in the expansion process.[10,11] An electronically switched two-channel (low level-high level) noise reduction system has also been developed recently by Mullin.[12]

* Presented April 25, 1967, at the 32nd Convention of the Audio Engineering Society, Los Angeles.

REQUIREMENTS

Referring to Fig. 1b, it is possible to draw up a set of requirements which any noise reduction system must meet if it is to be used without reservation in high-quality recording or transmission channels. The system will necessarily be of the complementary type.



a. Non-complementary system

b. Complementary system

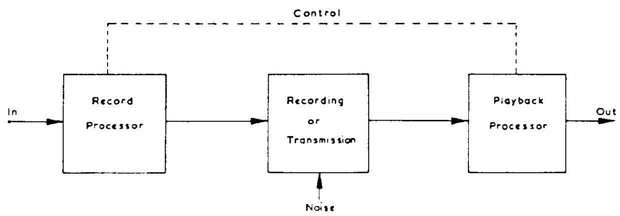


Fig. 1. Noise reduction systems, illustrating two basic types with reference to effects on signal. The control path shown in **b** is optional.

Overall Signal Quality Requirements

1. The output signal should not be perceptibly different from the input in frequency response, transient response, and dynamics; stereo signals thus should be perceptibly free of image wandering or shifting.

Reprinted from Journal of the Audio Engineering Society, October 1967

2. The system should not introduce perceptible non-linear distortion of transient or steady-state signals at any level or at any frequency or combination of frequencies; the overload point should be substantially above the normal peak signal level.

3. The system should have a low internal noise level and should not generate any additional perceptible noises in the presence of signals.

4. All the above requirements should be met in tandem operation of the system (i.e., with multiple processing and de-processing of the signal).

Requirements Relating to Recording or Transmission Channel

1. The output from the recording/sending processor should be suitable for transmission through one channel of normal audio bandwidth.

2. Correct operation should not be dependent upon linear phase-frequency response in the channel.

3. Normally encountered errors or fluctuations in gain and frequency response of the channel should not cause audibly significant changes in the system output.

4. The system should not modify significantly the overall steady-state or transient overload characteristics of the channel.

Interchangeability Requirements

1. The operating characteristics of the system should be fixed and reproducible.

2. The processing units should be sufficiently stable with time, temperature, and other factors to permit interchange of recordings or channels.

Noise Reduction Requirements

1. The amount of noise reduction should be perceptibly similar for all types of noises encountered.

2. The noise reduction action should be perceptibly free of signal-modulated noise effects with any normally encountered combination of program material and noise.

COMPANDORS

Of the possible noise reduction methods which have been investigated, the syllabic compressor and expander (compandor) technique (Fig. 2) has been the subject of the most development effort. Since the noise reduction system to be described in this paper may be roughly classified as a compandor, it is worth noting some of the limitations of previous approaches to compression and expansion.

Well-known compandor difficulties—which by now are regarded as classical—include poor tracking between recording/sending and reproducing/receiving, both statically and dynamically; high sensitivity to gain errors in recording or transmission; inadequate dynamic range (high noise level *vs* high distortion); overshooting with transient inputs; audible modulation-product generation under dynamic conditions; distortion of low frequencies by control-signal ripple modulation; and production of noticeable signal-modulated noise effects.

Comparison of compandor performance with the previously listed requirements for high-quality applications shows that the normal compression and expansion approach is inadequate. Compandors have thus been found to be usable without qualification only in relatively low-grade, narrow-band applications such as telephone circuits.

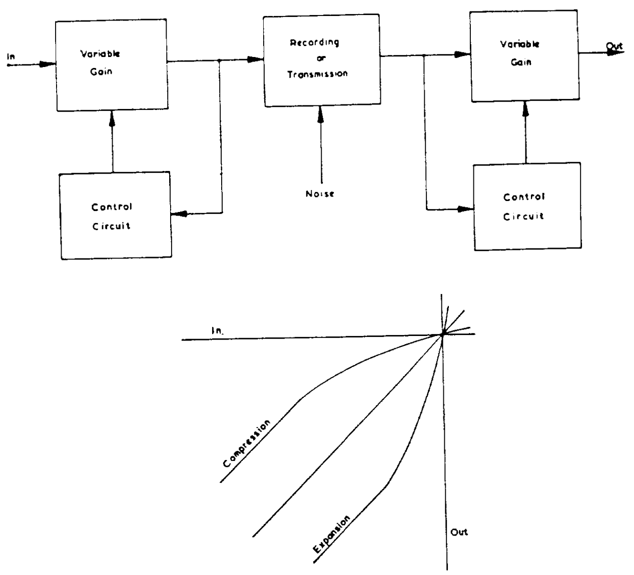


Fig. 2. Layout and input-output transfer characteristics of a compandor noise reduction system.

NOISE REDUCTION SYSTEM: DIFFERENTIAL METHOD

A noise reduction system which is capable of meeting the listed requirements has been developed and is described below.

In normal compression or limiting, a primary object is to modify high-level signal dynamics; it is thus unfortunately necessary to subject the signal as a whole to the hazards of passage through a variable-gain system. In applying compression techniques to the noise reduction problem, in which the objective does not include modification of signal dynamics, it is unnecessary and undesirable to operate upon high-level signal components; noise amplitude in a high-quality channel is only of the order of 0.1% of maximum signal amplitude. It would clearly be preferable to generate a small correction or differential component which could be appropriately subtracted from the signal, thereby cancelling or reducing noise while leaving the larger aspects of the signal untouched.

The differential treatment of the signal in the present noise reduction system is illustrated in Fig. 3. The net-

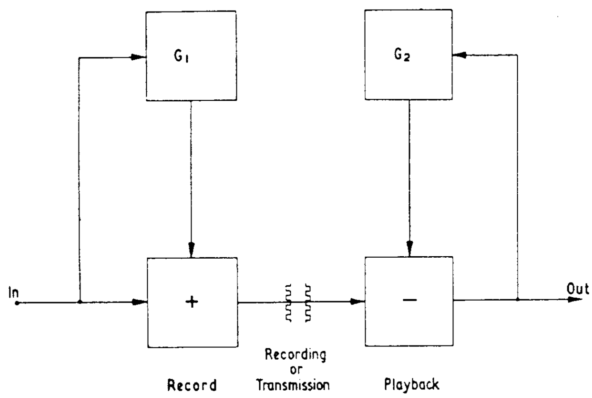


Fig. 3. Basic layout of noise reduction system. In practice, the operators $\mathbf{G_1}$ and $\mathbf{G_2}$ comprise identical sets of four filters and low-level compressors.

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works (operators) G_1 and G_2 are signal multipliers controlled by the amplitudes, frequencies, and dynamic properties of the signals fed into them. During reproduction, network G_2 passes low-level components (noise) back to the subtractor, which partially cancels these components in the signal from the channel. In the process of reducing noise, G_2 and the subtractor also partially cancel low-level signal components. To compensate for this cancellation, the network G_1, which has the same characteristics as G_2, adds an identical component prior to recording/sending.

These operations may be expressed in the following way. If the input to the recording processor is x (some function of time), the signal in the channel is y, and the output signal from the reproducing processor is z, we have

$$y = [1+G_1(x)]x \qquad (1)$$

and

$$z = y - zG_2(z) \quad \text{or} \quad z = \{1/[1+G_2(z)]\}y. \qquad (2)$$

Combining Eqs. (1) and (2),

$$z = \{[1+G_1(x)]/[1+G_2(z)]\}x. \qquad (3)$$

The solution of interest is: $G_1 = G_2$; $z = x$. Thus, the output signal will be equal to the input signal if the recording and reproducing differential networks (i.e., the operators G_1 and G_2) are identical, on condition that $G(z)$ is not allowed to become -1 (no oscillation) and that the functions in Eqs. (1) and (2) are continuous and single-valued (no tracking ambiguity).

The prime requirement of any high-quality noise reduction system—that the signal should be unchanged overall—is thereby satisfied, and it is necessary only to choose an operator that yields a recording/sending signal which is compatible with the channel and that produces satisfactory noise reduction properties.

Steady-State Properties

Referring to the steady-state transfer characteristics shown in Fig. 4, the noise reduction requirement, together with the desirability of interfering as little as possible with high-level signal components, dictates a reproducing (expansion) curve of the type shown in Fig. 4b; that is, the gain at low levels must be reduced, while a unity gain condition should prevail at high levels. The required differential component transfer characteristic, shown in Fig. 4c, is then determined, being linear up to the compression threshold, rising slightly with increasing input, and finally decreasing with larger inputs. In practice, such a characteristic is formed by deriving the compressor control voltage from a combination of feedforward and feedback signals.

The recording (compression) transfer characteristic shown in Fig. 4a is complementary to the reproducing characteristic, amplifying low-level signal components in order to compensate for the corresponding deficiencies produced by the noise reduction action during reproduction.

Comparison of the differential method of forming the compression and expansion laws with the conventional approach depicted in Fig. 2 shows that the scheme has several advantages. Nonlinear and modulation distortion are both reduced since the compressor (limiter) contribution is negligible at high levels. System noise problems are alleviated; the variable gain device can be operated at higher levels than would be possible if it were called upon to pass the whole dynamic range.

Tracking accuracy problems between units are also reduced, since the transfer characteristic is largely determined by two readily controlled factors: the compression threshold and the addition or subtraction coefficient of the differential component. At low and high levels the possibility of mistracking is minimal, and in the transition region it is not a difficult design matter to hold the error to a small fraction of a decibel.

A further tracking characteristic concerns compatibility of the system with the audio channel; to a first order, gain variation in the channel manifests itself only as a level change at the output, not as an alteration of signal dynamics. For the parameters used in the present system the maximum tracking error, having a decibel value approximately equal to that of the dB error in gain, occurs at about 30 dB below peak operating level, where its effect is unobtrusive. The method is thus, in practice, tolerant of moderate errors in gain. This tolerance is especially significant in stereo, as it enables the noise reduction system to operate without control signal interconnections.

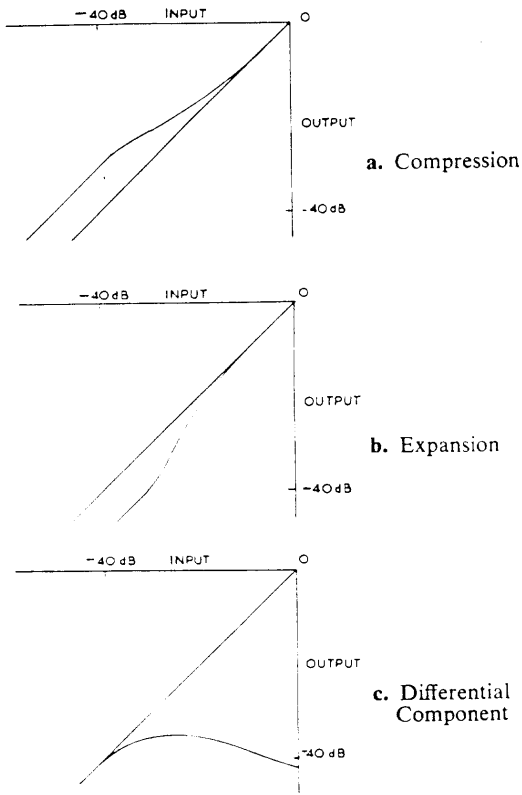

Fig. 4. Input-output transfer characteristics of the noise reduction system. The compression characteristic is formed by adding the differential component to the input signal; the expansion characteristic is formed by subtracting the differential component from recorded/transmitted signal in accordance with negative feedback configuration shown in Fig. 3.

A related matter is the tracking behavior of the system with channels having nonlinear phase-frequency response. For a given rms value, the peak and average values of a complex wave depend on the phase relationships of the various frequency components. With a channel of uncertain phase response it is in principle necessary to

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control the compression and expansion operations using the rms value of the signal, a procedure which at best is inconvenient. However, in practice, a combination of peak and average values is a sufficiently accurate indicator of the rms value to permit the use of relatively simple rectification and smoothing circuits; in the present system such circuits are used. Good tracking is thereby obtained even when the signal has suffered considerable phase distortion.

A further channel compatibility aspect concerns the possibility of overloading channels with frequency-dependent overload characteristics. The overload properties may be further complicated if pre-emphasis is used. Since the pre-emphasis is usually based on the energy probability distribution with frequency for normally encountered sounds, it is evident that any practical noise reduction system should not interfere unduly with this distribution. The compression of comparatively high-level signal components thus must be avoided; the transfer characteristic of the present system satisfies this condition (see Fig. 4a).

Dynamic Properties

Overshoots, arising because of control circuit time lag, normally have maximum amplitudes equal in value to the degree of compression. Such overshoots waste some of the dynamic range of the audio channel if they are passed linearly. Furthermore, if they are clipped by the channel, various undesirable side-effects can be created: for example, blocking of amplifiers, break-through from groove to groove with discs, and interference with other channels if modulated carriers are used. Controlled clipping of the output signal in the compressor itself is a method of avoiding these difficulties, but it has the disadvantage of reducing the overload margin.

The usual solution is to make the attack time as short as possible and either to clip within the device or to depend upon the shortness of the overshoot to minimize side effects with clipping in the channel. Unfortunately, the use of short attack times results in side effects in the signal. Rapid changes in gain cause significant modulation products to be generated, which may or may not be cancelled by reciprocal treatment following transmission.

With the method under discussion it is not only possible to confine overshoots to small values but to use relatively long attack times, thereby reducing modulation distortion. Referring to the differential network portion of Fig. 5, the method used is to follow the compressor circuit (linear limiter) with a conventional symmetrical clipper (nonlinear limiter). A suddenly applied signal is thus momentarily passed without attenuation to the clipper; the differential component is confined to an amplitude which results in negligible overshoot when added to the main signal. In the present system the clipping level has been chosen to limit the overshoot to 2 dB with peak amplitude step inputs.

The addition of the low-amplitude clipped signal to the large-amplitude pure main signal results in momentary distortion of a few percent, but the degradation is so small and of such short duration (1 msec or less, depending on the frequency) that it is masked by transient components present in the input signal, as well as subjectively attenuated by the relatively slow loudness-growth characteristics of the ear.[13] In practice, the clipper circuit is rarely called upon to perform its function, the compressor operating linearly except with the most percussive types of program material.

Regarding modulation distortion, it is evident that at high levels such effects are negligible because of the diminished influence of the differential component. By the use of nonlinear control signal smoothing circuits, distortion is minimized at low levels as well. A relatively long attack time (of the order of 0.1 second) is used for small variations in signal amplitude, the gain changes produced being slow enough that they do not generate audible modulation products. The time constant is decreased in accordance with the size of the amplitude transition, and for steps large enough to cause the compressor output to exceed the clipper threshold the attack time is reduced to such an extent that the modulation/clipping distortion produced is masked by the transient components present in the input signal.

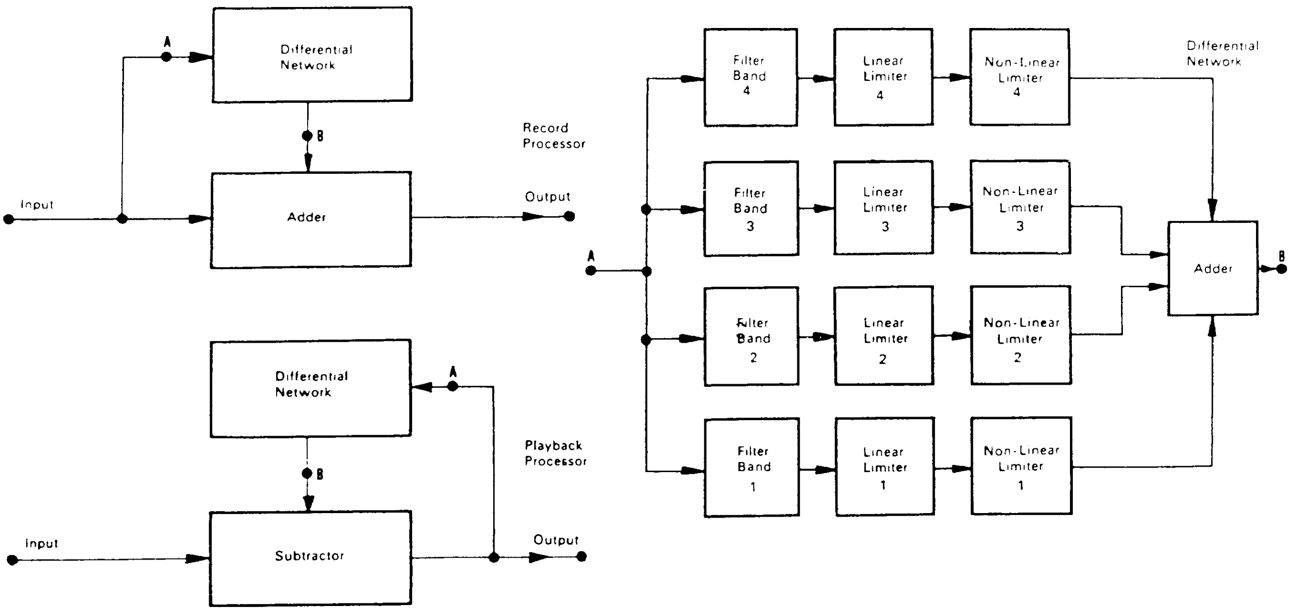

Fig. 5. Basic block diagram of the noise reduction system. Differential network, shown at right, is the same for both recording and playback; the filters and compressors work under identical conditions, both statically and dynamically, in the two modes.

The use of long attack times as described not only reduces modulation distortion but tends to improve the noise reduction of the system. Since the amount of noise reduction depends upon the amplitude of the differential component in relation to that of the signal in the main path, it is an advantage if short transients of moderate amplitude are prevented from causing unnecessary compression of the differential component. Overshoots of several dB may be produced under these conditions, but they are of such low amplitude compared with peak level that they are handled linearly in all respects.

While the attack behavior is undoubtedly the most important dynamic aspect of the system, particularly in relation to ensuring compatibility of the recording/sending processor output with the audio channel, the decay or recovery time is of equal significance when the noise reduction properties are considered. The problem in this regard is to reduce the recovery time to such a value that noise reduction following cessation of high-amplitude signals is provided adequately by the residual masking phenomenon,[14] by which the sensitivity of the ear is momentarily reduced. The noise reduction action of the system must thus be restored in an interval of the order of 0.1 sec, during which residual masking prevails.

The use of short recovery times in normal compressors results in high distortion at low frequencies. Furthermore, modulation distortion, which was discussed previously from the point of view of attack time, is a product of short recovery time as well as short attack time.

As with the attack aspects of the matter, the recovery problem in the present system is solved jointly by the differential method itself and by suitable choice of characteristics of the nonlinear control signal integration circuitry; the smoothing time constant is made long under equilibrium conditions but is decreased appropriately for large abrupt reductions in signal amplitude. In this way, low-frequency distortion in the recording/sending processor output is readily held to negligible values at high and low levels and to moderate values (a fraction of a percent) at intermediate levels, while the recovery time is made sufficiently fast that perceptible noise modulation effects are avoided following cessation of the signal.

Because of the undistorted character of the recording/sending processor output signal, the system does not depend upon subsequent distortion cancellation during reproduction for correct operation. Phase errors in the audio channel thus are not troublesome; the signal may be re-recorded a number of times or be sent through transmission lines, both being important applications in which nonlinear phase-frequency characteristics prevail. Also, the signal may be processed and de-processed repeatedly with negligible cumulative distortion effects.

It may be remarked that some of the operating characteristics discussed, which have generally been attributed to the differential method, are in fact properties of the overall compression law produced (see Fig. 4a). Good tracking, high tolerance of gain-errors in the audio channel, avoidance of steady-state overloading of highly equalized channels, and negligible formation of modulation products at high amplitudes are features of the overall transfer characteristic, not of the method of forming it. However, the weaknesses of a direct approach to such a transfer characteristic would appear in the usual forms: overshoots, high noise level, high nonlinear distortion, difficult reproducibility, and poor stability.

NOISE REDUCTION SYSTEM: BAND SPLITTING

The advantages of the differential method of deriving the compression law are dependent upon the existence of a large ratio between the maximum amplitude of the signal in the main path and the maximum amplitude of the differential component; it follows that the compression threshold must be set at a low value. Unfortunately, a low compression threshold is detrimental to good noise reduction properties. With moderate and high-level signals the noise reduction action technically disappears, so that if only one full frequency compression band were used, an unacceptably high degree of program-modulated noise would be evident. This difficulty has been overcome in the present system by splitting the differential component into four frequency bands (Fig. 5). Dependence is placed on the masking effect for subjective noise reduction in portions of the spectrum occupied by signals having amplitudes appreciably higher than the compression thresholds.

Beginning with the early studies of Wegel and Lane,[15] investigations of the masking effect have been concerned almost exclusively with the masking of pure tones by tones or noise.[16] The considerable body of results available is unfortunately not very relevant in the noise reduction application, in which the masking of a band of noise by one or more tones is of interest. A closer approach to the conditions required is the masking of one band of noise by another band of noise.[17] But systematic research into the use of wideband noise as the maskee has only recently been undertaken,[18] and it would appear that it will be some time before sufficient work has been done to permit the choice of noise reduction system design parameters simply by reference to published psychoacoustic data.

In applying the masking phenomenon to the noise reduction system design problem, the number of bands used (circuit complexity) must be balanced against other parameters and the overall system performance requirements. It was found that for normal high-quality audio channels the use of four bands yields satisfactory noise reduction properties while permitting the compression thresholds to be set at a value low enough to obtain the advantages of the differential method.

In the system under discussion, using four bands with compression thresholds of 40 dB below peak operating level, the frequency divisions are: Band 1, 80 Hz low-pass; Band 2, 80 Hz to 3 kHz band-pass; Band 3, 3 kHz high-pass; Band 4, 9 kHz high-pass. Bands 1, 3, and 4 are conventional 12 dB per octave filters, while Band 2 has a frequency response which is complementary to that of Bands 1 and 3. The outputs of all the bands are combined with the main signal in such a way as to produce a low-level output from the recording/sending processor which is uniformly 10 dB higher than the input signal up to about 5 kHz, above which the increase in level rises smoothly to 15 dB at 15 kHz.

The figure of 10 dB represents a compromise between a number of design factors, not the least of which is the desirability of minimizing sensitivity to gain-errors

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in the audio channel. At the high end of the spectrum an extra 5 dB of noise reduction is obtained without appreciable disadvantage in this regard; Bands 3 and 4 contribute approximately equally in this region, so that with normally encountered sounds the output of Band 3 is usually compressed substantially before the threshold is reached in Band 4. The maximum compression ratio is thereby reduced, and the possibility of occurrence of program-modulated frequency response under gain-error conditions is decreased.

Because of the use of four bands, with consequent interactions between these bands, the noise reduction properties of the system under signal conditions are not altogether simple. These properties are, however, amenable to investigation and measurement by the use of low-level probe tones.

The overall noise reduction action of the system may be summarized as follows: Band 1 provides noise reduction in the hum and rumble frequency range; Band 2, in the mid-audio range (broadband noise, crosstalk, print-through); Bands 3 and 4, in the hiss range. With average orchestral music, Band 1 is compressed fairly often; Band 2, almost all the time; Band 3, fairly often; and Band 4, rarely. The noise reduction action thus arises most of the time from low- and high-frequency pre-emphasis and complementary de-emphasis. The high-frequency de-emphasis not only attenuates hiss but in magnetic tape recording it reduces high-frequency modulation noise. High-frequency sidebands of lower-frequency signals suffering frequency modulation due to scrape flutter are treated similarly.

SUMMARY

The general principles of a noise reduction system suitable for high quality use have been described. Low-level signal components are amplified in four independent frequency bands prior to recording/sending, which is accomplished by adding the outputs of four filter and low-level compressor channels to the main signal. During reproduction, the filter and compressor network is connected in a complementary way. Low-level components are subtracted from the incoming signal, and noise acquired in the audio channel is thereby subtracted or reduced as well.

The noise reduction system described is capable of performance of a high order with regard not only to noise reduction, but also to signal quality, compatibility of the processed signal with the audio channel, and suitability of characteristics under non-ideal channel conditions.

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