Instruction Manual

DP5500 SERIES DSTL
Digital Studio-to-Transmitter Link

Models DP5501A,DP5502,DP5503A,DP5504

MARTI

IMPORTANT INFORMATION

EQUIPMENT LOST OR DAMAGED IN TRANSIT.

When delivering the equipment to you, the truck driver or carrier's agent will present a receipt for your signature. Do not sign it until you have: 1) inspected the containers for visible signs of damage and 2) counted the containers and compared with the amount shown on the shipping papers. If a shortage or evidence of damage is noted, insist that notation to that effect be made on the shipping papers before you sign them.

Further, after receiving the equipment, unpack it and inspect thoroughly for concealed damage. If concealed damage is discovered, immediately notify the carrier, confirming the notification in writing, and secure an inspection report. This item should be unpacked and inspected for damage WITHIN 15 DAYS after receipt. Claims for loss or damage will not be honored without proper notification of inspection by the carrier.

RF PRODUCT TECHNICAL ASSISTANCE - REPAIR SERVICE - REPLACEMENT PARTS.

Technical assistance is available from Marti Electronics by letter, prepaid telephone, fax, or E-mail. Equipment requiring repair or overhaul should be sent by common carrier, prepaid, insured, and well protected. Do not the mail equipment. We can assume no liability for inbound damage, and necessary repairs become the obligation of the shipper. Prior arrangement is necessary. Contact Marti Electronics for a Return Authorization.

Emergency and warranty replacement parts may be ordered from the following address. Be sure to include the equipment model number, serial number, part description, and part number.

Marti Electronics
421 Marti Drive P.O. Box 661
Cleburne, Texas 76033
Telephone: (817) 645-9163 (8 AM to 5 PM Central Time)
Fax: (817) 641-3869
E-Mail: General - marti@flash.net
Web Site: www.bdcast.com

RETURN, REPAIR, AND EXCHANGES.

Do not return any merchandise without our written approval and Return Authorization. We will provide special shipping instructions and a code number that will assure proper handling and prompt issuance of credit. Please furnish complete details as to circumstances and reasons when requesting return of merchandise. All returned merchandise must be sent freight prepaid and properly insured by the customer.

WARRANTY ADJUSTMENT.

Marti Electronics, Inc. warranty is included in the Terms and Conditions of Sale. In the event of a warranty claim, replacement or repair parts will be supplied F.O.B. factory. At the discretion of Marti Electronics, the customer may be required to return the defective part or equipment to Marti Electronics, Inc. F.O.B. factory. Warranty replacements of defective merchandise will be billed to your account. This billing will be cleared by a credit issued upon return of the defective item.

PROPRIETARY NOTICE.

MODIFICATIONS.

Marti Electronics, Inc. reserves the right to modify the design and specifications of the equipment in this manual without notice. Any modifications shall not adversely affect performance of the equipment so modified.

Users' Manual

For

DP5500 Series DSTL®
Digital Studio-to-Transmitter Link

Models DP5501A, DP5502, DP5503A, DP5504

Marti Electronics
421 Marti Drive, Box 661, Cleburne, Texas 76033
Tel: 817-645-9163; Fax: 817-641-3869, E-Mail: marti@flash.net

Thank you! We appreciate your purchase of the world's first fully integrated digital aural studio transmitter link system. Broadcast Electronics Model DP5501A/5503A DSTL Transmitter and Model DP5502/5504 DSTL Receiver combine the benefits of Dolby AC-2® digital audio source coding, spectrum-efficient digital modulation and performance-optimized transmitter and receiver subsystems for the first time in the history of aural STL applications. The revolutionary DSTL technology is a departure from conventional analog FM transmission techniques and their inherent performance limitations in audio quality and interference rejection in dense RF environments.

The key benefits of Broadcast Electronics DSTL technology are:

DP5501A/5502

- Simultaneous transmission capacity for two 15 kHz bandwidth main channels, one 7.0 kHz bandwidth auxiliary channel, and one 3.5 kHz bandwidth voice channel.
- The auxiliary and voice channels will accommodate 9600 and 1200 bit/s modems respectively for remote control and data transmission applications.
- Linear RF power amplifier technology and narrow 250 kHz transmission bandwidth provide for improved spectrum utilization and adjacent channel operation at 250 kHz spacing between two DSTL systems. This allows a two-for-one replacement of conventional 500 kHz analog FM STL systems.

DP5503A/5504

- Simultaneous transmission capacity for four 15 kHz bandwidth program channels, ideally suited for LMA/duopoly applications.
- The two RS-232 data channels will accommodate up to 4800 baud serial communications for remote control and data transmission applications.
- Linear RF power amplifier technology and narrow 400 kHz transmission bandwidth provide for improved spectrum utilization and adjacent channel operation when compared to conventional analog FM STL systems.

All Models

- Dolby AC-2® digital audio source coding designed to meet the most stringent quality demands in professional digital audio applications.
- Improved path margin compared with conventional FM aural STL technology.
- Fully integrated, performance-optimized digital modulation scheme based on quadrature partial response technology.
- Wide dynamic range RF receiver with low noise, high third-order intercept, and good front-end and IF selectivity for inherent robustness to strong signal interference from adjacent channel and out-of-band signals.
- Full featured, high reliability, convection cooled design coupled with modular construction for ease of service in the field.

DETAIL FEATURES

Because of the DSTL system's innovative systems design, there are numerous features that go well beyond what you may be accustomed to with STLs of previous generations. Some of the major points are detailed below.

- **Full Complement of Program/Data Channels**

  The Broadcast Electronics DP5501A/5502 DSTL system accommodates four channels of program and auxiliary information. The left and right main channels are intended for program audio. The Aux channel can be used to carry SCA audio and has a bandwidth of 7 kHz. Alternatively this channel can be configured for carrying data of up to 9600 baud. The voice channel can be used to carry voice-grade audio or, without any modification, can carry FSK audio for remote control applications, for example.

The Broadcast Electronics DP5503A/5504 DSTL system accommodates four channels of full bandwidth program audio, and two RS-232 serial data channels. The audio channels can be used to carry two separate stereo programs, or up to four fully independent mono programs. The data channels can be independently configured for data transmission up to 4800 baud, including remote control applications.

- **Dolby AC-2® Audio Coding**

The extremely high audio quality of the program channels in the DSTL system is attributable to the use of Dolby AC-2® audio coding, one of the world's foremost audio coding designs. AC-2 ®is based on the more than 25 years of psychoacoustic research at Dolby, which has produced well-known analog noise reduction systems as well as the successful Dolby AC-1® audio coding system. The version of Dolby AC-2® used in the program channels of the DSTL provides low time delay by using a data rate of approximately 180 kbps per channel.

The DP5501A/5502 Auxiliary channel uses a modified version of Dolby AC-2® with a bandwidth of 7 kHz and a sample rate of 16 kHz. The sound quality is excellent.

The DP5501A/5502 Voice channel does not use Dolby AC-2®, but rather the G.721 voice standard, which is adequate for voice intelligibility and use with FSK remote control signals.

- **9-QPRS Digital Modulation**

The unique performance capabilities of the DSTL system are the result of combining Dolby AC-2® audio coding with advanced digital modulation techniques. This results in a system that provides extremely high audio quality while maintaining very high signal spectrum efficiency (DP5501A-250 kHz, DP5503A-400kHz occupied bandwidth) as well as immunity to interference and fade that exceeds analog radio STL systems. The modulation type is 9-QPRS, a method that maps digital data into 9 different points in level and phase. This provides a spectrum efficiency higher than schemes such as QPSK, which are less complex. But 9-QPRS is not so complex as to compromise the robustness of the DSTL system. More complex modulation schemes require higher carrier/noise ratios in order to demodulate the received signal with minimal errors. In essence, the chosen modulation scheme provides an excellent balance between spectrum efficiency and signal robustness.

- **Sythesizer and Ultra linear Power Amplifier**

  The sythesizer and power amplifier section employs a microprocessor controlled frequency sythesizer, a Q + I modulator, and single final RF power module device. The excellent IM performance, along with Dolby AC-2® encoding technology and 9-QPRS modulation, guarantees a transmission bandwidth of 250 kHz in the DP5501A and 400 kHz in the DP5503A.

- **Receiver**

  By incorporating the latest microwave devices and design techniques, the receiver in the DSTL exhibits considerable dynamic range capability. Sensitivity is accomplished with a low-noise, dual-gate, gallium arsenic (GaAs) microwave FET. Selectivity is accomplished with state-of-the-art ceramic block and surface-acoustic-wave (SAW) devices which exhibit excellent selectivity and temperature stability. The frequency synthesizer design is similar to the high stability design that is used in the DSTL transmitter.

- **Modular Construction**

  To facilitate maintenance in the field as well as to provide a convenient means for upgrading the system, the DSTL system employs modular construction techniques. All modules and boards are accessible from the front. The only board requiring disassembly of the unit is the mother board; however it contains no active circuitry and is thus not likely to require replacement.

  Of the removable modules, only the RF and the power supply modules require removal of fastening hardware attaching the modules to the rear panel. In addition, the short-length power supply and alarm boards share a front retaining bracket.

- **Diagnostics Capabilities / Hot Standby**

  The DSTL system, unique among STL equipment, has a full complement of diagnostic capabilities to aid in troubleshooting should there ever be a need for field service. Almost every module contains diagnostic circuitry that is tied to a summary alarm system. If a detectable fault occurs on a module, a **STATUS** indicator on the module changes from green to red. Furthermore, a **Summary Alarm** indicator lights on the front panel of the DSTL unit and a Form C relay closure is activated on the rear panel. This status indication can be used in conjunction with a remote control system to alert station personnel that a fault has occurred. A troubleshooting section is included in this manual. Other Form C relay closures are also provided to indicate the operational status of the DSTL unit.

  The **Summary Alarm** indicator, as well as other status voltages, are tied into a Hot Standby system. Separate Hot Standby units can be installed to provide backup between two pairs of DSTL systems, or by taking appropriate systems considerations into account, conventional analog STLs can provide the backup.

- **Digital Repeater**

  In installations where a repeater is unavoidable, the DSTL system allows built-in capability for repeater operation. As an improvement over analog systems, this repeater capability is performed in the digital domain. Converting a DSTL transmitter and receiver for repeater operation is easily performed in the field by the simple movement of jumpers on the transmitter and receiver. For dedicated repeater operation that does not require demodulated audio at the repeater site, the DP5500 series is available with audio modules deleted. Be sure to read Appendix C for systems issues relating to repeater operation.

- **Digital Stereo Generator**

  The DSTL receiver accepts a digital stereo generator, Cat. No. 460. Performance is excellent, and setup and alignment features are comprehensive. As part of the DSTL receiver, it receives digital audio from the Audio Decoder module, thus keeping the signal in the digital domain and eliminating additional stages of D/A and A/D conversion.

This enhancement of the DSTL system permits added flexibility in configuring a station's operation by allowing the audio processing to remain at the studio. It is a high quality, **digital** substitute for analog stereo generators often supplied as part of the audio processing equipment.

- **Convenience Features**

The DSTL system incorporates subtle touches to make its use more convenient. For example:

1. Alignment LEDs are provided for initial audio level calibration.

2. Headphone monitors are provided on both transmitter and receiver.

3. RF (transmitter) and IF (receiver) monitor points are provided for operational monitoring and troubleshooting. Test modes can be activated and test points are also provided on certain modules.

4. The front panel itself snaps off for quick access to the user controls. An alignment tool ("tweaker") is provided, with a storage location on the back side of the front panel.

5. On the subpanel, an area is provided to note the operating frequency of the DSTL system along with the 3-digit code used to program the frequency synthesizer's switch settings S1, S2, and S3.

6. On the rear panel, numerous notices are included as operational reminders.

REGULATORY NOTICES

FCC

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

The DP5501A/5503A DSTL Transmitters have received type notification pursuant to part 74.550 of the FCC rules. The FCC ID for the DP5501A is JGLDSTL5501-2W and the DSTL emission type is 250KD9W. The FCC ID for the DP5503A is JGLDSTL5503 and the DSTL emission type is 400KD9W.

UL

Troubleshooting must be performed by trained technicians. Do not attempt to service the transmitter or receiver unless you are qualified to do so.

WARNING: Check that the units have been set to the correct supply voltage and that the correct fuses have been installed. To reduce the risk of fire, replace the fuses only with the same type and rating.

REGULATORY NOTICES (Con't)

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.

IEC NOTICES

IMPORTANT SAFETY NOTICE

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

(GB)

- o Ensure the **voltage selector** is set to the correct mains voltage for your **supply**.
- o Ensure **fuses** fitted are the **correct rating and type** as marked on the unit.
- o The unit **must be earthed** by connecting to a correctly wired and **earthed** power outlet.
- o The **power cord** supplied with this unit must be wired as follows:

Live—Brown Neutral—Blue Earth—Green/Yellow

IMPORTANT – NOTE DE SECURITE

Ce materiel est conforme à la norme IEC65. Pour vous assurer d'un fonctionnement sans danger et de prévenir tout choc électrique ou tout risque d'incendie, veillez à observer les recommandations suivantes.

(F)

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

Phase—Brun Neutre—Bleu Terre—Vert/Jaune

WICHTIGER SICHERHEITSHINWEIS

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

(D)

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

Phase—braun Nulleiter—blau Erde—grün/gelb

NORME DI SICUREZZA – IMPORTANTE

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

(I)

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

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

AVISO IMPORTANTE DE SEGURIDAD

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

(E)

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

Vivo—Marrón Neutro—Azul Tierra—Verde/Amarillo

VIKTIGA SÄKERHETSÅTGÄRDER!

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

(S)

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

Fas—Brun Neutral—Blå Jord—Grön/Gul

IEC NOTICES

BELANGRIJK VEILIGHEIDS-VOORSCHRIFT:

Deze unit voldoet aan de IEC65 veiligheids-standaards. Voor een veilig gebruik en om het gevaar van electrische schokken en het risico van brand te vermijden, dienen de volgende regels in acht te worden genomen:

(NL)

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

Fase—Bruin Nul—Blauw Aarde—Groen/Geel

SECTION 1
QUICK START

This section has been prepared for those with prior experience installing a Broadcast Electronics DSTL system or for those desiring a quick overview of the steps required in the proper installation of a DSTL transmitter and receiver. References have been provided to sections of the manual that contain complete information. Please consult the appropriate section for any additional assistance before contacting Brodcast Electronics - Marti Facility.

Unpacking

Unpack the unit carefully. Be sure to save all packing material for possible future use. Check to make sure no accessories are missing. Refer to Section 3.1.

Inspection

Inspect for damage by performing a visual inspection of the unit and modules, if deemed necessary. Refer to Section 3.2.

Fusing

Check that the units have been set to the correct supply voltage and that the correct fuse values have been installed. Refer to Section 3.3.

System Jumper Configuration

CAUTION! Power must not be applied when removing or replacing modules.

The default jumper settings assume the following operating configuration. Modify jumper settings per Section 3.4 as needed.

Input Impedance	**10k** ohms.
Pre-Emphasis	**Not engaged** (Pre-emphasis is not added to the main channels by the DSTL unit).
Aux Channel Use (DP5501A/5502 only)	**Audio** (not modem).
Data Channel Use (DP5503A/5504 only)	**4800** baud, 8 data bits, **No parity**.
Signal Adaptation	**Emphasized** (assumes pre-emphasized audio is present at the left and right inputs).

Composite Source (DP5504 only)	**Left 1, Right 1**.
RF Power Amp Over-Temperature Shutdown	**Defeated**.
DSTL Application	**Normal** (non-repeater operation).
Rear Panel MTR/BER Signal	A dc voltage proportional to the **meter reading** appears on rear panel connector TB102 (for serial numbers greater than 21).

Programming Carrier Frequency

Using the table in Appendix A, set the transmitter and receiver frequencies to your operating frequency, if the factory or distributor has not already done so. You may want to jot down the frequency and hex code in the space provided on the upper left corner of the sub-panel.

Bench Testing/Checkout (Optional)

If desired, perform back-to-back testing to confirm operation prior to installation. Refer to Section 3.7.

Installation

Install the transmitter and receiver in their respective racks (see Section 4.1). Make audio, RF and status/remote control connections as necessary (see Sections 4.2 and 4.3). If necessary, also make repeater and hot standby connections. See Sections 4.2.6 and 4.2.7 (transmitter) and Sections 4.3.7 and 4.3.8 (receiver).

Operation

Place units in **OPERATE**. Adjust transmitter output power (Section 6.2) and verify receiver signal strength (Section 6.3) and error rate (Section 6.4).

Setup

Calibrate audio levels per Section 7, including the digital stereo generator, if installed.

2.1 Introduction

To achieve optimum performance from the Broadcast Electronics DSTL system, you will want to familiarize yourself with the enhanced capabilities and features of this product as compared with analog STLs with which you may already be familiar. In addition, the digital nature of the DSTL system involves certain operational considerations.

2.2 Comparison with Composite Installations

The Broadcast Electronics DSTL system carries left and right channels in a discrete fashion, not as a composite signal. However, the ability of the DSTL to accept fully processed, pre-emphasized audio, in conjunction with the built-in digital stereo generator, makes for a "virtual composite" link. All processing can remain at the studio, and the DSTL receiver output is a composite signal (See Figure 2.1). L and R signals are also available for secondary use.

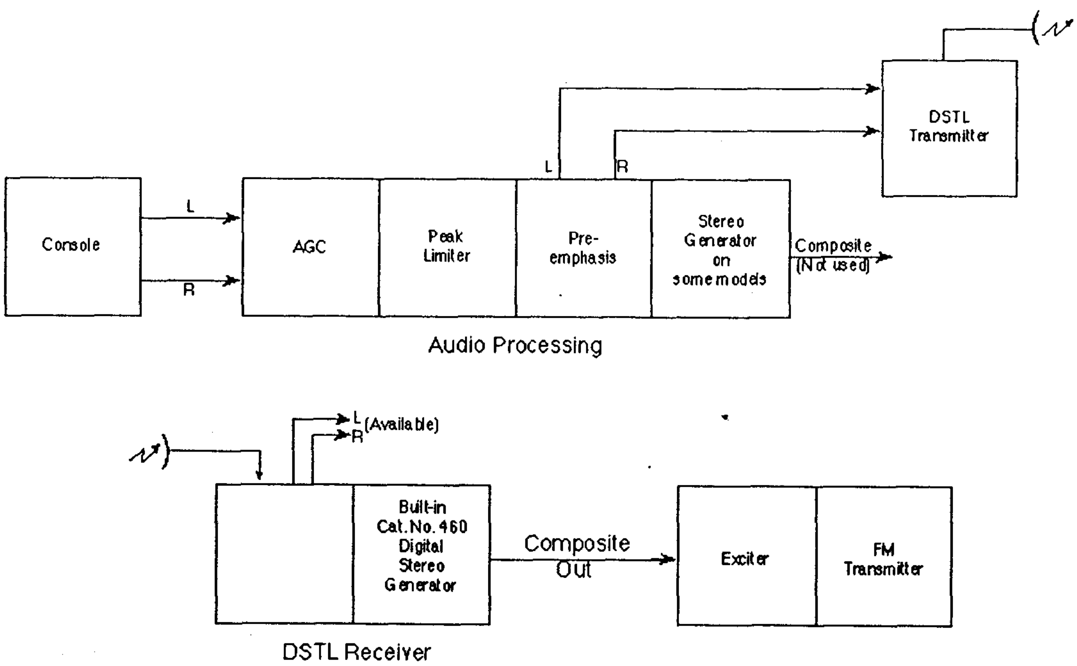


Fig. 2.1 **DSTL with internal digital stereo generator creates a "virtual composite" link**

The DP5501A/5502 DSTL system also accepts Aux and Voice signals at baseband, rather than on subcarriers. In these cases, the subcarrier generator for SCA use must be at the transmitter site, and the remote control system should be configured to accept an FSK signal (See Figure 2.2).

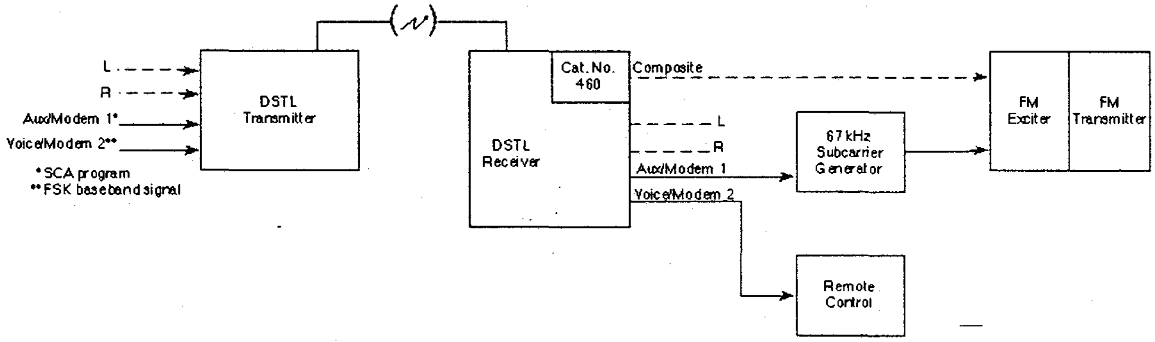


Fig. 2.2 Typical usage of Aux and Voice channels with DP5501A/5502

The DP5503A/5504 DSTL system also accepts two RS-232 data signals at data rates up to 4800 baud for remote control and data transmission applications.

2.3 Discrete Channel Operation

In installations where left and right channels are carried discretely, the DSTL system can handle unprocessed, partially processed, or fully processed audio as station requirements dictate. Pre-emphasis can be applied within the DSTL system if necessary. Final processing and an external stereo generator typically follow the DSTL receiver in the signal chain.

2.4 Location of Audio Processing and Stereo Generator

An STL installation incorporating the Broadcast Electronics DSTL gives you the flexibility to locate your audio processing either at the studio or at the transmitter, or both, depending on your preferences or operational practices.

The wide dynamic range of the DSTL system, as well as its freedom from overshoots when handling processed audio programming, permits complete flexibility in configuring the audio signal path to suit station requirements. Some, none, or all of the processing may precede or follow the DSTL system.

Should you keep part or all of the audio processing at the studio, an overshoot limiter ensures that the DSTL system can transmit your signal intact (see below). The left and right outputs of the DSTL receiver can then be fed to the Cat. No. 460 digital stereo generator module, or even to a stand-alone stereo generator, if desired.

For those audio processors that include a built in stereo generator and are located at the studio, provision must be made to obtain the signal just prior to the stereo generator and feed it to the inputs of the DSTL transmitter. One of the most popular audio processor/stereo generators on the market today has "test jacks" that perform this function very well. The original stereo generator located at the studio is replaced with the Cat. No. 460 digital stereo generator. The DSTL system includes jumper options to accommodate pre-emphasized or flat signals.

2.5 Audio Signal Overload: Safety Limiter

While a digital audio system has a much wider dynamic range than analog FM channels, a digital system exhibits a hard clip point. The Main and Auxiliary channels of the DSTL are provided with limiters which limit the input signal before the analog-to-digital converters can be driven into digital overload. This limiting is designed to improve the sound quality if the DSTL is driven with input signals above 0 dB as measured on the input level meters, and to protect the converter inputs from being driven beyond their supply rails. The limiting action is relatively soft and adds no additional distortion for signals below 0 dB on the meter, yet fully limits at +1.5 dB.

Adjustment of input levels should be done with some care, especially when feeding the DSTL transmitter signals with little or no audio processing. The DSTL system includes calibration LEDs for setup that provide for 12 dB of headroom between nominal 0 dB operating level and overload, sufficient for most program material. Operating at too low a signal level may compromise signal-to-noise ratio.

For installations with part or all of the processing at the studio, the peak to average ratio is lower; therefore, less headroom is required.

2.6 Overshoot Limiter

The DSTL incorporates an overshoot limiter in the Decoder module (Cat. No. 463/483). This limiter is used to prevent an increase in the audio peak level due to the interaction between any loudness enhancement processing and the AC-2® coding system. Although the AC-2® data reduction process does not degrade perceived audio quality, some musical signals experience a small increase in overall peak levels (e.g., 1–2 dB), especially if they have been heavily processed. Therefore, the limiter is useful in applications requiring tight control of peak levels and when aggressive audio processing is performed before the input to the STL.

Proper use of the limiter is accomplished by applying the processed audio to the program channel inputs and setting the peak program level to 0dB. Under this condition the limiter occasionally operates to prevent the decoded program level from exceeding 100% modulation. Careful attention has been given to the limiter design so that these occasional modifications in level produce negligible audible effect.

The limiter is implemented entirely in the digital domain and employs the same powerful psychoacoustic techniques that have resulted in the successful development of the AC-2® coding technology. Overmodulation is prevented by occasionally gain reducing spectral components above 5 kHz in such a way that any audible artifacts and reduction of loudness are negligible.

2.7 Time Delay

The DSTL system exhibits a 9 millisecond time delay. When announcers monitor themselves using the off air signal, the comb-filter effect caused by the interaction of the direct, bone-conducted sound and the time-delayed sound from their headphones can cause timbral changes in their speech. They should be able to adjust to this effect with minimal training.

If it is important that they hear themselves without unusual effects, you may need to modify your monitoring procedures: Either announcers can monitor the program bus at all times; or monitor by means of a switching system that would be added in order to monitor off-air *except* when when the mic is open, at which time they would monitor the program bus.

2.8 Meter Ballistics

The meter circuits of the DSTL employ a two stage peak hold system that allows the panel meter to accurately indicate peak program levels to within 0.5 dB if they last more than 50 microseconds, and to hold the reading for between 220–300 milliseconds.

All other meter switch selections measure DC voltages from various modules and are subject to only the second stage of the meter conditioning process. As a result, when the meter selection knob is rotated rapidly, you will notice that the highest meter indication appears to "stick" for 220–300 milliseconds before acquiring its final value. This is caused by the second stage of meter conditioning and represents normal operation.

2.9 Hot Standby

The DSTL system includes hot standby capabilities with either another DSTL system, or with analog composite systems (certain systems considerations must be taken into account. See below). In addition, the DSTL receiver can operate with another receiver (DSTL or analog composite) in a master/slave mode, whereby operation is relinquished to the secondary unit in the event of a problem with the DSTL receiver. For installation, refer to the installation instructions supplied with the Hot Standby units or Appendix D for master/slave DSTL receiver operation.

2.10 DSTL System with Analog Composite Backup

Because of the capability of operating as a "virtual" composite system, an existing analog composite STL system can be set-up as a backup for the DSTL system, if accommodation is made for subcarrier operations. As mentioned above, the DP5501A/5502 DSTL system accepts baseband Aux and Voice signals, rather than subcarriers. Therefore, a hybrid DSTL-main/composite-backup system requires the addition of subcarrier generators at the transmitter site (See Figure 2.3).

Note The Hot Standby units provide a relay closure to indicate switchover. This closure can be used to control the external switching of subcarrier signals.

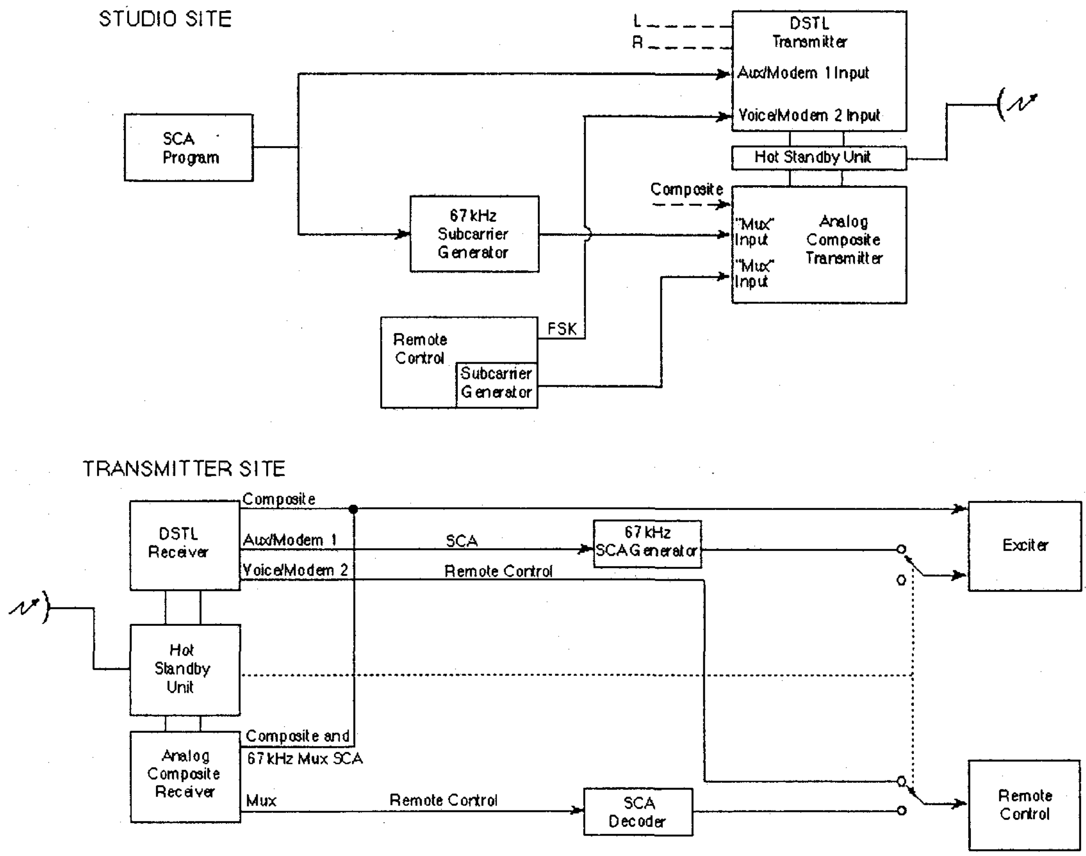


Fig. 2.3 DP5501A/5502 DSTL/Composte STL—Subcarrier Aspects

2.11 Spectrum Usage (DP5501A/5502)

The DSTL system allows new opportunities for channel allocation in areas of crowded STL spectrum. Because the occupied bandwidth is only 250 kHz, two DSTL channels can occupy the spectrum previously required for a single wideband composite analog STL system. Meanwhile, the lower signal operating requirements and the DSTL's tightly controlled spectrum permit friendly co-existence with analog systems.

2.12 Path Considerations

For existing paths, the DSTL system can usually be substituted for existing analog equipment without undue concern. However, a path analysis should be performed prior to installation to confirm that adequate fade margin has been designed into the system. A DSTL system does not defy the laws of physics!

Path calculations for the DSTL are identical to analog systems; only the numbers are different. Transmitter power output is 2 watts (+33dBm) and DP5502 receiver sensitivity (at squelch) is 2μV (–100dBm); DP5504 sensitivity is -96dBm. Because of the need for relatively low carrier-to-noise (C/N) ratios, the transmitter power output is lower than with analog systems; however the receiver sensitivity is much higher. Please refer to Appendix B for information on conducting a thorough path analysis.

3.1 Unpacking

The Broadcast Electronics DSTL system has been shipped in two cartons, one for the transmitter and one for the receiver. A copy of the manual and an accessory kit has been provided in both cartons for your convenience.

Before proceeding further, be sure to inspect the outer cartons for shipping damage. If there has been any penetration of the carton, be sure to inspect the units for any physical damage in those areas. The cartons have been designed to prevent damage to the units during transit, and at least one carton should be kept as protection for any unit that may require shipment in the future.

Several accessories are inlcuded in the cartons. Please compare them with the following list to ensure that there are no missing items:

Manual
Power cord
Alignment tool
Spare fuses
Spare subpanel screws and washers
Repeater cable (Repeater units only)
RF jumper cable (DP5501A/DP5503A only)

3.2 Inspection

The front cover of the DSTL unit has been packed in the accessory box. Remove, inspect and set it aside. Next carefully remove the DSTL from the carton. Remove the plastic wrapping and place on a flat surface. Before applying ac power, we strongly suggest that you gain access to the plug-in modules and inspect them for any possible damage or dislodging. Proceed as follows:

3.2.1 Access to Modules

To access the modules, you will need a #1 Phillips head screwdriver and a ¾" nut remover (or small crescent wrench) for rear panel fasteners.

Remove the screws around the periphery of the sub-panel bearing the legends for the controls and trimmers. By grasping the protruding meter section, gently pull the sub-panel and the meter assembly forward, then remove the connector for the ribbon cable behind the meter. Carefully set aside this sub-assembly.

Note In order to remove the power amplifier module on the DSTL transmitter or the receiver module on the DSTL receiver, you will need to remove the nut holding the RF "N" connector to the rear panel.

To remove the Power Supply modules on both units, you will need to remove two screws that secure the ac power entry socket to the rear panel. Also, remove the two screws associated with the retaining bracket located just in front of these modules (the bracket also secures the Alarm/Control module).

Gently unplug and re-seat each of the modules in turn (some modules are provided with pull-tabs).

If there are no signs of physical damage, proceed to "Fusing Information" below. Do not replace the sub-panel at this stage, as you may need to move jumpers per Section 3.4.

3.2.2 Claims for Shipping Damage

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

3.3 Fusing Information

WARNING Check that the units have been set to the correct supply voltage and that the correct fuse values have been installed. To reduce the risk of fire, replace the fuse only with the one of the same type and rating.

---

110 Vac System (Uses Cat. No. 457 Power Supply module):

Model DP5501A/5503A Transmitter: Use T2.0A 250V 20mm time lag fuse.
Model DP5502/5504 Receiver: Use T1.0A 250V 20mm time lag fuse.

240 Vac System (Uses Cat. No. 457-240 Power Supply module):

Model DP5501A/5503A Transmitter: Use T1.0A 250V 20mm time lag fuse.
Model DP5502/5504 Receiver: Use T630 ma 250V 20mm time lag fuse.

The power input connector on the Model DP5501A/5503A or DP5502/5504 power supply has a fuse drawer which accepts a fuse carrier for 20 mm fuses. Select the appropriate fuse and insert it into the fuse carrier. (**Note:** a spare fuse of the same type and rating can be stored in the right-hand compartment of the drawer.) When inserting the fuse carrier into the drawer, make sure that the correct ac voltage shows through the drawer window.

3.4 Jumper/Switch Settings

CAUTION Remove AC power prior to removing or replacing modules.

Numerous jumper and switch settings are provided in order to provide the greatest flexibility of installation. Certain default settings, however, have been established, which satisfy the operational requirements of many stations. The default settings assume the following:

Audio Input Impedance	**10k** ohms.
Pre-Emphasis	**Not engaged** (Pre-emphasis is not added to the main channels by the DSTL unit).
Aux Channel Use	**Audio** (not modem). [DP5501A/5502 only]
Data Channel Use	**4800** baud, 8 data bits, **No parity** [DP5503A/5504 only]
Signal Adaptation	**Emphasized** (assumes pre-emphasized audio is present at the program audio inputs, or that the Pre-emphasis jumpers are enabled).
Composite Source	**Left 1, Right 1** [DP5504 only]
RF Power Amp Over-Temperature Shutdown	**Defeated**.
DSTL Application	**Normal** (non-repeater operation).
Rear Panel MTR/BER	A dc voltage proportional to the **meter reading**

Signal appears on rear panel connector TB102 (for serial numbers greater than 21).

Note The default settings are shown in the jumper/switch descriptions below within [brackets].

The actual labels printed on the circuit boards may not be identical to the labels in the manual figures. The labels shown in the manual figures were chosen to clearly describe the jumper/switch functions. The jumper/switch positions in the manual figures match the positions on the boards.

For future reference:

Jot down in pencil the jumper/switch positions for your particular installation.

Transmitter

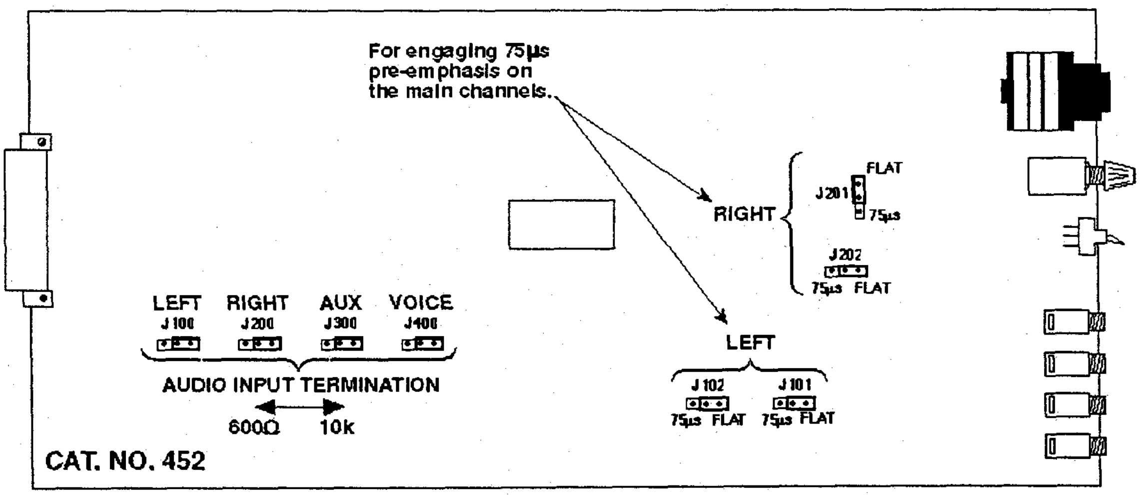
Module	Cat. No.	Jumper/ Switch	Default Position	Actual Position	By	Date
A/D Converter	452/472	J100	10k			
		J200	10k			
		J300	10k			
		J400	10k			
	452	J101/102	FLAT			
		J201/202	FLAT			
	472	J101/102/103	FLAT			
		J201/202/203	FLAT			
		J301/302/303	FLAT			
		J401/402/403	FLAT			
Audio Encoder	453	J702	AUDIO			
		J703	EMPH			
	473	S700-1/700-2	OFF OFF			
		S700-4/700-5	OFF OFF			
		S700-8	ON			
		S701-1/701-2	OFF OFF			
		S701-4/701-5	OFF OFF			
		S701-8	ON			
Modulator	454/474	J3/J4	SYS			
Freq Syn/Power Amp	476	J200	OFF			
Alarm/Control	458	J102	NORM			

Receiver

Module	Cat. No.	Jumper/Switch	Default Position		Actual Position		By	Date
Audio Decoder	463	J702	AUDIO					
		J703	EMPH					
	483	S700-1/700-2	OFF	OFF				
		S700-4/700-5	OFF	OFF				
		S700-8	ON					
		S701-1/701-2	OFF	OFF				
		S701-4/701-5	OFF	OFF				
		S701-8	ON					
Alarm/Control	458	J102	NORM					
		J104	METER					
Backplane	---	J1	LR1					
		J2	LR1					

3.4.1 Transmitter Jumpers/Switches

Cat. No. 452 A/D Converter Jumpers [DP5501A Only]



J100, 200, 300, 400 Audio Input Termination: 600 ohms/[10k]

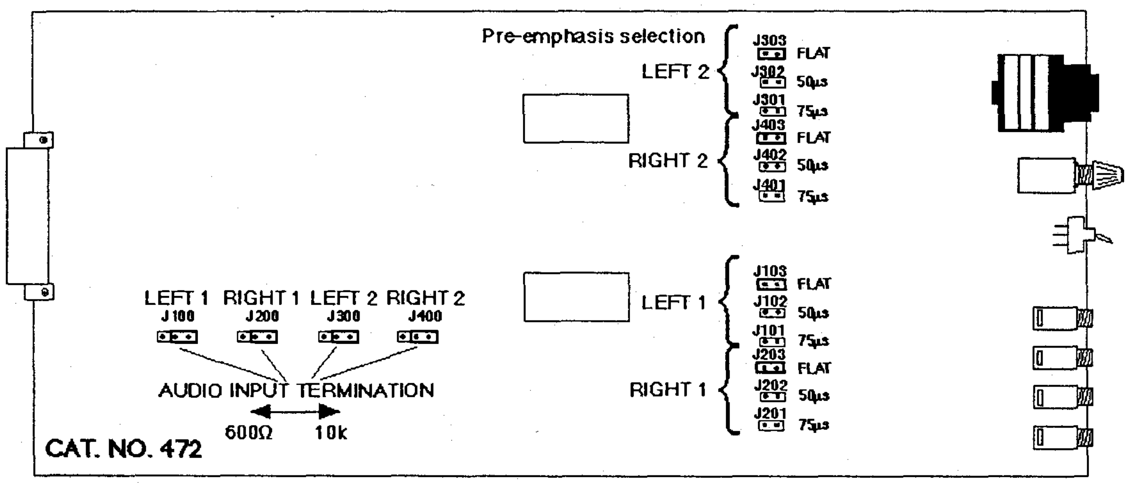
The output sections of most modern audio electronics equipment are designed with low output impedance and are intended to drive high input impedance loads. If that is the case, leave these jumpers in the **10k** position. If, on the other hand, you have an output device that requires a 600 ohm load, move the jumpers to the **600 ohm** position.

Transmitter Jumpers, cont.

J101, 102 & J201, 202 Pre-Emphasis: [flat]/75µs (50µs in some countries)

These jumpers engage 75µs (50µs) pre-emphasis. Place the jumpers in the **75µs** position only if there is no other means in the audio signal path to apply pre-emphasis. Otherwise, leave these jumpers in the **flat** position.

Cat. No. 472 A/D Converter Jumpers [DP5503A Only]



J100, 200, 300, 400 Audio Input Termination: 600 ohms/[10k]

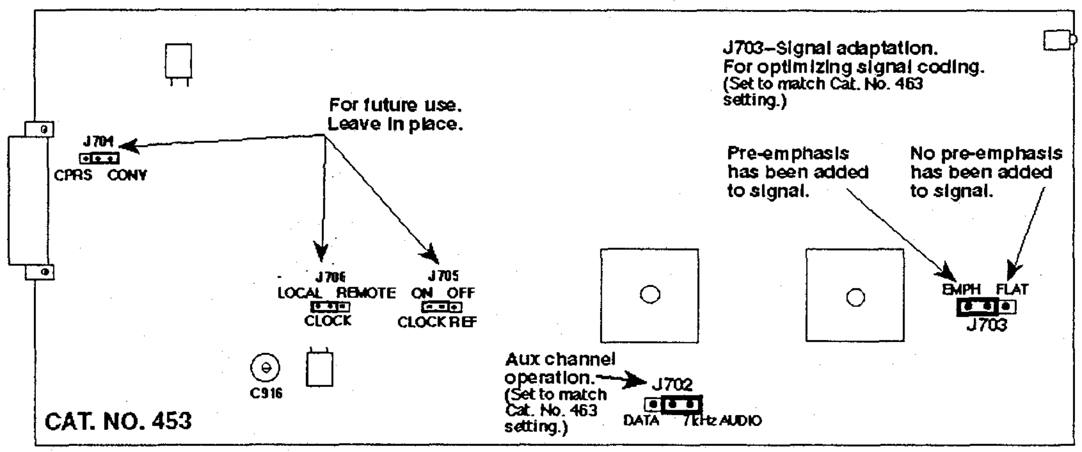
The output sections of most modern audio electronics equipment are designed with low output impedance and are intended to drive high input impedance loads. If that is the case, leave these jumpers in the **10k** position. If, on the other hand, you have an output device that requires a 600 ohm load, move the jumpers to the **600 ohm** position.

J101/2/ 3, J201/2/3, J301/2/3, J401/2/3
Pre-Emphasis: [flat]/75µs/50µs

These jumpers engage 75µs or 50µs pre-emphasis. Place the jumpers in the **75µs** or **50µs** position only if there is no other means in the audio signal path to apply pre-emphasis. Otherwise, leave these jumpers in the **flat** position.

Transmitter Jumpers, cont.

Cat. No. 453 Audio Encoder Jumpers [DP5501A Only]



J702 Aux Channel Use: DATA/[AUDIO]

This jumper (along with a comparable jumper in the receiver) optimizes the auxiliary channel for audio or data capabilities. If audio programming is utilized on the auxiliary channel, leave the jumper in the **7 kHz AUDIO** (SCA) position. On the other hand, if data is to be transmitted on the auxiliary channel, move the jumper to the **DATA** (modem) position.

J703 Signal Adaptation: [EMPH]/FLAT

This jumper (along with a comparable jumper in the receiver) is used to tailor the response in the digital domain for optimum performance of the Dolby AC-2 audio coding system. The jumper is placed in the **EMPH** position under the following conditions:

a) That there is audio processing ahead of the DSTL system that applies pre-emphasis;

or

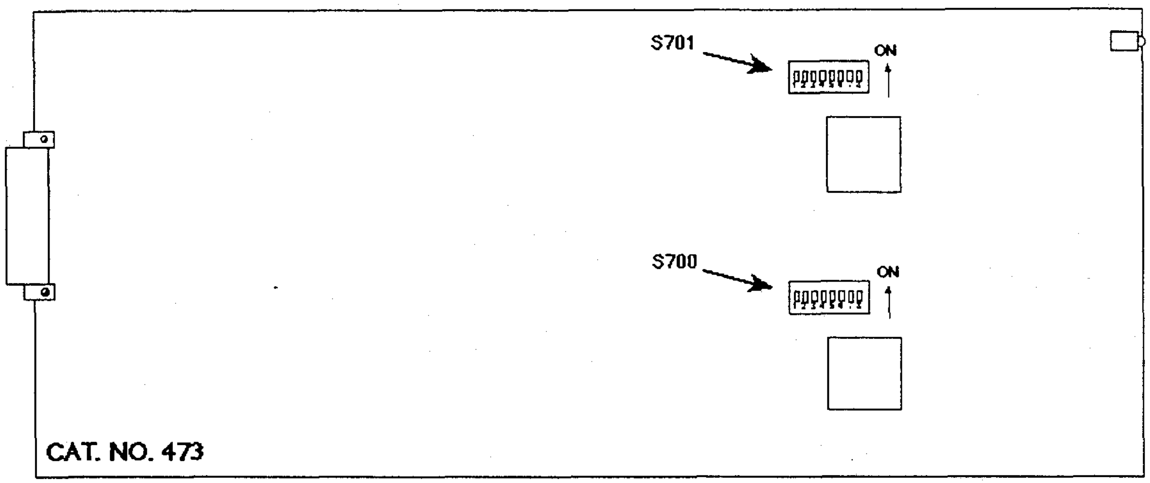
b) That the pre-emphasis jumpers J101, 102 and J201, 202 on the Cat. No. 452 module (see above) are set to apply pre-emphasis to the signal.

If neither of these conditions is present, place the jumper in the **FLAT** position.

NOTE Other jumpers not specifically mentioned are for factory use only and should not be disturbed.

Transmitter Jumpers, cont.

Cat. No. 473 Audio Encoder Switches [DP5503A Only]



S700-1&2 and S701-1&2 Data Baud Rate: [4800]/2400/1200/300
These switches are used to set the input baud rates of Data Channel 1 (S700) and Data Channel 2 (S701) according to the following table:

Baud Rate	Baud A Switch S700-1 / S701-1	Baud B Switch S700-2 / S701-2
4800	OFF	OFF
2400	ON	OFF
1200	OFF	ON
300	ON	ON

Switch settings should be selected to match the baud rates of the attached data terminal equipment. Baud rate settings in the DP5503A should be less than or equal to the comparable settings in the DP5504. Use of a higher rate in the transmitter could produce an overflow condition in the data channel, with the resultant loss of data characters.

Transmitter Jumpers, cont.

S700-4&5 and S701-4&5 Data Parity: [NONE]/EVEN/ODD

These switches are used to set the input data parity of Data Channels 1 (S700) and 2 (S701) according to the following table:

Data Parity	Mode A Switch S700-4 / S701-4	Mode B Switch S700-5 / S701-5
None	OFF	OFF
Even	ON	OFF
Odd	OFF	ON
-----	ON	ON

Switch settings should be selected to match the parity modes of the attached data terminal equipment; data character length is 8 bits in all modes. DP5503A and DP5504 parity may be set independently with the following exception: if transmit and receive baud rates are identical and DP5503A parity is set to "None", then DP5504 parity should also be set to "None" to avoid the possibility of data loss. (The addition of an output parity bit reduces the maximum output character rate relative to the input character rate, and could result in data channel overflow.)

S700-8 and S701-8 Signal Adaptation: [EMPH]/FLAT

These switches (along with comparable switches in the receiver) are used to tailor the response in the digital domain for optimum performance of the Dolby AC-2 audio coding system. S700-8 selects adaptation for the Left 1 / Right 1 channel pair, and S701-8 provides the same function for Left 2 / Right 2. These switches are placed in the **EMPH** [ON] position under the following conditions:

a) That there is audio processing ahead of the DSTL system that applies pre-emphasis;

or

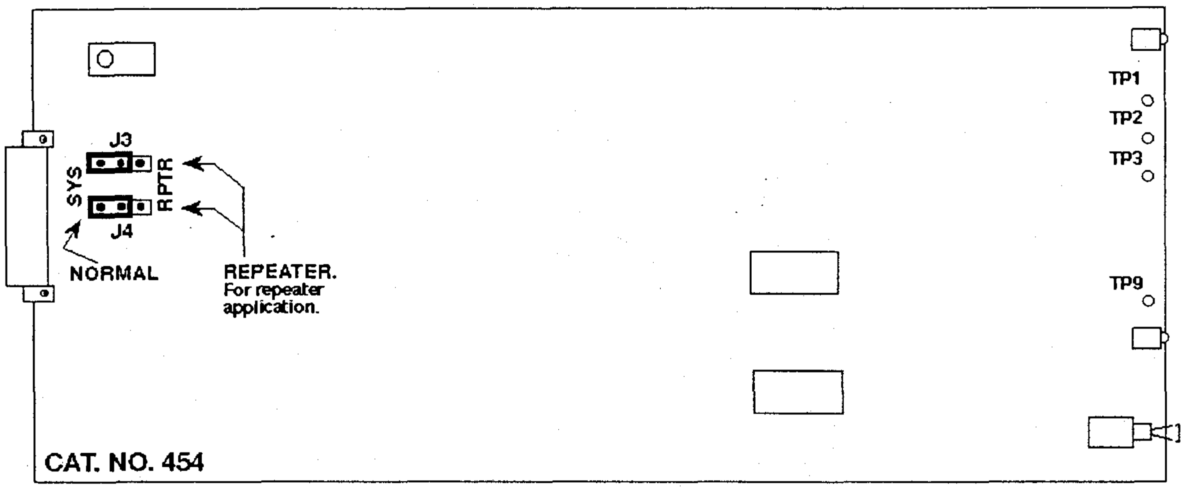
b) That the pre-emphasis jumpers J101-103, J201-203, J301-303, and J401-403 on the Cat. No. 472 module (see above) are set to apply pre-emphasis to the signal.

If neither of these conditions is present, place the switches in the **FLAT** [OFF] position.

NOTE Other jumpers not specifically mentioned are for factory use only and should not be disturbed.

Transmitter Jumpers, cont.

Cat. No. 454/474 Modulator Jumpers



J3/J4 DSTL Application: REPEATER/[SYS (Normal)]

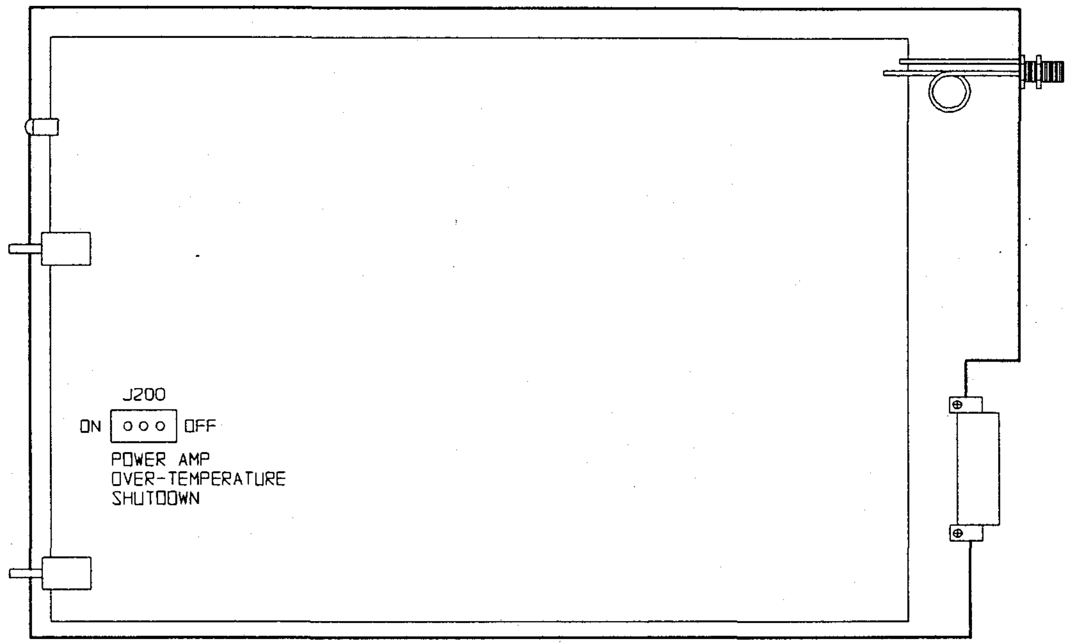
If the DSTL transmitter is used as a digital repeater, move these jumpers (along with jumper **J102** on the Cat. No. 458 Alarm/Control module and a jumper on the receiver) to the **RPTR** position.

Otherwise leave the jumpers in the **SYS** (normal) position.

NOTE Other jumpers not specifically mentioned are for factory use only and should not be disturbed.

Transmitter Jumpers, cont.

Cat. No. 476 Frequency Sythesizer/Power Amplifier Jumper



Cat. No. 476

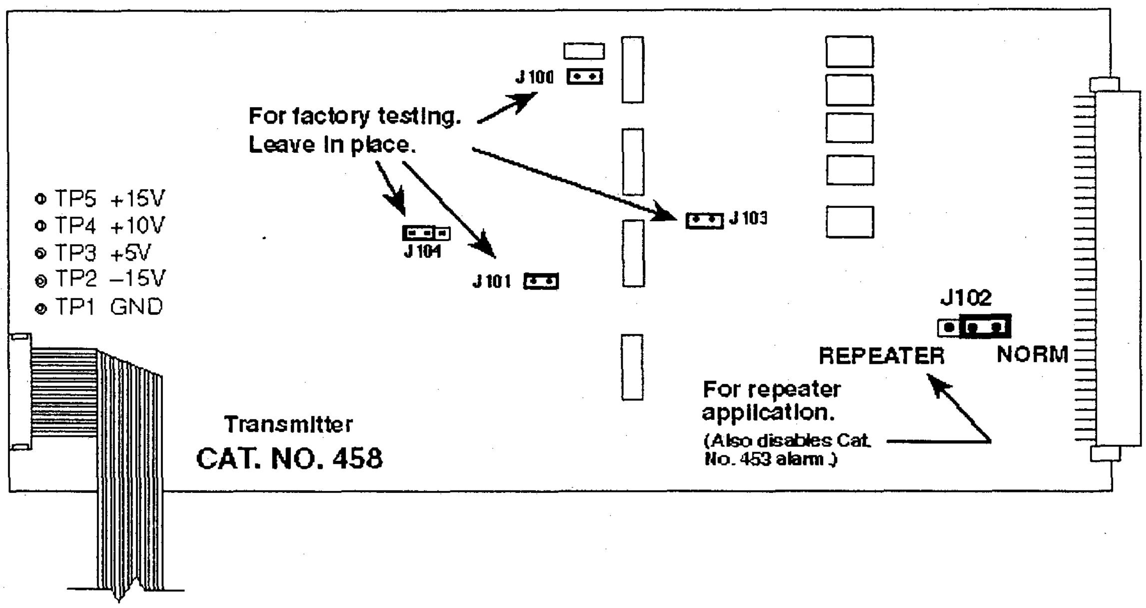
J200 RF Power Amp Over-Temperature Shutdown: ON/[OFF]

This jumper is used to select whether or not the transmitter shuts down if the RF power amplifier heat sink temperature rises above 90° C, thus preventing possible self-destruction. A **MODULE FAULT** and **SUMMARY ALARM** will occur with the jumper in either position. Move this jumper to **ON** if you also want the transmitter to shut off for this condition.

Note To gain access to this jumper, remove the screws on the shield cover and carefully lift off the cover. Be sure to replace the cover before re-installing the module in the DSTL chassis.

Transmitter Jumpers, cont.

Cat. No. 458 Alarm/Control Jumper



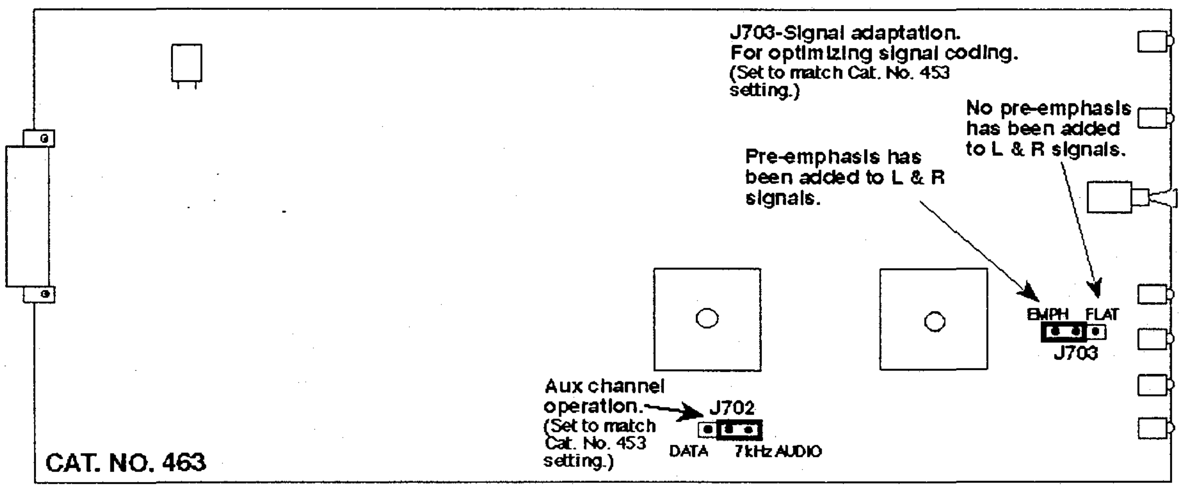
J102 DSTL Application: Repeater/[Normal]

If the DSTL transmitter is used as a digital repeater system, and a Cat. No. 453/473 Encoder module is not installed, move this jumper (along with jumpers J3 & J4 on the Cat. No. 454, and a jumper on the receiver) to the **REPEATER** position. This will disable the alarm which would occur due to the absence of a Cat. No. 453/473 Encoder module. Otherwise leave in the **NORM** position.

Note Other jumpers in the transmitter not specifically mentioned are for factory use only and should not be disturbed.

3.4.2 Receiver Jumpers/Switches

Cat. No. 463 Audio Decoder Jumpers [DP5502 Only]



J703 Signal Adaptation: [EMPH]/FLAT

This jumper (along with a comparable jumper in the transmitter) is used to tailor the response in the digital domain for optimum performance of the Dolby AC-2® audio coding system. The jumper is placed in the **EMPH** position under the following conditions:

a) That there is audio processing ahead of the DSTL system that applies pre-emphasis;

or

b) That the pre-emphasis jumpers J101, 102 and J201, 202 on the Cat. No. 452 module (in the transmitter) are set to apply 75µs pre-emphasis (50µs in some countries) to the signal.

If neither of these conditions is present, place the jumper in the **FLAT** position.

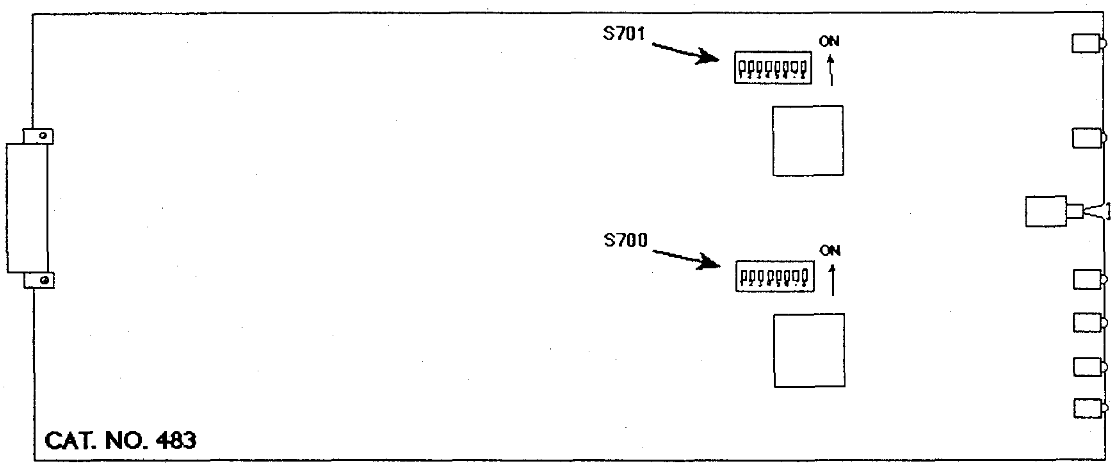
Receiver Jumpers, cont.

J702 Aux Channel Use: DATA/[AUDIO]

This jumper (along with a comparable jumper in the transmitter) optimizes the auxiliary channel for audio or data capabilities. If audio programming is utilized on the auxiliary channel, leave the jumper in the **7 kHz AUDIO** (SCA) position. On the other hand, if data is to be transmitted on the auxiliary channel move the jumper to the **DATA** (modem) position.

NOTE Other jumpers not specifically mentioned are for factory use only and should not be disturbed.

Cat. No. 483 Audio Decoder Switches [DP5504 Only]



S700-1&2 and S701-1&2 Data Baud Rate: [4800]/2400/1200/300

These switches are used to set the output baud rates of Data Channel 1 (S700) and Data Channel 2 (S701) according to the following table:

Baud Rate	Baud A Switch S700-1 / S701-1	Baud B Switch S700-2 / S701-2
4800	OFF	OFF
2400	ON	OFF
1200	OFF	ON
300	ON	ON

Receiver Jumpers, cont.

Switch settings should be selected to match the baud rates of the attached data terminal equipment. Baud rate settings in the DP5504 should be greater than or equal to the comparable settings in the DP5503A. Use of a lower rate in the receiver could produce an overflow condition in the data channel, with the resultant loss of data characters.

S700-4&5 and S701-4&5 Data Parity: [NONE]/EVEN/ODD

These switches are used to set the output data parity of Data Channels 1 (S700) and 2 (S701) according to the following table:

Data Parity	Mode A Switch S700-4 / S701-4	Mode B Switch S700-5 / S701-5
None	OFF	OFF
Even	ON	OFF
Odd	OFF	ON
-----	ON	ON

Switch settings should be selected to match the parity modes of the attached data terminal equipment; data character length is 8 bits in all modes. DP5503A and DP5504 parity may be set independently with the following exception: if transmit and receive baud rates are identical and DP5503A parity is set to "None", then DP5504 parity should also be set to "None" to avoid the possibility of data loss. (The addition of an output parity bit reduces the maximum output character rate relative to the input character rate, and could result in data channel overflow.)

S700-8 and S701-8 Signal Adaptation: [EMPH]/FLAT

These switches (along with comparable switches in the transmitter) are used to tailor the response in the digital domain for optimum performance of the Dolby AC-2 audio coding system. S700-8 selects adaptation for the Left 1 / Right 1 channel pair, and S701-8 provides the same function for Left 2 / Right 2. These switches are placed in the **EMPH** [ON] position under the following conditions:

a) That there is audio processing ahead of the DSTL system that applies pre-emphasis;

or

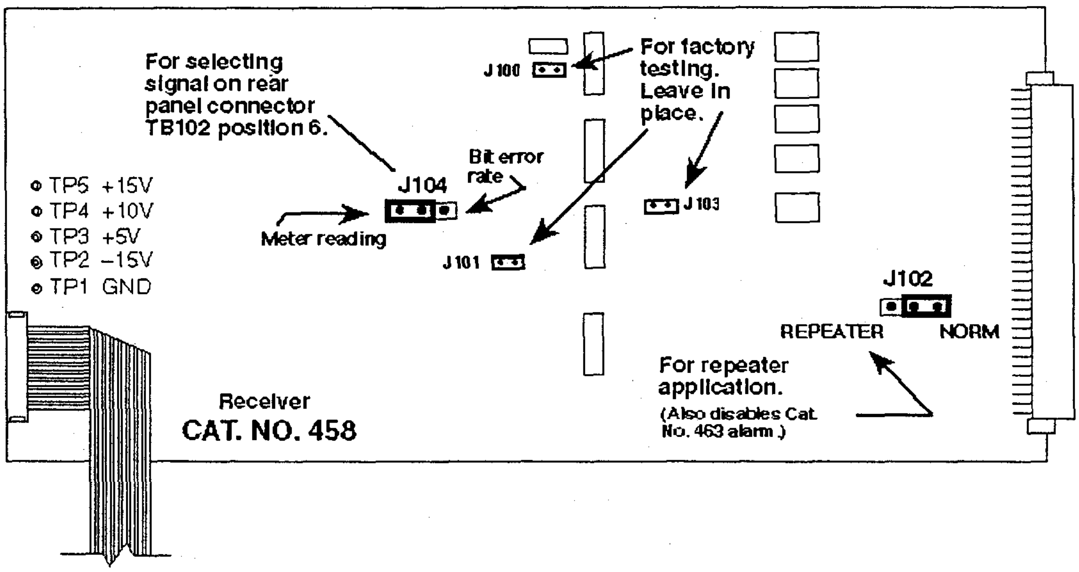
b) That the pre-emphasis jumpers J101-103, J201-203, J301-303, and J401-403 on the Cat. No. 472 module (see above) are set to apply pre-emphasis to the signal.

Receiver Jumpers, cont.

If neither of these conditions is present, place the switches in the **FLAT** [OFF] position.

NOTE Other jumpers not specifically mentioned are for factory use only and should not be disturbed.

Cat. No. 458 Alarm/Control Jumpers



J102 DSTL Application: Repeater/[Normal]
If the DSTL receiver is used as a digital repeater system and the Cat. No. 463/483 Decoder module is not installed, move this jumper (as well as jumpers on the repeater transmitter) to the **REPEATER** position. This will disable the alarm which would occur due to the absence of a Cat. No. 463/483 Decoder module. Otherwise leave in the **NORM** position.

J104 MTR/BER signal on rear panel TB102: [Meter]/Bit Error Rate

Note This jumper is incorporated in units with serial numbers greater than 21. For serial numbers below 21, only **BER** is available and no jumpers are involved.

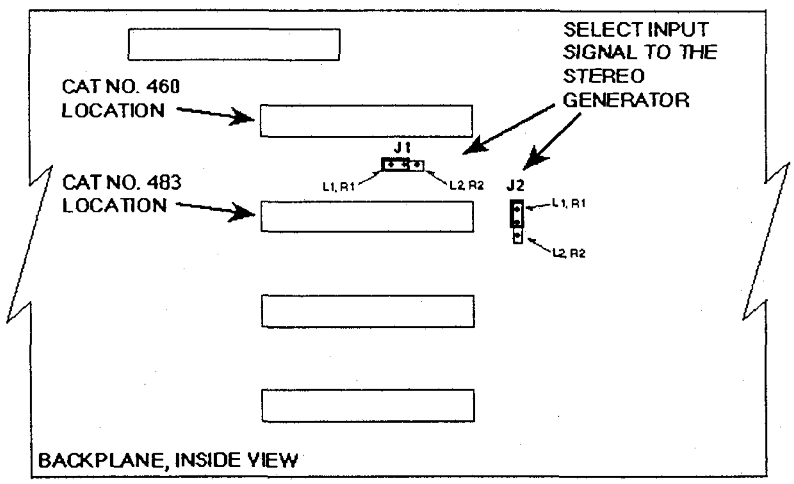
Receiver Jumpers, cont.

This jumper selects the signal which appears at position **6** of terminal strip **TB102** located on the rear panel of the receiver. With the jumper in the **MTR** position, a voltage proportional to the front panel meter reading appears on the terminal. This test signal is active for all positions of the front panel meter selector switch including the **Error Rate** switch position.

With the jumper in the **BER** position, the voltage on pin **6** of the terminal strip is proportional to **Error Rate** exclusively, independent of the position of the front panel meter selector switch.

NOTE Other jumpers not specifically mentioned are for factory use only and should not be disturbed.

Backplane Jumpers [DP5504 Only]



J1,2 Composite Source: [L1,R1]/L2,R2

Note To gain access to these jumpers it is necessary to remove the Cat.No. 460 and Cat.No. 483 modules.

These jumpers are used to select the program audio source (L1,R1 or L2,R2) for the optional Cat.No. 460 Digital Stereo Generator. With appropriate cabling at the time of installation, the default channel assignment (L1,R1) should be adequate for most applications. For systems that require an L2,R2 composite source (e.g., multi-hop), simply place both of these jumpers in the **L2,R2** position.

Note Other jumpers in the receiver not specifically mentioned are for factory use only and should not be disturbed.

3.5 Re-assembling DSTL Transmitter and Receiver

After all necessary jumpers have been set, return the modules to their respective slots, re-attach the retaining bracket for the Alarm/Control and Power Supply modules, replace the ribbon cable connecting the meter sub-assembly, and replace the sub-panel.

On the DSTL transmitter, replace the RF jumper between the Transmit Frequency Synthesizer and Transmit Power Amplifier.

On both transmitter and receiver, be sure to replace the nut securing the RF "N" connector to the rear panel. Also be sure to replace the screws affixing the power mains socket to the rear panel.

3.6 Programming Carrier Frequency

Your DSTL units may have arrived with the operating frequency pre-set by Broadcast Electronics. However, it is wise to reconfirm the switch settings on the Transmit Frequency Synthesizer module of the DSTL transmitter and the Receiver module of the DSTL receiver.

The operating frequency is programmed by the rotary hex switches S1, S2 and S3. The codes are identical for both transmitter and receiver.

Confirm that the switch settings correspond to the desired operating frequency according to the table in Appendix A. Upon power-up, the transmitter and receiver will operate at the frequency selected. If the frequency has been changed while the units are powered-up, the **Reset** button on each of these modules must be pressed before the newly-programmed operating frequency takes effect.

For your convenience, each sub-panel includes a write-in area for you to note your operating frequency and the corresponding settings for switches S1, S2, and S3.

Note The switches on the transmitter and receiver are mounted differently (they are rotated 180 degrees between the two modules). Do not judge switch position by physical orientation of the arrowed indicator with respect to the chassis.

3.7 Bench Testing / Checkout (Optional)

If you wish to perform preliminary testing on the bench prior to installation, you will need either a dummy load, or preferably, about 100 dB of attenuators connecting the RF output of the DSTL transmitter to the RF input of the DSTL receiver.

If only a dummy load is available, place the DSTL transmitter and receiver in close proximity and attach the load to the transmitter. Insert a length of hookup wire into the hot side of the receiver RF "N" connector and position the other end in the vicinity of the dummy load.

If neither attenuators or a dummy load are available, you can run the transmitter unterminated. Use hookup wire for a receiver antenna, as in the previous setup. When the DSTL transmitter is in **OPERATE**, you will get a red **STATUS** indication on the Transmit Power Amplifier module due to high VSWR and also a **SUMMARY FAULT** indication. You will, however, be able to verify all other functions.

3.7.1 Test 1—System Integrity

Place both units in the **OPERATE** mode. Place the selector switch on the transmitter to the **FORWARD PWR** position and place the switch on the receiver to the **RF RCV LEVEL** position. You should see an indication of about 2 Watts on the transmitter and a mid-scale indication on the receiver of the RF signal being received. No LED should be red (unless you are testing the system with the transmitter unterminated. See above).

3.7.2 Test 2—Audio Signal Integrity

Place the meter selector on the transmitter and on the receiver to the **MAX (L,R)** position. Attach an audio oscillator output signal to either the Left/L1 or Right/R1 channel input barrier strips on the transmitter. Adjust the oscillator frequency to 1 kHz or other convenient frequency and the output level to about 0 dBu (0.775v). Verify that the meter reading on the transmitter and the receiver are approximately equal. Vary the oscillator output level, and check that the meters track accordingly.

Note After these tests, return the units to the **STANDBY** position.

4.1 Rack Mounting

Each Broadcast Electronics DSTL transmitter and receiver requires a standard 19" (482.6mm) EIA rack width and 7" (177.8mm), 4 rack units, of height. The depth of each unit is 17" (431.8mm). Certain installations may require consideration for this depth. To accommodate RF connections, for example, right angle RF "N" connectors may be required. While rated for continuous operation at elevated temperatures, an extra rack unit of space is recommended for ventilation above and/or below each unit wherever practical.

Refer to the rear panel fold-outs accompanying this section for connector identification.

4.2 Transmitter Connections

4.2.1 Audio / Modem

Audio and modem inputs are made to conventional barrier strip terminals, **TB110** and **TB111** at the rear of the transmitter. Be sure to observe correct polarity in order to maintain correct phase between the left and right channels.

When connecting the DSTL transmitter inputs to equipment with **balanced** outputs, observe standard conventions: The "high" output should connect to the "+" input of the DSTL transmitter. The "low" output connects to the "–" input, and the shield of the cable connects to either the chassis ground position marked "**G**", or the chassis of the source equipment, **but not both**.

If connecting the DSTL transmitter to **unbalanced** equipment, we suggest that you still use 2-conductor shielded cable and use the following connection scheme: Connect the high lead to the "+" input, and the low lead to the "–" input. Connect the shield to the low lead **at the source equipment only** to avoid ground loops.

4.2.2 Data (DP5503A Only)

Data channel input connections are made to 9 pin D connectors **J119** (Data 1) and **J120** (Data 2) on the rear panel of the transmitter. These connectors follow the interface specifications of EIA standard RS-232, and pinouts conform to industry convention (see block diagram, Section 10). Under RS-232, the DP5503A is categorized as a DCE device, and is typically connected to data terminal equipment (DTE). Refer to Section 3.4.1 for data channel baud rate and parity settings.

In order for data transmission to be enabled, input signal DTR (pin 4) must be held in the "ON", or high level state. A low level or open DTR signal at either connector will disable the corresponding data channel, and place output signal DCD in the "OFF" state at the receiver data connector.

4.2.3 RF

The RF output is a conventional "N" female connector. Be sure to use a short length of male-to-male coax "pigtail" as a strain relief between the output of the transmitter and the transmission line.

4.2.4 Status Relays

Numerous status relay closures are provided on **TB101** and **TB102**. Note that for maximum flexibility, form C relay contacts are provided. Use either the normally closed or normally open contacts depending on your remote control system requirements. A table of the relaxed and energized conditions for the various relays follows:

Relay		Normally Open (**NO**)	Normally Closed (**NC**)
ALARM	T	open	closed
	F	closed	open
OPERATE	T	closed	open
	F	open	closed
IN REMOTE	T	closed	open
	F	open	closed

Relay Definitions:

ALARM

A fault has been detected in the transmitter that has caused the SUMMARY ALARM LED to light, as well as activating this relay closure. See Section 9 for troubleshooting information.

OPERATE

The unit has been placed into operational mode, either by the front panel **MODE** switch, or under Hot Standby control, if connected.

If a serious fault condition occurs, the unit will switch out of **OPERATE** and into **STANDBY**.

IN REMOTE

The front panel **MODE** switch has been placed in the **REMOTE** position to enable control by a Hot Standby unit or to enable operation by remote control.

4.2.5 Meter Reading Output "MTR"

Note Applies only to units with serial numbers 21 and above.

An analog voltage proportional to the front panel meter selection is available on terminal **6** of **TB102**, per the following table:

Switch Position	Voltage (volts)	
MAX (L,R), LEFT, RIGHT Audio Levels	0dB = 3.97 volts (DC)	
	Watts	Voltage (DC) ± 10%
FORWARD, REVERSE PWR	0	0
	0.25	1.8
	0.4	2.35
	0.6	3.0
	1.0	3.85
	1.5	4.55
	2.0	5.0

Connect terminal 6 to your telemetry system if desired. Terminal 5 is ground.

Caution! If you use this output to monitor the selection chosen by the Meter Selector, be sure to instruct your staff to leave the switch undisturbed.

4.2.6 Hot Standby Remote Control Connections

If you are making hot standby connections, use the 15 pin D connector **J109** on the rear panel to connect the DSTL transmitter with the Hot Standby unit (the cable is supplied with the Hot Standby unit). Place the **MODE** switch in **REMOTE** to enable control by the Hot Standby unit.

The hot standby connector **J109** can also be used to enable the DSTL transmitter via remote control. With the **MODE** switch in **REMOTE**, a latching closure between pins **1** and **2** of **J109** will switch the transmitter into **OPERATE**.

4.2.7 Repeater Connections

If the DSTL transmitter is being used for digital repeater operation, use the 9 pin D connector **J107** on the back of the unit to connect the transmitter to the receiver. The audio inputs are bypassed in favor of the digital input from the receiver via this connector. Connector **J109** must also be linked to the corresponding connector on the receiver via a special 15 pin D cable, and the transmitter mode switch should be placed in the **Remote** position. Remember that you need to reconfigure jumpers to enable repeater operation (see Section 3.4).

Note See Appendix C for additional information and repeater cable construction.

4.3 Receiver Connections

4.3.1 RF

The RF input is a conventional "N" female connector. Be sure to use a short length of male-to-male coax "pigtail" as a strain relief.

4.3.2 Data (DP5504 Only)

Data channel output connections are made to 9 pin D connectors **J119** (Data 1) and **J120** (Data 2) on the rear panel of the receiver. These connectors follow the interface specifications of EIA standard RS-232, and pinouts conform to industry convention (see block diagram, Section 10). Under RS-232, the DP5504 is categorized as a DCE device, and is typically connected to data terminal equipment (DTE). Refer to Section 3.4.1 for data channel baud rate and parity settings.

Output signal DCD (pin 1) indicates the status of the corresponding data channel. A high level state ("ON") at this pin indicates that the channel is active and able to receive valid data. A low level DCD signal indicates that signal DTR is "OFF" at the transmitter (see Section 4.2.2), or that the receiver has lost synchronization with the transmitter. In either case, data is not available at the corresponding output pin.

4.3.3 Audio / Modem

Connect the outputs at **TB110** and **TB111** to the next stage of equipment using the barrier strip terminals provided. For **balanced** installations, the "+" connects to the "high" lead, the "–" to the "low" lead and the "**G**" to the shield. Do not connect the shield at the other end.

If you are connecting **unbalanced** equipment to the DSTL receiver, we suggest that you still use 2-conductor shielded cable and use the following connection scheme: Connect the high lead to the "+" output, and the low lead to the "–" output. Connect the shield to the low lead **at the unbalanced equipment only** to avoid ground loops.

4.3.4 Digital Stereo Generator

If the Digital Stereo Generator, Cat. No. 460, is installed, connect its output to the FM transmitter exciter using the BNC output connector **J108** on the rear panel.

Mono/stereo switching of the digital stereo generator can be remotely controlled. A latching closure connected between pins **4** and **5** on **TB102** causes the stereo generator to switch to mono operation. To enable remote control, be sure to switch the **MODE** switch on the Digital Stereo Generator to **Remote**.

Note (DP5504 only) The program audio source is factory set to the L1,R1 channels. See page 3-17 for backplane jumper locations to select the L2,R2 channels as inputs to the digital stereo generator.

4.3.5 Status Relays

Numerous status relay closures are provided on **TB101**, **TB102**, and **TB103**. Note that for maximum flexibility, form C relay contacts are provided. Use either the normally closed or normally open contacts depending on your remote control system requirements. A table of the relaxed and energized conditions for the various relays follows:

Relay		Normally Open (NO)	Normally Closed (NC)
Alarm	T	open	closed
	F	closed	open
Operate	T	closed	open
	F	open	closed
In Remote	T	closed	open
	F	open	closed
Audio Mute	T	open	closed
	F	closed	open
Aux Mute	T	open	closed
	F	closed	open

Relay Definitions:

ALARM - A fault has been detected in the receiver (or a transmitter fault or path problem has occurred) that has caused the SUMMARY ALARM LED to light, as well as activating this relay closure. See Section 9 for troubleshooting information.
OPERATE - The unit has been placed into operational mode, either by the front panel MODE switch, or under Hot Standby control, if present.

If a fault condition occurs (including absence of carrier), the unit will switch out of **OPERATE** and into **STANDBY**.

IN REMOTE - The front panel MODE switch has been placed in the REMOTE position to enable control by a Hot Standby unit.

AUDIO MUTE - Excessive bit errors have caused the main audio channels (Left and Right in the DP5502, Left 1 and Right 1 in the DP5504) to mute. If the Digital Stereo Generator is installed, the composite output will also be muted in the DP5502, and in the DP5504 (if Left 1 and Right 1 are the source of the composite signal). This condition will also cause an ALARM relay closure.

AUX MUTE - Excessive bit errors have caused the auxiliary audio channels to mute (Aux channel in the DP5502, Left 2 and Right 2 in the DP5504). If the Digital Stereo Generator is installed in the DP5504, the composite output will also be muted if Left 2 and Right 2 are the source of the composite signal. This condition will also cause an ALARM relay closure. (Note: Without a Hot Standby unit, the main audio channels in the DP5502 will continue to operate. With a Hot Standby unit, a changeover will occur.)

4.3.6 Meter Reading Output "BER/MTR"

An analog voltage proportional to either the **ERROR RATE** or any of the other selections on the front panel **Meter Selector** switch is available on terminal **6** of **TB102** (Terminal 5 is ground), per the following table. A jumper on the Alarm module, Cat. No. 458, selects the error rate output or the meter selection. See Section 3.4 for jumper location.

Switch Position	Voltage (Volts)	
MAX (L,R), LEFT, RIGHT Audio Levels	0dB = 3.97 volts (DC)	
	(μV)	Voltage (DC)
RF RCV LEVEL	1	0.35
	2	0.9
	3	1.25
	10	2.2
	30	2.8
	100	3.4
	300	3.9
	1000	4.4
	3000	5.0
ERROR RATE	10^{-9}	0.0
	10^{-7}	0.4
	10^{-6}	0.75
	10^{-5}	1.5
	10^{-4}	2.75
	10^{-3}	5.0

Caution! If you use the output to monitor the selection chosen by the Meter Selector, be sure to instruct your staff to leave the switch undisturbed.

4.3.7 Hot Standby / Remote Control Connections

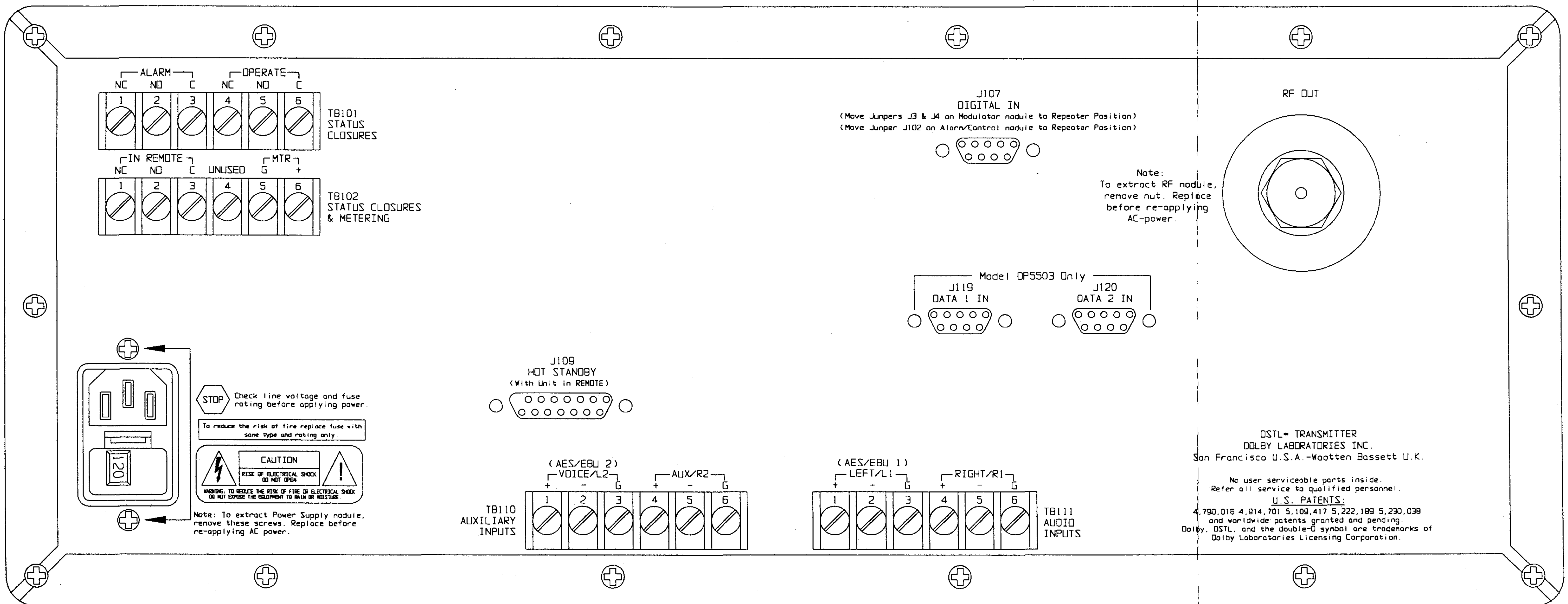
If making hot standby connections, use the rear panel 15 pin D connector **J109** to connect the DSTL receiver with the Hot Standby unit (the cable is supplied with the Hot Standby unit). Place the **MODE** switch in **REMOTE** to enable control by the Hot Standby unit. In addition, this connector can also be used to enable the DSTL receiver via remote control. With the **MODE** switch in **REMOTE**, a latching closure between pins **1** and **2** of **J109** will switch the receiver into **OPERATE**. This connector can also be used to connect to another DSTL receiver or an analog STL receiver in a master/slave configuration. Refer to the instructions provided in Appendix C.

4.3.8 Repeater Use

If the DSTL receiver is being used as a digital repeater, use the rear panel 9 pin D connector **J107** to connect the DSTL receiver to a DSTL transmitter. Connector **J109** must also be linked to the corresponding connector on the transmitter via a special 15 pin D cable, and the transmitter mode switch should be placed in the **Remote** position. No jumpers need to be moved on the receiver if the Audio Decoder Module, Cat. No. 463/483, is in place. Otherwise, a jumper on the Alarm/Control Module, Cat. No. 458, needs to be moved. (Refer to Section 3.4.)

If the D/A Converter Module (Cat. No. 462/482) is in place, all audio outputs are available at the intermediate site (including the composite signal, if the Cat. No. 460 Digital Stereo Generator is also installed). If the Cat.No.483 Audio Decoder Module is in place (DP5504), data outputs are also available at the intermediate site.

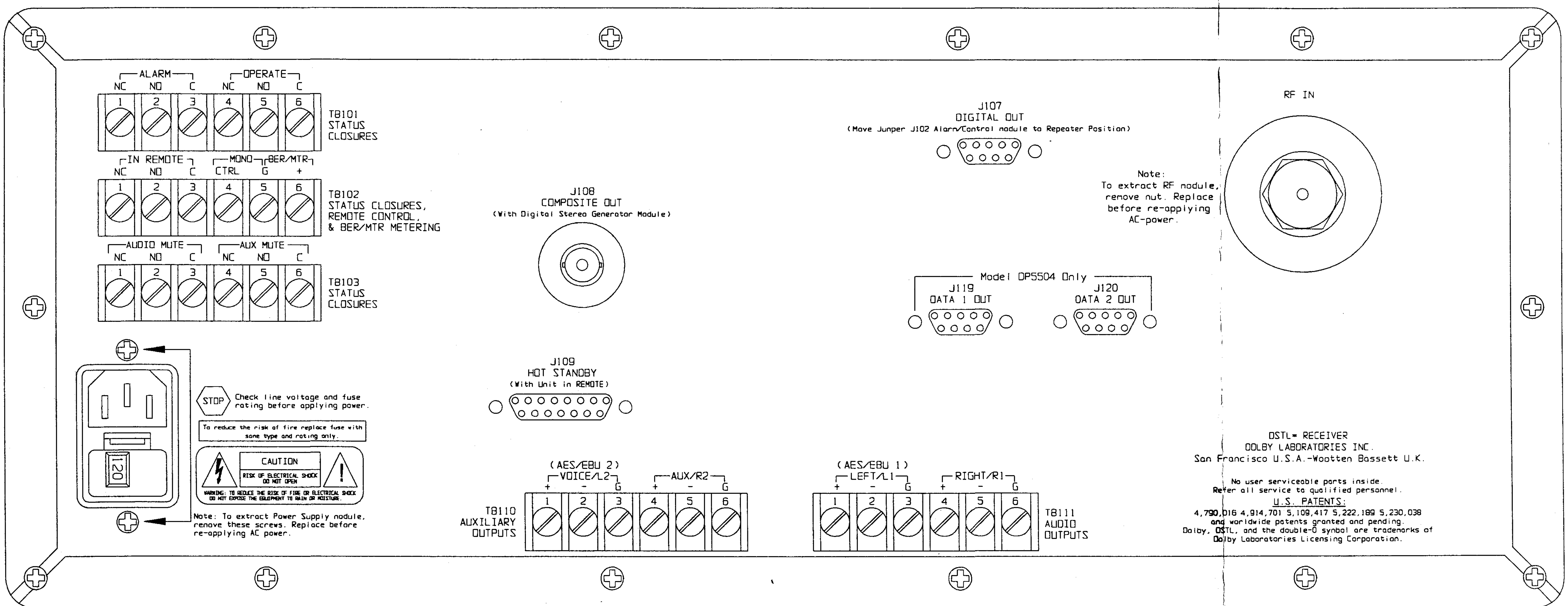
Note See Appendix C for additional information on repeater operation, repeater cable construction, and frequency spacing.



597-8100-11

MODEL DP5501A/DP5503A TRANSMITTER REAR PANEL

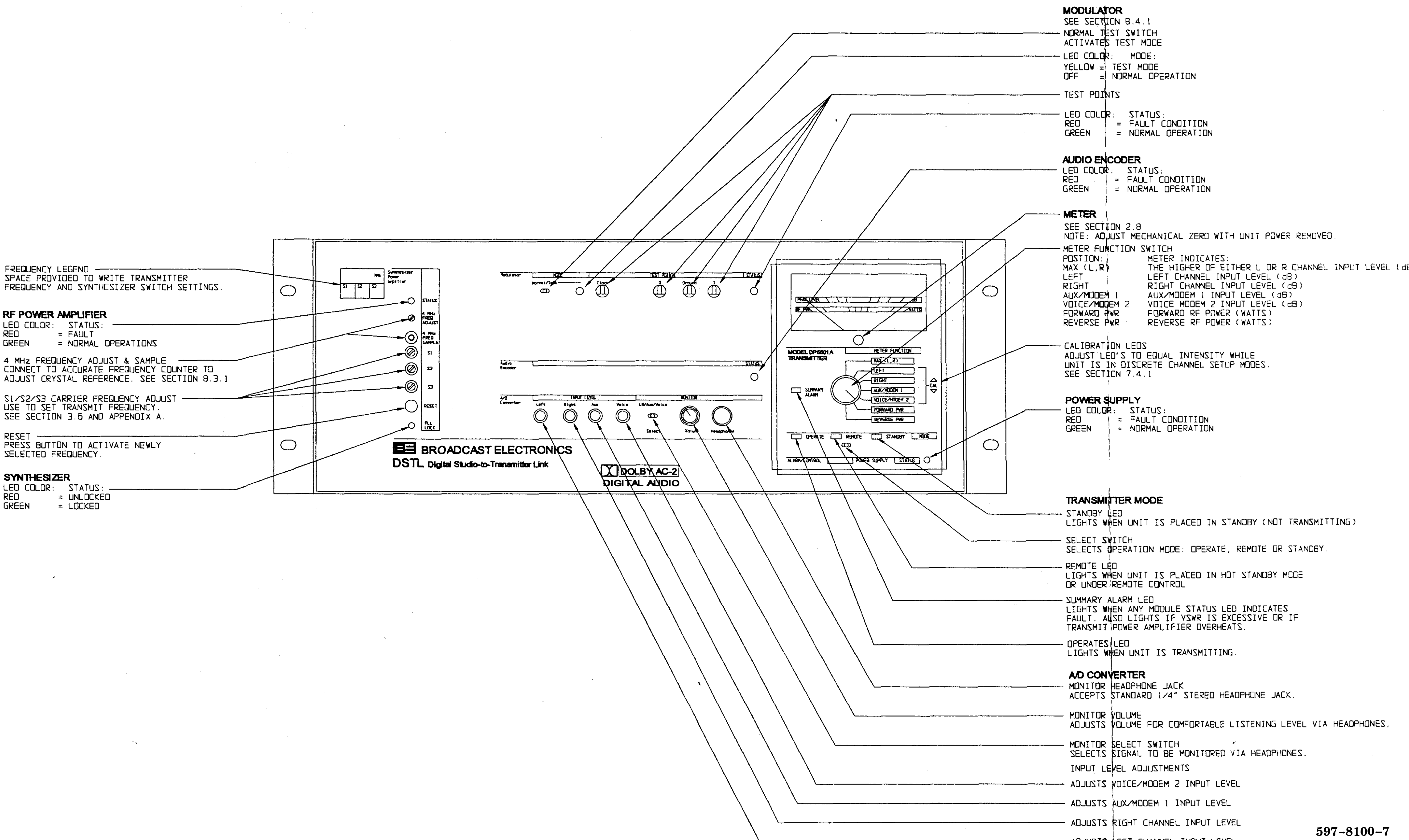
(4-9/4-10)



597-8100-12

MODEL DP5502/DP5504 RECEIVER REAR PANEL

(4-11/4-12)



597-8100-7

MODEL DP5501A TRANSMITTER

(5-2/5-3)

DIGITAL STEREO GENERATOR
LED COLOR: STATUS:
RED = FAULT
STEADY GREEN = NORMAL OPERATION
FLASHING GREEN = TEST MODE

OUT
ADJUSTS COMPOSITE OUTPUT LEVEL.

L- R LEVEL FINE
FINE ADJUST FOR L- R LEVEL (SEPARATION).

L- R LEVEL COARSE
COARSE ADJUST FOR L- R LEVEL (SEPARATION).

PILOT INJECTION
ADJUSTS PILOT INJECTION LEVEL.

PILOT OFF/ON
TURNS PILOT OFF OR ON.

R CHAN OFF/NORM/INV
INVERTS PHASE OF RIGHT CHANNEL OR TURNS RIGHT CHANNEL OFF.

TEST OSC 400/OFF/1K
SELECTS FREQUENCY OF TEST OSCILLATOR.

TEST OSC TRIG
CONNECT TO OSCILLOSCOPE FOR TRIGGERING DURING SETUP.

MODE REM/STER/MONO
SELECTS MODE:
REMOTE - STEREO/MONO/ SWITCHING BY REMOTE CONTROL.
STEREO - STEREO, LOCALLY
MONO - MONO, LOCALLY

FREQUENCY LEGEND
SPACE PROVIDED TO WRITE RECEIVER FREQUENCY AND SYNTHESIZER SWITCH SETTINGS.

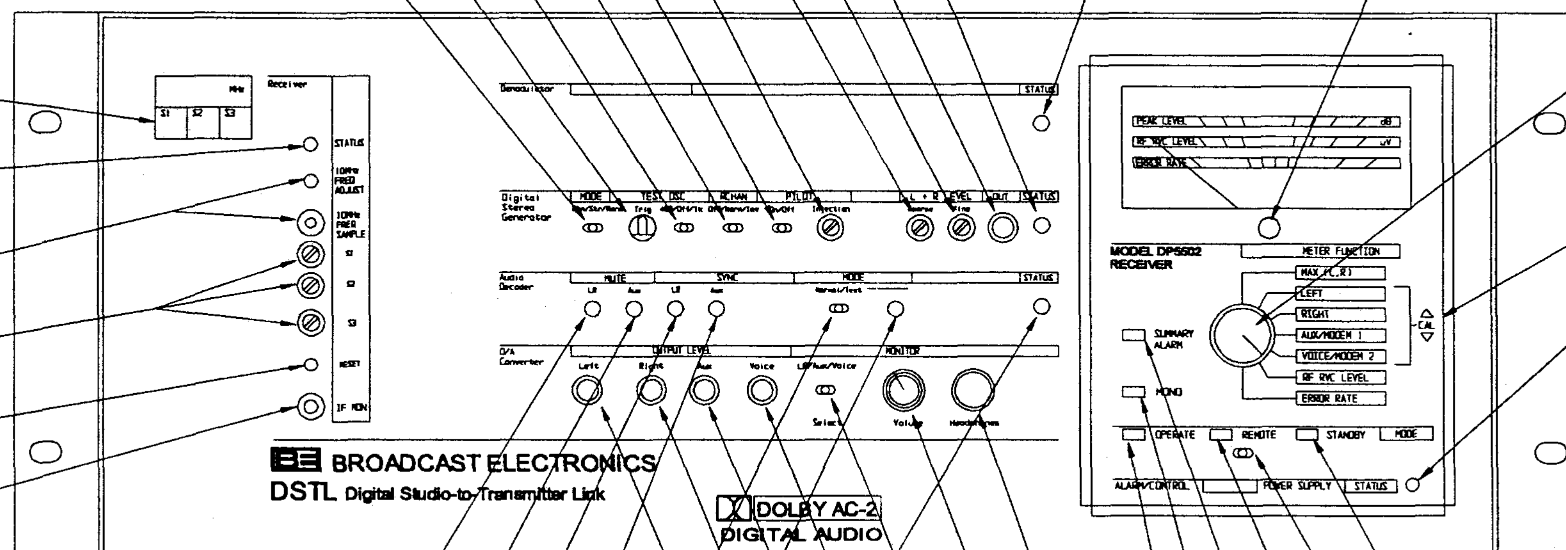
RECEIVER
LED COLOR: STATUS:
RED = FAULT
GREEN = NORMAL OPERATION

10MHz FREQUENCY ADJUST & SAMPLE
CONNECT TO ACCURATE FREQUENCY COUNTER TO ADJUST CRYSTAL REFERENCE. SEE SECTION 8.3.2

S1/S2/S3 RECEIVER FREQUENCY ADJUST
USE TO SET RECEIVER FREQUENCY.
SEE SECTION 3.6 AND APPENDIX A.

RESET
PRESS BUTTON TO ACTIVATE NEWLY SELECTED FREQUENCY.

IF MONITOR
SAMPLE OF 70MHz RF AFTER THE RECEIVER. CAN BE USED TO CHECK FOR CO-CHANNEL INTERFERENCE WHEN THE TRANSMITTER IS OFF.



DEMODULATOR
LED COLOR: STATUS:
RED = FAULT CONDITION
GREEN = NORMAL OPERATION

METER
SEE SECTION 2.8
NOTE: ADJUST MECHANICAL ZERO WITH UNIT POWER REMOVED.

METER FUNCTION SWITCH

POSITION:	METER INDICATES:
MAX (LR)	THE HIGHER OF EITHER L OR R CHANNEL OUTPUT LEVEL (dB)
LEFT	LEFT CHANNEL OUTPUT LEVEL (dB)
RIGHT	RIGHT CHANNEL OUTPUT LEVEL (dB)
AUX/MODEM 1	AUX/MODEM 1 OUTPUT LEVEL (dB)
VOICE/MODEM 2	VOICE/MODEM 2 OUTPUT LEVEL (dB)
RF RCV LEVEL	RF RECEIVED SIGNAL STRENGTH (uV)
ERROR RATE	BIT ERROR RATE

CALIBRATION LEDS
ADJUST LEDS TO EQUAL INTENSITY WHILE UNIT IS IN DISCRETE CHANNEL SETUP MODES.
SEE SECTION 7.4.2

POWER SUPPLY
LED COLOR: STATUS:
RED = FAULT CONDITION
GREEN = NORMAL OPERATION

AUDIO DECODER
SEE SECTION 8.5.1
LED COLOR: STATUS:
YELLOW = LEFT AND RIGHT CHANNELS (AND COMPOSITE) MUTED.
OFF = NORMAL

LED COLOR: STATUS:
YELLOW = AUX/MODEM CHANNEL MUTED.
OFF = NORMAL

LED COLOR: STATUS:
GREEN = NORMAL
OFF = LOSS OF SYNC BETWEEN TRANSMITTER AND RECEIVER AFFECTING LEFT AND RIGHT CHANNELS (AND COMPOSITE.)

LED COLOR: STATUS:
GREEN = NORMAL
OFF = LOSS OF SYNC BETWEEN TRANSMITTER AND RECEIVER AFFECTING AUX/MODEM 1 CHANNEL.

NORMAL/TEST SWITCH
ACTIVATES TEST MODE.

LED COLOR: STATUS:
YELLOW = TEST MODE
OFF = NORMAL

LED COLOR: STATUS:
RED = FAULT CONDITION
GREEN = NORMAL OPERATIONS

RECEIVER MODE

STANDBY LED
LIGHTS WHEN UNIT IS PLACED IN STANDBY (NOT RECEIVING)

SELECT SWITCH
SELECTS OPERATION MODE: OPERATE, REMOTE OR STANDBY.

REMOTE LED
LIGHTS WHEN UNIT IS PLACED IN HOT STANDBY MODE OR UNDER REMOTE CONTROL

SUMMARY ALARM LED
LIGHTS WHEN ANY MODULE STATUS LED INDICATES FAULT.

MONO LED
LIGHTS WHEN STEREO GENERATOR IS SWITCHED TO MONO OPERATION.

OPERATE LED
LIGHTS WHEN MAIN CHANNELS ARE NOT MUTED.

D/A CONVERTER

MONITOR HEADPHONE JACK
ACCEPTS STANDARD 1/4" STEREO HEADPHANES PLUGS.

MONITOR VOLUME
ADJUSTS VOLUME FOR COMFORTABLE LISTENING LEVEL VIA HEADPHONES.

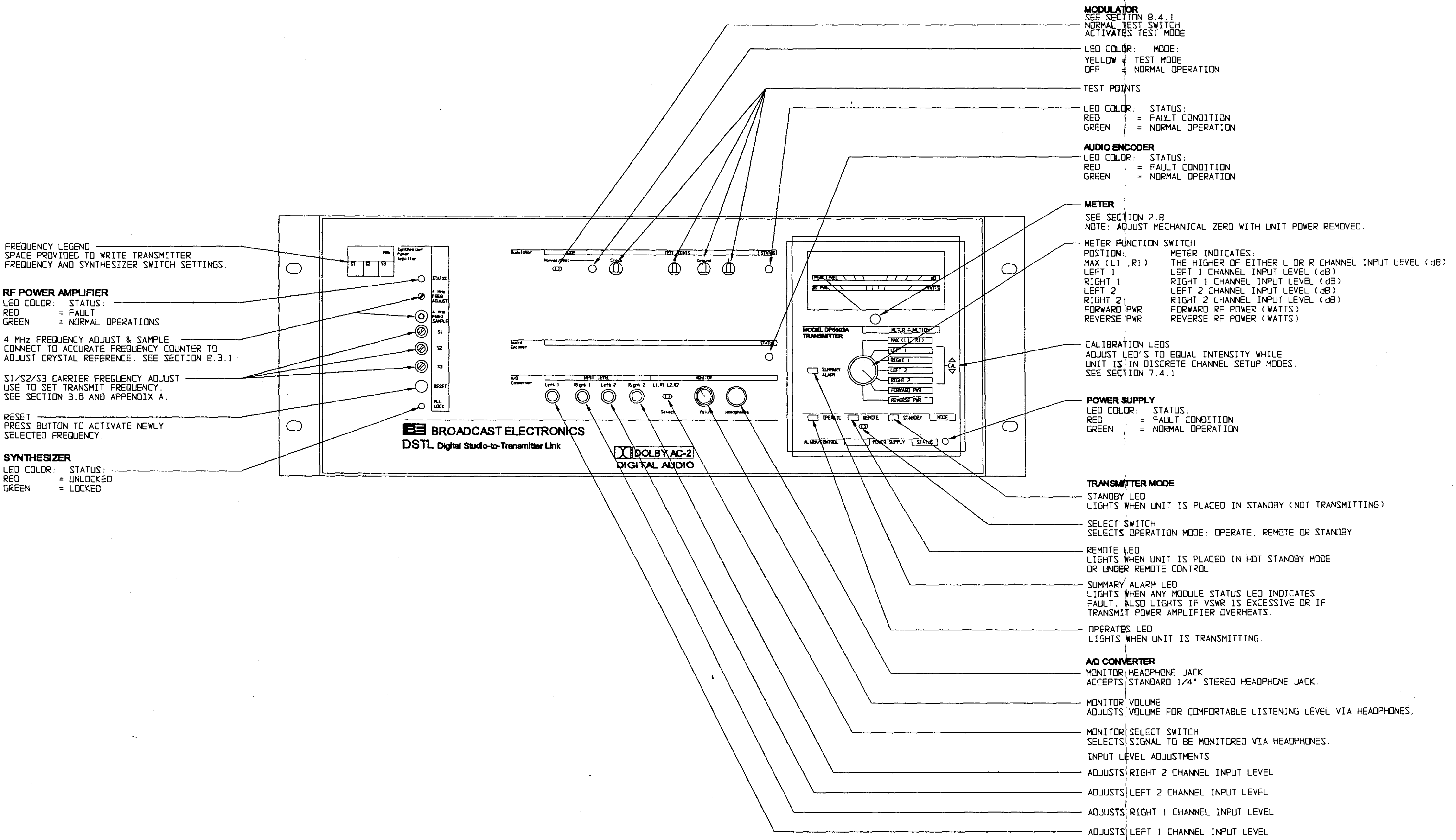
MONITOR SELECT SWITCH
SELECTS SIGNAL TO BE MONITORED VIA HEADPHONES.

OUTPUT LEVEL ADJUSTMENTS
ADJUSTS VOICE/MODEM 2 OUTPUT LEVEL.
ADJUSTS AUX/MODEM 1 OUTPUT LEVEL.
ADJUSTS RIGHT CHANNEL OUTPUT LEVEL.
ADJUSTS LEFT CHANNEL OUTPUT LEVEL.

597-8100-5

MODEL DP5502 RECEIVER

(5-4/5-5)



MODULATOR
SEE SECTION 8.4.1
NORMAL TEST SWITCH
ACTIVATES TEST MODE
LED COLOR: MODE:
YELLOW = TEST MODE
OFF = NORMAL OPERATION
TEST POINTS
LED COLOR: STATUS:
RED = FAULT CONDITION
GREEN = NORMAL OPERATION
AUDIO ENCODER
LED COLOR: STATUS:
RED = FAULT CONDITION
GREEN = NORMAL OPERATION
METER
SEE SECTION 2.8
NOTE: ADJUST MECHANICAL ZERO WITH UNIT POWER REMOVED.
METER FUNCTION SWITCH
POSTION: METER INDICATES:
MAX (L1,R1) THE HIGHER OF EITHER L OR R CHANNEL INPUT LEVEL (dB)
LEFT 1 LEFT 1 CHANNEL INPUT LEVEL (dB)
RIGHT 1 RIGHT 1 CHANNEL INPUT LEVEL (dB)
LEFT 2 LEFT 2 CHANNEL INPUT LEVEL (dB)
RIGHT 2 RIGHT 2 CHANNEL INPUT LEVEL (dB)
FORWARD PWR FORWARD RF POWER (WATTS)
REVERSE PWR REVERSE RF POWER (WATTS)
CALIBRATION LEDS
ADJUST LED'S TO EQUAL INTENSITY WHILE
UNIT IS IN DISCRETE CHANNEL SETUP MODES.
SEE SECTION 7.4.1
POWER SUPPLY
LED COLOR: STATUS:
RED = FAULT CONDITION
GREEN = NORMAL OPERATION
FREQUENCY LEGEND
SPACE PROVIDED TO WRITE TRANSMITTER
FREQUENCY AND SYNTHESIZER SWITCH SETTINGS.
RF POWER AMPLIFIER
LED COLOR: STATUS:
RED = FAULT
GREEN = NORMAL OPERATIONS
4 MHz FREQUENCY ADJUST & SAMPLE
CONNECT TO ACCURATE FREQUENCY COUNTER TO
ADJUST CRYSTAL REFERENCE. SEE SECTION 8.3.1
S1/S2/S3 CARRIER FREQUENCY ADJUST
USE TO SET TRANSMIT FREQUENCY.
SEE SECTION 3.6 AND APPENDIX A.
RESET
PRESS BUTTON TO ACTIVATE NEWLY
SELECTED FREQUENCY.
SYNTHESIZER
LED COLOR: STATUS:
RED = UNLOCKED
GREEN = LOCKED
BROADCAST ELECTRONICS
DSTL Digital Studio-to-Transmitter Link
DOLBY AC-2
DIGITAL AUDIO
MODEL DP5503A
TRANSMITTER
TRANSMITTER MODE
STANDBY LED
LIGHTS WHEN UNIT IS PLACED IN STANDBY (NOT TRANSMITTING)
SELECT SWITCH
SELECTS OPERATION MODE: OPERATE, REMOTE OR STANDBY.
REMOTE LED
LIGHTS WHEN UNIT IS PLACED IN HOT STANDBY MODE
OR UNDER REMOTE CONTROL
SUMMARY ALARM LED
LIGHTS WHEN ANY MODULE STATUS LED INDICATES
FAULT. ALSO LIGHTS IF VSWR IS EXCESSIVE OR IF
TRANSMIT POWER AMPLIFIER OVERHEATS.
OPERATES LED
LIGHTS WHEN UNIT IS TRANSMITTING.
A/D CONVERTER
MONITOR HEADPHONE JACK
ACCEPTS STANDARD 1/4" STEREO HEADPHONE JACK.
MONITOR VOLUME
ADJUSTS VOLUME FOR COMFORTABLE LISTENING LEVEL VIA HEADPHONES.
MONITOR SELECT SWITCH
SELECTS SIGNAL TO BE MONITORED VIA HEADPHONES.
INPUT LEVEL ADJUSTMENTS
ADJUSTS RIGHT 2 CHANNEL INPUT LEVEL
ADJUSTS LEFT 2 CHANNEL INPUT LEVEL
ADJUSTS RIGHT 1 CHANNEL INPUT LEVEL
ADJUSTS LEFT 1 CHANNEL INPUT LEVEL

DIGITAL STEREO GENERATOR
LED COLOR: STATUS:
RED = FAULT
STEADY GREEN = NORMAL OPERATION
FLASHING GREEN = TEST MODE

OUT
ADJUSTS COMPOSITE OUTPUT LEVEL.

L- R LEVEL FINE
FINE ADJUST FOR L- R LEVEL (SEPARATION).

L- R LEVEL COARSE
COARSE ADJUST FOR L- R LEVEL (SEPARATION).

PILOT INJECTION
ADJUSTS PILOT INJECTION LEVEL.

PILOT OFF/ON
TURNS PILOT OFF OR ON.

R CHAN OFF/NORM/INV
INVERTS PHASE OF RIGHT CHANNEL OR TURNS RIGHT CHANNEL OFF.

TEST OSC 400/OFF/1K
SELECTS FREQUENCY OF TEST OSCILLATOR.

TEST OSC TRIG
CONNECT TO OSCILLOSCOPE FOR TRIGGERING DURING SETUP.

MODE REM/STER/MONO
SELECTS MODE:
REMOTE - STEREO/MONO/ SWITCHING BY REMOTE CONTROL.
STEREO - STEREO, LOCALLY
MONO - MONO, LOCALLY

FREQUENCY LEGEND
SPACE PROVIDED TO WRITE RECEIVER FREQUENCY
AND SYNTHESIZER SWITCH SETTINGS.

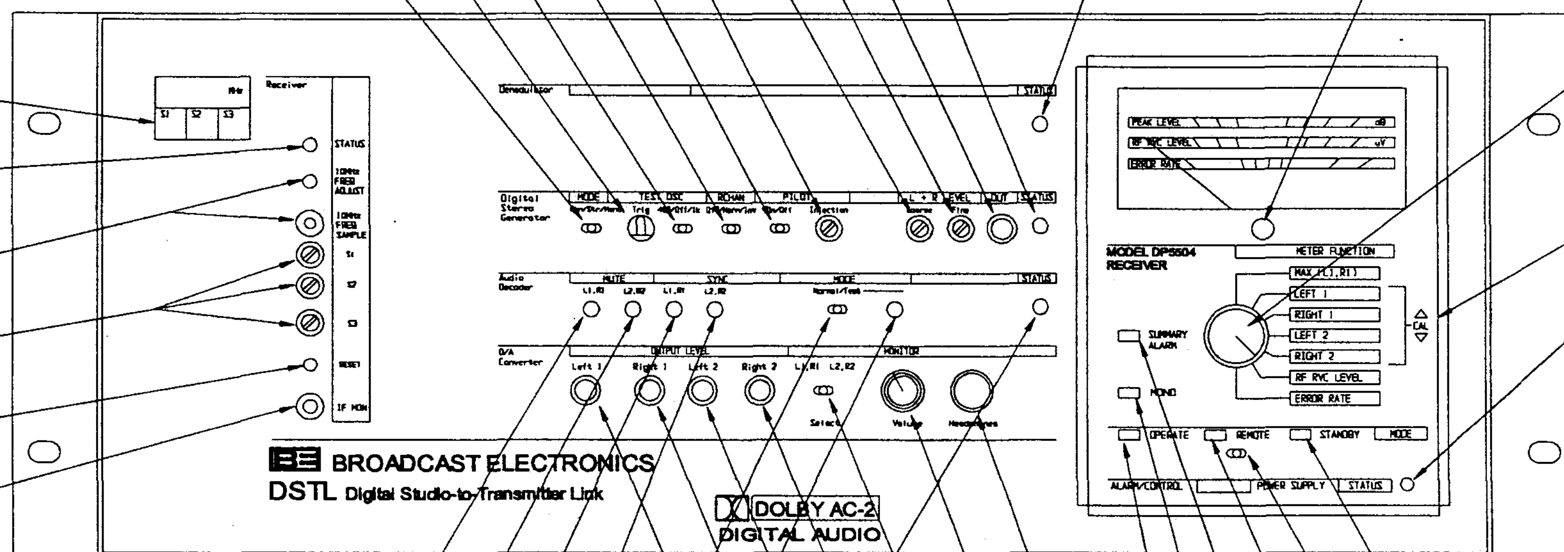
RECEIVER
LED COLOR: STATUS:
RED = FAULT
GREEN = NORMAL OPERATION

10MHz FREQUENCY ADJUST & SAMPLE
CONNECT TO ACCURATE FREQUENCY COUNTER
TO ADJUST CRYSTAL REFERENCE. SEE SECTION 8.3.2

S1/S2/S3 RECEIVER FREQUENCY ADJUST
USE TO SET RECEIVER FREQUENCY.
SEE SECTION 3.6 AND APPENDIX A.

RESET
PRESS BUTTON TO ACTIVATE NEWLY
SELECTED FREQUENCY.

IF MONITOR
SAMPLE OF 70MHz RF AFTER THE RECEIVER. CAN BE USED TO
CHECK FOR CO-CHANNEL INTERFERENCE WHEN THE TRANSMITTER
IS OFF.



AUDIO DECODER
SEE SECTION 8.5.1
LED COLOR: STATUS:
YELLOW = LEFT 1 AND RIGHT 1 CHANNELS MUTED.
OFF = NORMAL

LED COLOR: STATUS:
YELLOW = LEFT 2 AND RIGHT 2 CHANNELS MUTED.
OFF = NORMAL

*LED COLOR: STATUS:
GREEN = NORMAL
OFF = LOSS OF SYNC BETWEEN TRANSMITTER AND RECEIVER
AFFECTING LEFT 1 AND RIGHT 1 CHANNELS.

*LED COLOR: STATUS:
GREEN = NORMAL
OFF = LOSS OF SYNC BETWEEN TRANSMITTER AND RECEIVER
AFFECTING LEFT 2 AND RIGHT 2 CHANNELS.

NORMAL/TEST SWITCH
ACTIVATES TEST MODE.

LED COLOR: STATUS:
YELLOW = TEST MODE
OFF = NORMAL

LED COLOR: STATUS:
RED = FAULT CONDITION
GREEN = NORMAL OPERATIONS

* NOTE: LOSS OF SYNC'S CAUSED BY A HIGHER
BIT ERROR RATE CONDITION THAN THE MUTE LOAD.

DEMODULATOR
LED COLOR: STATUS:
RED = FAULT CONDITION
GREEN = NORMAL OPERATION

METER
SEE SECTION 2.8
NOTE: ADJUST MECHANICAL ZERO WITH UNIT POWER REMOVED.

METER FUNCTION SWITCH

POSITION:	METER INDICATES:
MAX (LR)	THE HIGHER OF EITHER L OR R CHANNEL OUTPUT LEVEL (dB)
LEFT 1	LEFT 1 CHANNEL OUTPUT LEVEL (dB)
RIGHT 1	RIGHT 1 CHANNEL OUTPUT LEVEL (dB)
LEFT 2	LEFT 2 OUTPUT LEVEL (dB)
RIGHT 2	RIGHT 2 OUTPUT LEVEL (dB)
RF RCV LEVEL	RF RECEIVED SIGNAL STRENGTH (uV)
ERROR RATE	BIT ERROR RATE

CALIBRATION LEDS
ADJUST LEDS TO EQUAL INTENSITY WHILE
UNIT IS IN DISCRETE CHANNEL SETUP MODES.
SEE SECTION 7.4.2

POWER SUPPLY
LED COLOR: STATUS:
RED = FAULT CONDITION
GREEN = NORMAL OPERATION

RECEIVER MODE

STANDBY LED
LIGHTS WHEN UNIT IS PLACED IN STANDBY (NOT RECEIVING)

SELECT SWITCH
SELECTS OPERATION MODE: OPERATE, REMOTE OR STANDBY.

REMOTE LED
LIGHTS WHEN UNIT IS PLACED IN HOT STANDBY MODE OR
UNDER REMOTE CONTROL

SUMMARY ALARM LED
LIGHTS WHEN ANY MODULE STATUS LED INDICATES FAULT.

MONO LED
LIGHTS WHEN STEREO GENERATOR IS SWITCHED TO MONO OPERATION.

OPERATE LED
LIGHTS WHEN MAIN CHANNELS ARE NOT MUTED.

D/A CONVERTER

MONITOR HEADPHONE JACK
ACCEPTS STANDARD 1/4" STEREO HEADPHANES PLUGS.

MONITOR VOLUME
ADJUSTS VOLUME FOR COMFORTABLE LISTENING LEVEL VIA HEADPHONES.

MONITOR SELECT SWITCH
SELECTS SIGNAL TO BE MONITORED VIA HEADPHONES.

OUTPUT LEVEL ADJUSTMENTS
ADJUSTS RIGHT 2 CHANNEL OUTPUT LEVEL.
ADJUSTS LEFT 2 CHANNEL OUTPUT LEVEL.
ADJUSTS RIGHT 1 CHANNEL OUTPUT LEVEL.
ADJUSTS LEFT 1 CHANNEL OUTPUT LEVEL.

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MODEL DP5504 RECEIVER

(5-8/5-9)

SECTION 5
OPERATING CONTROLS

Refer to the fold-out pages showing the front sub-panels of the DSTL units.

6.1 Remote/Operate/Standby

Before connecting power to the DSTL units, place the **MODE** switches of both units in the far right **STANDBY** position.

Be sure that the Frequency Synthesizer switch settings **S1**, **S2**, and **S3** on both units have been set to the correct positions (see Section 3.6).

Connect mains power to each unit. Verify that there are no red or yellow LEDs illuminated, and that there are no green LEDs flashing. If you experience any of these conditions, see Section 9 "Troubleshooting."

Place the DSTL units in **REMOTE** if:

A. You are operating under control of a hot standby unit

or

B. You are operating the units under remote control, by means of the Hot Standby connector **J109** on the rear panel.

If under hot standby control, the selected pair of DSTL units will be activated..

If under remote control, continue by enabling the DSTL Transmitter and Receiver with the remote control system.

Otherwise, place the DSTL units in **OPERATE** when you are ready to commence DSTL transmission and reception.

Then proceed to verify STL operation by means of the following sections.

6.2 Adjusting Transmitter Output Power

Under normal circumstances, the DP5501A/5503A Transmitter should be operated at its nominal 2 watt output. Turn the meter selector knob to **FORWARD PWR** to observe the power output. Adjust the **RF LEVEL ADJUST** on the **Transmit Power Amplifier** if necessary, to achieve 2 watt output.

Conditions under which you may want to lower the output include particularly short paths with excess fade margin. Reducing the power minimizes the likelihood of interfering with adjacent channels.

Note If you contemplate lowering the power, keep in mind that lowering the power to 1.0 watt is a reduction of 3 dB; reducing the power to 0.5 watt is a reduction of 6 dB. These reduce the fade margin by a corresponding amount.

6.3 Received Signal Strength

Turn the Meter Selector switch on the DSTL Receiver to **RF RCV LEVEL**. If the received signal strength is lower by more than a few tens of microvolts than the predicted value determined from your path calculations, one or more of the following conditions may be contributing to the discrepancy:

- Misaligned transmit and/or receive antennas;
- Unforeseen terrain obstructions or path considerations;
- Circumstances causing a fade situation;
- Transmission line and/or connectors in need of servicing;
- Additional hardware causing greater losses than figured into the path calculations (additional connectors, items that had been forgotten, such as cavity filters, duplexers, splitters, combiners,dirty or aging connectors, etc.);
- An error in the path calculations.

Where possible, you should take corrective action, particularly if the fade margin you have designed into the path is relatively low for the path length.

6.4 Error Rate

Turn the DSTL receiver's Meter Selector switch to **ERROR RATE**. Under normal circumstances, the error rate displayed on the meter should read about 10^{-9}.

Periodically monitor the error rate, either by directly observing the meter or making use of the rear panel analog voltage output, to monitor path performance. If the reading fluctuates or the error rate creeps upward, signal fade and/or interference may be plaguing your path. See Section 9.2 for details.

7.1 Jumper/Switch Reminder

Please refer to Section 3.4 to ensure that the **Pre-emphasis** and **Signal Adaptation** jumpers/switches have been configured to suit your installation requirements. Briefly:

Pre-emphasis (on the **A/D Converter** module should be applied if no other stage in the signal chain has provision for applying 75 μs (or 50 μs) pre-emphasis.

Signal Adaptation (on the **Audio Encoder** module and **Audio Decoder** module) is required in installations where pre-emphasis precedes the DSTL transmitter, or is applied by the **Pre-emphasis** jumper/switch mentioned above.

Note Refer to Section 2 for a discussion of operating considerations.

7.2 Operating Levels

The Operating Level for unprocessed or partially processed signals has been defined as -12 dB relative to 0 dB on the meter. The 0 dB level on the meter represents the maximum instantaneous audio level that produces proper operation of the DSTL unit.

Alignment in Discrete Channel Operation (Section 7.4 below) utilizes this 0 dB Operating Level to allow 12 dB of headroom before onset of overload.

Alignment in "Virtual Composite" mode (Section 7.3 below) takes advantage of the knowledge that the audio has been fully processed, and matches the maximum output of the audio processor to the 0 dB level of the DSTL system. No headroom is needed.

For optimum sound quality, the input level should never, or only rarely, exceed 0 dB on the input level meter. The safety limiter, active on signals exceeding 0 dB, is just that; it is not intended to be used as additional audio processing. The 90 dB dynamic range of the DSTL eliminates the need to push signal levels above zero, and the meter is an accurate indicator of the actual maximum operating level of the system.

7.3 Setup 1: "Virtual Composite" Operation

Note For this setup, the **CAL** indicators next to the METER FUNCTION labels are not used. References to left and right channels in this section apply to both program pairs in the DP5503A.

7.3.1 At the Studio

1. Select program material (left and right channels) that is the loudest typically played.

2. Set audio processor controls to the highest degree of processing typically used.

3. The following step not only maximizes S/N ratio through the DSTL system, but also activates the overshoot limiter when needed, and matches the output of the digital stereo generator in the DSTL receiver to 100% modulation:

   Feed this "loud" audio into the processor, and in turn, to the DSTL transmitter. Monitor level with DSTL **METER FUNCTION** switch set at **LEFT**. Adjust DSTL's **Left** channel **INPUT LEVEL** so that peak meter indication approaches but never goes over 0 dB on the meter.

The following steps balance left and right channel levels:

4. Replace "loud" program material with 400 Hz tone from the console and feed both channels of the audio processor.

5. Note the **Left** channel input level on DSTL Transmitter meter. Move **METER FUNCTION** switch to **RIGHT** and adjust **Right** channel **INPUT LEVEL** to match left channel level.

6. Replace tone with program material. Return processor to the usual operating settings.

Note If you do not have a modulation monitor at the transmitter site, you will need someone at the studio to read the monitor indications when completing stereo generator alignment at the transmitter site, as described in the section below.

7.3.2 At the Transmitter Site / Stereo Generator Alignment

100% Modulation & Pilot Injection

1. Set the DSTL **MODE** switch to **OPERATE.**

2. On the **Digital Stereo Generator** module, flip the **TEST OSC** switch to **400 Hz** (**STATUS** light blinks to indicate non-standard operational mode). Using the modulation monitor with its mode switched to monitor left and right channels, adjust the module's composite **OUT**put level for a reading of 100% modulation at the monitor. Return the **TEST OSC** switch to **Off**.

3. Monitor Pilot with the modulation monitor and adjust the **PILOT Injection** control to the desired operating level (typically 8-10 %).

4. Upon activation of the DSTL system (see Section 6, Operation), you may need to adjust the composite **OUT**put level slightly to set modulation levels in accordance with your operating practices. Since changing the composite output level also changes the pilot level, be sure to also re-adjust the **PILOT Injection** control.

Notes When monitoring the headphone **MONITOR** outputs of either DSTL transmitter or receiver, you will hear the pre-emphasized input signal.

The meter of both the DSTL transmitter and receiver display the pre-emphasized signal. Slight discrepancy between transmitter and receiver meter deflections are normal, due to very slight overshoots through the system.

Some of the above procedures use tones at 100% modulation. Be aware that you will be broadcasting these tones if you activate your transmitter carrier during these setup procedures.

Separation

The **Digital Stereo Generator**, Cat No. 460, has been optimized for channel separation before leaving the factory. However, due to the uniqueness of each station's setup, it may be necessary to trim the **L-R LEVEL** for best separation according to the following steps:

1. Configure your modulation monitor to read channel separation (residual on right channel with left channel driven).

2. On the **Digital Stereo Generator** module, turn the **TEST OSC** switch to **400 Hz** and the **RIGHT CHAN** switch to **Off**.

3. Note the current position of the **Coarse** and **Fine** switches of **L-R LEVEL** in case you stray too far from reasonable separation performance. Then, alternately adjust the **Coarse** and **Fine** switches until best separation is achieved.

4. Return the **TEST OSC** switch to **OFF** and the **RIGHT CHAN** switch to **NORM** after completing the adjustments.

Note The above procedures use tones at 100% modulation. Be aware that you will be broadcasting these tones if you activate your transmitter carrier during these setup procedures.

7.4 Setup 2: Discrete Channel Operation

Note References to left and right channels in this section apply to both program pairs in the DP5503A.

7.4.1 At the Studio

1. Send 400 Hz tone from the console at nominal 0 dB level and feed both channels of the studio's program processor (if it feeds the DSTL transmitter). Otherwise feed the tone to the DSTL transmitter inputs directly.

The following steps calibrate input levels to the DSTL system for 12 dB of headroom between nominal 0 dB operating level and the maximum input level to the DSTL system (threshold of safety limiter action):

2. Select the **LEFT** channel on the **METER FUNCTION** switch. Adjust the **Left** input control until the two LEDs flanking the **CAL** indicator (next to the **METER FUNCTION** labels) show equal brightness.

3. Move the **METER FUNCTION** switch to **RIGHT** and adjust **Right** channel input control, again for equal brightness of LEDs.

4. If a different amount of headroom is required, offset the reference input signal by the degree that you want to increase or decrease the headroom. For example, for 9 dB of headroom, supply a -3 dB reference tone, and adjust input levels for equal brightness of the LEDs. For 15 dB of headroom, supply a +3 dB reference tone.

5. Replace tone with program material.

7.4.2 At the Transmitter Site

The following steps calibrate the equipment following the DSTL system (probably an audio processor/stereo generator) to nominal operating levels based on the 12 dB headroom level established at the studio:

1. On the **Audio Decoder** Module, place the **MODE** toggle switch to **Test.** This feeds a -12 dB test tone to the outputs.

2. Using either the **OUTPUT LEVEL** controls on the DSTL receiver and/or the input level controls of the equipment following the DSTL receiver, calibrate the equipment to accept the test tone as its nominal 0 dB operating point.

3. If a different degree of headroom is required, offset the calibration to the test signal by the degree that you want to increase or decrease the headroom. For example, for 9 dB of headroom, adjust the gain for a -3 dB measurement; for 15 dB of headroom, adjust the gain for a +3 dB measurement.

4. Proceed with remainder of setup using instructions provided by the manufacturers of the processing and stereo generator equipment.

Note The meter of the DSTL receiver displays the incoming levels of the DSTL *transmitter* and does not indicate the levels adjusted by the receiver's **OUTPUT LEVEL** controls. Use the metering capabilities of the equipment *following* the DSTL receiver for setting levels.

7.5 Aux Channel Calibration (DP5501A/5502 Only)

Audio Signals

The Aux channel calibration procedure when used for an audio signal is identical to the procedure outlined in 7.4 above, with the exception of substituting **Aux** channel meter positions and input and output trimmers on the DSTL transmitter and receiver.

Modem Signals

When used for modem signals, which are constant in amplitude, adjust the **Aux** channel input control on the DSTL transmitter for a display on the **AUX/MODEM 1** position of the **METER FUNCTION** switch for a display of between –6 dB and –1 dB.

Adjust the **Aux** channel output control on the DSTL receiver to produce sufficient output signal for proper operation of the equipment accepting the modem signal. Alternatively, by using a voltmeter or other appropriate test equipment, you can compare the level of the modem output at the studio, and adjust the **Aux** channel output of the DSTL receiver to an identical reading.

7.6 Voice Channel Calibration (DP5501A/5502 Only)

Voice Signals

The Voice channel calibration procedure when used for an audio signal is similar to the procedure outlined in 7.4 above, with one exception being the substitution of the **Voice** channel meter positions and input and output trimmers on the DSTL transmitter and receiver.

Setup at the studio is identical. However, at the transmitter site, the **MODE** toggle switch does not produce a –12 dB calibration signal at the Voice channel output. Therefore, the output level may be set as convenient in non-critical applications, or by the use of a voltmeter or other appropriate test equipment for more critical applications: Compare the level of the Voice output source at the studio, and adjust the **Voice** channel output of the DSTL receiver to an identical reading.

FSK Signals

When used for FSK signals, which are constant in amplitude, adjust the **Voice** channel input control on the DSTL transmitter for a display on the **VOICE/MODEM 2** position of the **METER FUNCTION** switch for a display of between –6 dB and –1 dB.

Adjust the **Voice** channel output control on the DSTL receiver to produce sufficient output signal for proper operation of the equipment accepting the FSK signal. Alternatively, by using a voltmeter or other appropriate test equipment, you can compare the level of the FSK output source at the studio, and adjust the **Voice** channel output control of the DSTL receiver to an identical reading.

8.1 Maintenance

The DSTL system is inherently stable and does not require routine maintenance nor calibration.

Adequate ventilation and protection from environmental hazards (excessive dust and humidity) will go far towards ensuring long and trouble-free service. See Section 4 for proper guidelines for installation.

8.2 Cleaning

The front panel, sub-panel, and chassis can be cleaned periodically with a soft, slightly damp cloth. Mild household cleaners can be employed to remove smudges if necessary. Avoid the use of abrasive cleaners, however. Be especially careful when cleaning the meter face.

Dust accumulation on the chassis may hinder proper ventilation. A thorough vacuuming is suggested as conditions warrant.

8.3 Alignment

The Broadcast Electronics DSTL system does **not** require routine alignment.

Nevertheless, for completeness, provision is made for adjusting the crystal reference frequency for the synthesizers used in the DSTL transmitter and receiver. For verification or adjustment of these reference frequencies, a frequency counter of an accuracy better than or equal to 1 Hz display resolution with a time base drift of better than 1×10^{-7}/month and with a meter calibration within a 6 month period is **required**. Do **not** attempt to perform adjustments if you do not have equipment of this accuracy.

The Digital Stereo Generator, Cat No. 460, does not require re-alignment once it has been properly set up. For completeness, initial alignment instructions described in Section 7 are repeated below and are followed by instructions for performing crosstalk measurements for Proof-of-Performance.

8.3.1 Transmitter

Cat. No. 476 Frequency Synthesizer/RF Amplifier – 4 MHz Frequency Adjust

The 4 **MHz FREQ ADJUST** sets the reference for the PLL master oscillator frequency synthesizer. The adjustment can be made from the front panel.

Prior to any frequency adjustment, power should be applied to the transmitter and the frequency counter for at least 30 minutes.

Connect the frequency counter to the 4 **MHz FREQ SAMPLE** test point and carefully adjust the 4 **MHz FREQ ADJUST** until the frequency counter displays 4.000 000 MHz ± 4 Hz.

8.3.2 Receiver

Cat. No. 466/486 Receiver Module – 10 MHz Frequency Adjust

The **10 MHz FREQ ADJUST** sets the reference for the PLL master oscillator frequency synthesizer on the receiver module. The adjustment can be made from the front panel.

Prior to any frequency adjustment, power should be applied to the receiver and the frequency counter for at least 30 minutes.

Connect the frequency counter to the **10 MHz FREQ SAMPLE** test point and carefully adjust the **10 MHz FREQ ADJUST** until the frequency counter displays 10.000 000 MHz ± 1 Hz.

8.4 Test Functions – Transmitter

8.4.1 Cat. No. 454/474 Modulator Module

The Normal/Test switch is used in conjunction with the I or Q and Clock test points to confirm proper operation of the modulator without first demodulating the output. This is done by observing the eye patterns formed when an oscilloscope is triggered by the Clock output while observing either the I or Q modulated outputs. The **TEST** switch position produces a frequency response from the modulator digital filter that includes both the response of the transmit roofing filter and the receive baseband filter. This composite response is necessary in order to observe normal open eye patterns.

Note Do not operate the DSTL with the Normal/Test switch in the **TEST** position. Degraded BER performance will result.

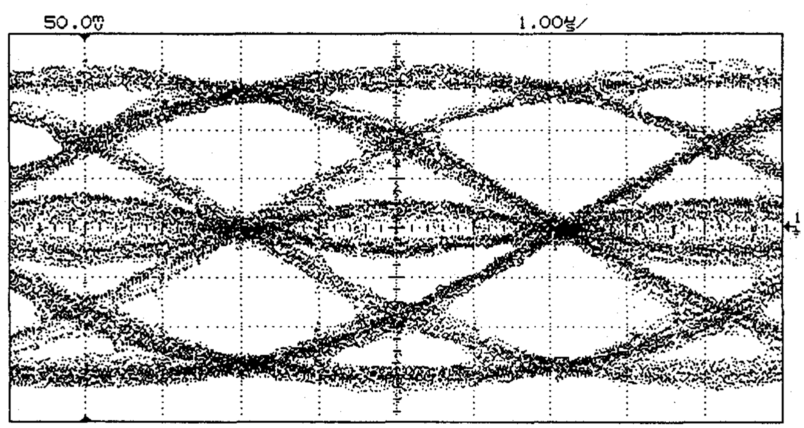


Fig. 8.1 Typical eye pattern in **TEST** mode

The eye pattern above is representative of the display on a scope connected to either the I or Q test point. Glitches may be observed depending on the bandwidth of the scope or probe type. See section 9.5.1 for troubleshooting applications.

8.4.2 Cat. No. 476 Frequency Sythesizer/RF Power Amplifier

RF Level Adjustment

The **RF LEVEL ADJUST** controls the input drive level into the solid state power amplifier module within the Cat. No. 476. Turning counterclockwise will decrease the RF power output from the **RF OUT** port. To set for 2 Watt output, set front panel meter switch to **FORWARD PWR** and monitor the meter while adjusting the **RF LEVEL ADJUST** control.

RF Transmit Spectrum

A typical DP5501A transmit spectrum is shown in Figure 8.2. The spectrum shown is centered at 950 MHz ±125 kHz. Obtain an RF sample to check IMD levels and conformance to RF spectrum masks governing digital STL transmission systems (the USA FCC mask is shown in Figure 8.3).

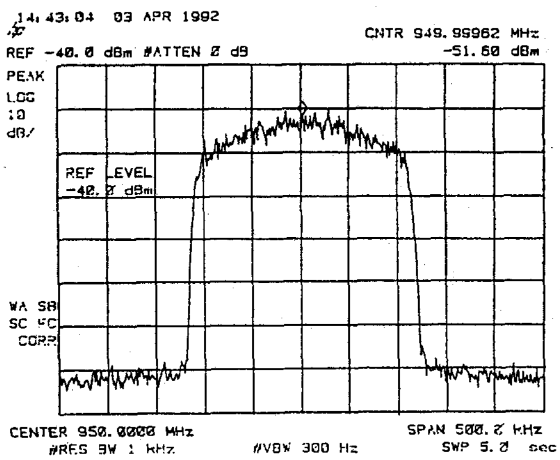


Fig. 8.2 Typical DP5501A transmit spectrum

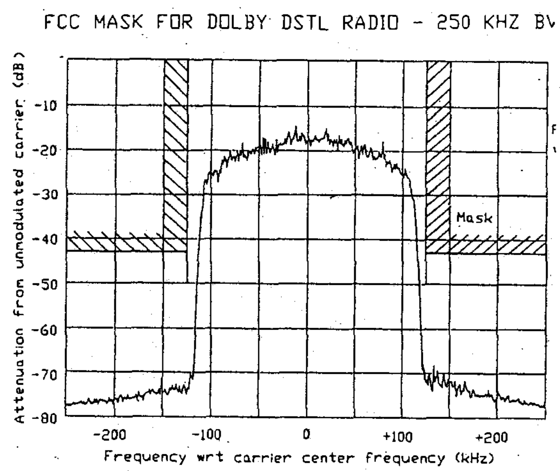


Fig. 8.3 Typical DP5501A transmit spectrum with mask applied

8.5 Test Functions – Receiver

8.5.1 Cat. No. 463/483 Audio Decoder Module

The Cat. No. 463/483 Audio Decoder provides a **Normal/Test** mode switch in order to facilitate system alignment and allow stand-alone verification of audio performance in the receiver. When the test mode of operation is selected, calibrated audio test tones are substituted in place of the demodulated audio data from the Cat. No. 464/484 Demodulator. Digitally generated 400 Hz sine waves at a level of -12 dB are produced for the Main and Aux audio channels. These can be used in conjunction with the output level controls of the Cat. No. 462/482 D/A Converter to set system operating levels, and to verify operation and performance of the Cat. Nos. 462/482 and 463/483, independent of transmitter operation.

Caution When switching between Normal and Test mode, a significant level difference may exist between audio program and the test tone. Be aware that you will be broadcasting these tones if you activate your FM transmitter while operating in Test mode.

8.5.2 Cat. No. 466 Receiver Module

RF Receive Spectrum

A typical DSTL receive spectrum in the presence of adjacent analog STL channels is shown in Figure 8.4. A large signal adjacent to the desired signal can cause a receiver to AGC (level track) to the larger undesired signal. This phenomenon can cause high BER values.

The DP5502 DSTL combines advanced SAW filters with excellent selectivity and a sophisticated AGC circuit that is dependent only on the desired receive signal strength. This combined with the use of 9-QPRS modulation makes the DP5502 highly tolerant to adjacent channel interference. In a real life example, a DP5502 with a receive level of –70 dBm can tolerate a –56 dBm FM adjacent signal 225 kHz away producing a BER of 10^{-4}. The margin increases to +52 dB if the adjacent interference is 500 kHz away from the desired signal.

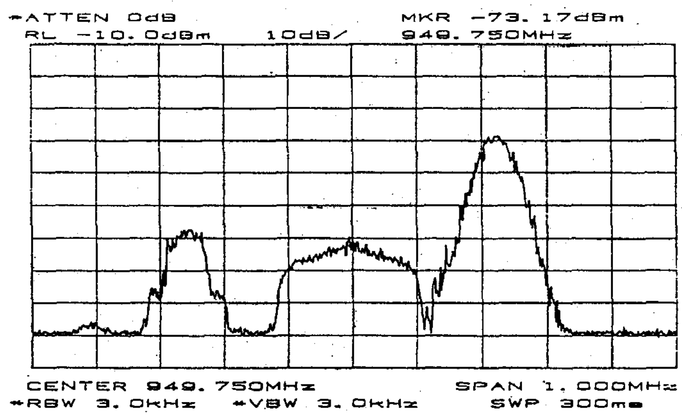


Fig. 8.4 Receive spectrum in the presence of adjacent channel STLs

Figure 8.5 is an illustration of a FM STL spectrum co-channel signal interfering with a DSTL signal. This representation cannot actually be observed with a spectrum analyzer. This particular example is a double plot, once with both signals present and a second plot with the DSTL transmitter turned off. The presence of co-channel interference may not always be apparent because it can be masked by the DSTL transmit spectrum. A high BER reading however, is an indication of significant co-channel interference. The DSTL 10^{-4} BER threshold from another DSTL or FM STL co-channel interferer is –15 dBc.

To confirm the presence of co-channel interference, connect a spectrum analyzer to the **IF MON** port on the DP5502 receiver and observe the 70 MHz spectrum with the DSTL transmitter turned on. Compare this spectrum with the DSTL transmitter turned off. If the co-channel interference is caused by a FM STL, the spectrum would look like the enclosed spectrum portion only of Figure 8.5. (Although the figure shows the spectrum at 950 MHz, the display at 70 MHz will look the same.) You can troubleshoot co-channel interference from another DSTL by the same method. The interfering signal would look like a typical DSTL spectrum.

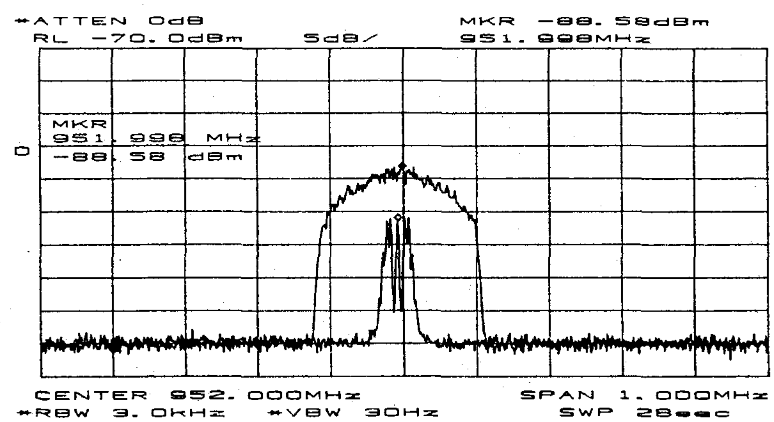


Fig. 8.5 Co-channel interference from an analog STL

IF MON

The **IF MON** port can be used to monitor the IF output power and the frequency spectrum of the receiver module. The spectrum at this port is centered at 70 MHz ± 125 kHz for a DSTL. Figures 8.6 and 8.7 are typical spectra observable at the **IF MON** port. A spectrum analyzer can be connected to this port for troubleshooting interference problems. The power level out of the monitor port is nominally 11 dB below the receiver IF output.

Figure 8.6 shows a down-converted DSTL (DP5502) receive spectrum with adjacent channel analog STL spectra. The DSTL spectrum is at the center of the plot with the interference signals to the left. One is 200 kHz and the other is 375 kHz away from the DSTL signal. For a BER of 10^{-4}, the DSTL can sustain a +11dB interference signal 200 kHz away and +49 dB signal 375 kHz away.

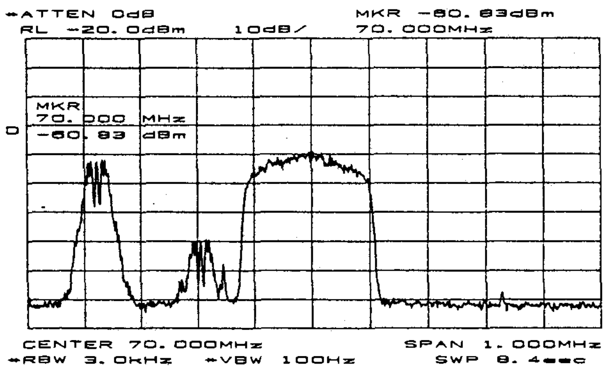


Fig. 8.6 Adjacent channel analog STL interference

A down-converted receive spectrum of two 2-channel DSTL channels adjacent to each other is shown in Figure 8.7. The desired DSTL channel can tolerate a +40 dB higher adjacent DSTL signal 250 kHz away for a BER of 10^{-4}.

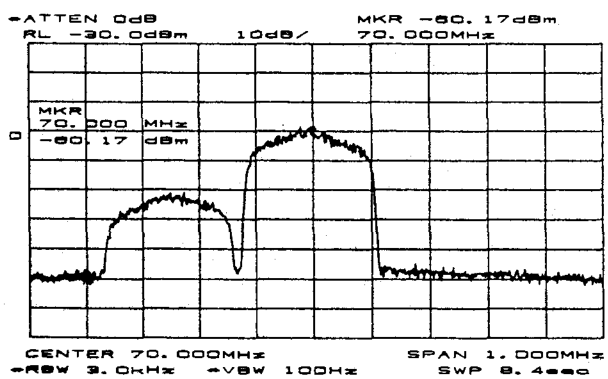


Fig. 8.7 Two DSTL channels adjacent to each other

The table below shows typical measured results from a FM STL to DSTL and DSTL to DSTL adjacent interference tests. Columns 2 and 3 indicate the required interference level needed to produce a 10^{-4} BER at the DP5502 DSTL.

Test Conditions:

Desired receive level:	–70 dBm at 952 MHz
Interference signal:	FM STL, 2 Channel Stereo, plus 185kHz SCA, 500 kHz bandwidth
DSTL:	DP5501A

FREQ SEPARATION	FM STL into DSTL	DSTL into DSTL
(kHz)	(dB)	(dB)
500	52	44
375	49	45
350	48	44
275	23	42
250	16	40

8.5.3 Cat. No. 460 Digital Stereo Generator

Various test functions are incorporated into the design of the Digital Stereo Generator to facilitate alignment and Proof-of-Performance.

8.5.3.1 Initial Setup (repeated from Section 7, "Virtual Composite Operation")

Notes For this setup, the **CAL** indicators next to the **METER FUNCTION** labels are not used. References to left and right channels in this section apply to both program pairs in the DP5503A.

At the Studio:

1. Select program material (left and right channels) that is the loudest typically played.

2. Set audio processor controls to the highest degree of processing typically used.

3. The following step not only maximizes S/N ratio through the DSTL system, but also activates the overshoot limiter when needed, and matches the output of the digital stereo generator in the DSTL receiver to 100% modulation:

Feed this "loud" audio into the processor, and in turn, to the DSTL transmitter. Monitor level with DSTL **METER FUNCTION** switch set at **LEFT**. Adjust DSTL's **Left** channel **INPUT LEVEL** so that peak meter indication approaches but never goes over 0 dB on the meter.

The following steps balance left and right channel levels:

4. Replace "loud" program material with 400 Hz tone from the console and feed both channels of the audio processor.

5. Note the **Left** channel input level on DSTL Transmitter meter. Move **METER FUNCTION** switch to **RIGHT** and adjust **Right** channel **INPUT LEVEL** to match left channel level.

6. Replace tone with program material. Return processor to the usual operating settings.

At the Transmitter Site:

Note If you do not have a modulation monitor at the transmitter site, you will need someone at the studio to read the monitor indications when completing stereo generator alignment at the transmitter site.

100% Modulation & Pilot Injection

1. Set the DSTL **MODE** switch to **OPERATE**.

2. On the **Digital Stereo Generator** module, Cat. No. 460, flip the **TEST OSC** switch to **400 Hz** (**STATUS** light blinks to indicate non-standard operational mode). Using the modulation monitor with its mode switched to monitor left and right channels, adjust the module's composite **OUT**put level for a reading of 100% modulation at the monitor. Return the **TEST OSC** switch to **Off**.

3. Monitor Pilot with the modulation monitor and adjust the **PILOT Inj**ection control to the desired operating level (typically 8-10 %).

4. Upon activation of the DSTL system (see Section 6, Operation), you may need to adjust the composite **OUT**put level slightly to set modulation levels in accordance with your operating practices. Since changing the composite output level also changes the pilot level, be sure to also re-adjust the **PILOT Inj**ection control.

Notes When monitoring the **HEADPHONE MONITOR** outputs of either DSTL transmitter or receiver, you will hear the pre-emphasized input signal.

The meter of both the DSTL transmitter and receiver display the pre-emphasized signal. Slight discrepancy between transmitter and receiver meter deflections are normal, due to very slight overshoots through the system.

Some of the above procedures use tones at 100% modulation. Be aware that you will be broadcasting these tones if you activate your FM transmitter carrier during these setup procedures.

Separation

The Digital Stereo Generator has been optimized for channel separation before leaving the factory. However, due to the uniqueness of each station's setup, it may be necessary to trim the **L-R LEVEL** for best separation, according to the following steps:

1. Configure your modulation monitor to read channel separation (residual on right channel with left channel driven).

2. On the **Digital Stereo Generator** module, turn the **TEST OSC** switch to **400 Hz** and the **RIGHT CHAN** switch to **Off**.

3. Note the current position of the **Coarse** and **Fine** switches of **L-R LEVEL** in case you stray too far from reasonable separation performance. Then, alternately adjust the **Coarse** and **Fine** switches until best separation is achieved.

Note The above procedures use tones at 100% modulation. Be aware that you will be broadcasting these tones if you activate your FM transmitter carrier during these setup procedures.

8.5.3.2 Crosstalk Measurements

Note If you do not have a modulation monitor at the Transmitter site, you will need someone at the Studio to read the monitor indications when performing the following procedures at the transmitter site.

1. **Main Channel-to-Subchannel Crosstalk**
   On the Digital Stereo Generator module, Cat. No. 460, flip the **TEST OSC** switch to 400 Hz (**STATUS** light blinks to indicate non-standard operational mode). Leave the **R CHAN** switch in **Norm**. This generates an equal level test tone in both left and right channels at 100% modulation (an L + R signal). Using the modulation monitor with its mode switched to measure Main-to Sub Crosstalk, read the residual signal in the L - R subchannel.

2. **Subchannel-to-Main Channel Crosstalk**
   On the Digital Stereo Generator module, Cat. No. 460, flip the **TEST OSC** switch to 400 Hz (**STATUS** light blinks to indicate non-standard operational mode). Also flip the **R CHAN** switch to **INV**. This generates an equal level, but out-of-phase test tone in both left and right channels at 100% modulation (an L - R signal). Using the modulation monitor with its mode switched to measure Sub-to-Main Crosstalk, read the residual signal in the L + R channel.

Note Be sure to return switches to their "**normal**" positions to restore normal operation.

9.1 Introduction

Note Please review Section 11 regarding Return Exchange Order (RXO) procedures for obtaining replacement modules in the event that your troubleshooting indicates that certain DSTL modules may be defective.

The Models DP5501A/5503A and DP5502/5504 are equipped with numerous features to assist in troubleshooting. Individually or in combination, they are useful tools to gauge the general condition of the DSTL installation. They include:

- LEDs indicating module status;
- Relay closures reflecting summary alarm and signal mute conditions;
- Panel meters indicating RF parameters, error rate, received signal strength, or audio levels;
- Test points and test modes (see also Section 8); and headphone outputs.

Because the DSTL system is part of a radio station's audio signal chain that may consist of numerous pieces of audio equipment and extensive wiring and connections, failure of any one of them, or inadvertent "cockpit errors," could cause the station to be off the air. Thus, prudent troubleshooting practices dictate that adequate investigation of the complete signal chain be performed to ensure that a problem attributed to the DSTL system does not lie elsewhere.

Within the DSTL system, comprising antennas, towers, transmission line, related accessories, and the path itself, there may be factors that contribute to signal loss. Be sure to examine these potential causes as well.

Finally, if the investigation points to trouble with the DSTL transmitter or receiver, use of the troubleshooting guidelines outlined below should result in speedy diagnosis and problem resolution. Because transmitter faults may manifest themselves in the receiver, keep in mind that a problem which appears at the receiver may be attributable to the transmitter.

An effective procedure for troubleshooting a broadcast chain incorporating the DSTL units is as follows:

Check external equipment/connections ahead of the DSTL system. Also check for "cockpit errors."

↓

Check path-related factors:
Transmission line, antennas, connectors, path problems, etc.

↓

Check DSTL Transmitter

↓

Check DSTL Receiver

Fig. 9.1 DSTL Troubleshooting Strategy

9.2 Eliminating non-DSTL Problems from Consideration

Check External Equipment / Connections

- Interconnections not secure;
- Connectors intermittent, broken, etc.;
- Audio processing, external stereo generators, etc., defective
- Line voltage below specifications

Check for "Cockpit Errors"

Make sure that:

- Outputs feeding the DSTL have not been inadvertently shut off or disconnected;

- Operational modes on the DSTL units have not been changed:
  MODE switch;
  Synthesizer **S1**, **S2**, **S3** switch settings;
  Switching to any of the test modes;
- Jumper settings are correct:
  Some jumpers set on the transmitter require matching settings on the receiver. Non-complementary settings between transmitter and receiver may give unusual meter readings (e.g.: setting the Signal Adaptation jumper/switch on the transmitter without setting the corresponding jumper/switch on the receiver will give receiver audio readings that are much higher than on the transmitter).
- Fuse is installed and is good; power supply module voltage selector is set to the correct position.

Check the Path

Check RF interconnections among "pigtails," main transmission line, antennas; any auxiliary devices such as combiners, splitters, isocouplers, cavities, etc. Also insure signal passage through this equipment.

In areas of spectrum congestion, do not discount the possibility of new sources of interference.

Determine if unusual circumstances (high winds, storms, precipitation, inversion layers) are adversely affecting your path. Wind may shift STL antenna orientation, resulting in a weaker signal and/or increased interference.

Note Refer to Section 8 for procedures on using test points and test modes to aid in troubleshooting path problems.

Using DSTL Meter Indications to Troubleshoot RF/Path Problems

Transmitter

Use the **FWD PWR** and **REV PWR** indications to ensure that the transmitter is properly terminated. Forward power should read nominally 2 Watts (or lower, if you adjusted the output to a lower amount), and reverse power should be low. If VSWR exceeds 3:1, the **Summary Alarm** indicator will light.

Receiver

The combination of the **RF RCV LEVEL** and **ERROR RATE** meter positions on the DSTL Receiver are useful in assessing the condition of your DSTL path, as summarized in Figure 9.2.

RF RCV LEVEL	ERROR RATE	PROBABLE CAUSE	COMMENTS
High	Low	**Desired Operation**	
Creeping Low	Creeping High	Signal Fade Occurring	Monitor carefully. If system never mutes, you've done your path calculations properly.
Low	Medium to High	In Deep Signal Fade	If muting occurs, not enough fade margin was designed into the path or an extremely unusual problem is causing deeper fades than anticipated.
High	High	Interference or Multipath Problem	Troubleshoot the path using test points discussed in Section 8 or consult with the local frequency coordinating committee

Fig. 9.2 Receiver meter indications for monitoring path conditions

9.3 Signal Flow in the DSTL System

As explained in the next section, DSTL faults can be identified by red **STATUS** LED indications on most DSTL modules, which are also tied into a **Summary Alarm** indication.

Fault isolation may not be as simple as just replacing modules with a red LED. Because signal flow is unidirectional, faulty output from one module can cause a red **STATUS** LED indication on the following module. Even if it is the only one lit, it may be caused by an undetected problem on a module earlier in the signal flow.

Where more than one module has a red LED, the problem is likely to reside on the module earliest in the signal flow.

Some knowledge of signal flow is useful to troubleshooting, and is depicted in Figures 9.3 and 9.4 below.

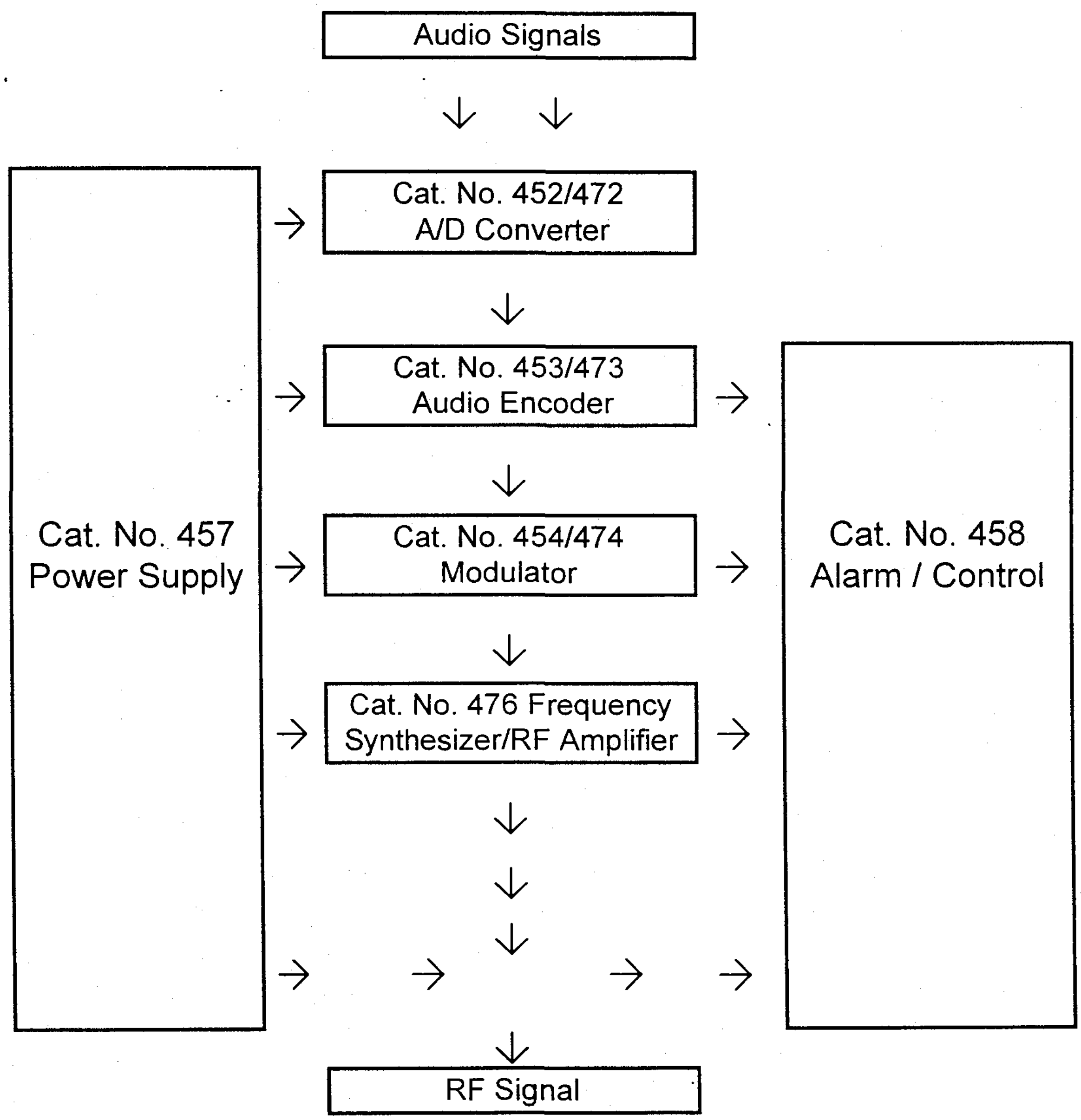


Fig. 9.3 Signal Flow, DSTL Transmitter

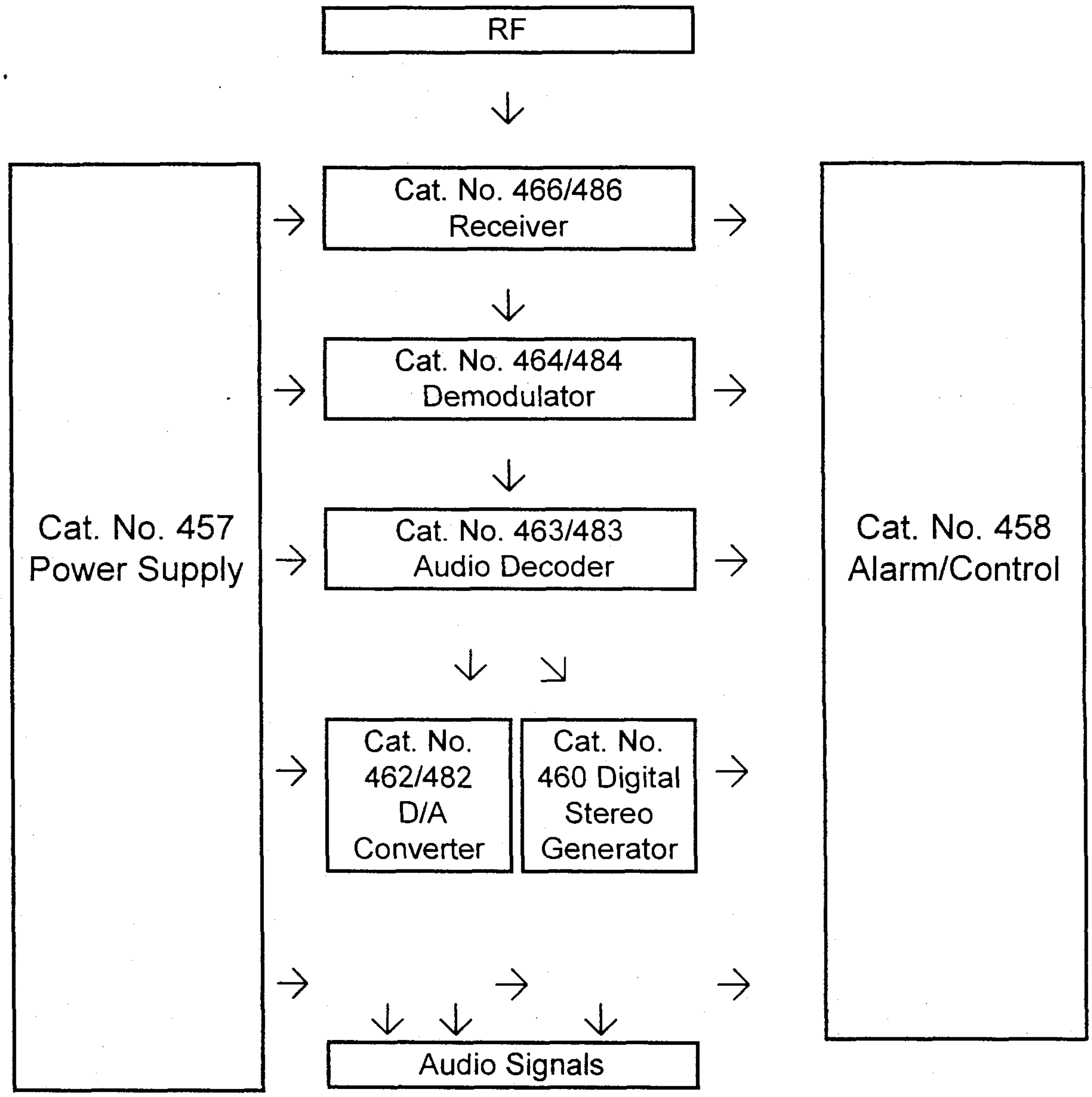


Fig. 9.4 Signal Flow, DSTL Receiver

Note In the above figures:

The power supply feeds all modules, including the Alarm/Control module, Cat. No. 458.

The only modules **not** monitored by the Alarm/Control module are:
a) A/D Converter module, Cat. No. 452/472 (transmitter);
b) D/A Converter module, Cat. No. 462/482 (receiver).

See Section 9.4 for procedures on troubleshooting these modules.

9.4 Isolating Faulty Modules using Module Status Indicators

The primary method used to isolate DSTL faults is based on the examination of the color of the module **STATUS** indicators. The color of all module **STATUS** LEDs are normally green. They turn red due to either a detected malfunction on the module or incorrect input conditions to the module.

Malfunctioning modules can be isolated using the following tables. Isolate defective modules by identifying the first module in the signal chain with a red status LED.

9.4.1 DP5501A/5503A Transmitter

Red STATUS LED On	Comments	Replace Module(s)/ Remedies:
Cat. No. 453/473	See below	Cat. No. 453/473 + Cat. No. 452/472
Cat. No. 454/474		Cat. No. 453/473 + Cat. No. 454/474
Cat. No. 476	Synthesizer out of lock. Check that **S1, S2, S3** agree between transmitter and receiver. Push **RESET** button.	Cat. No. 476
	First check **REV PWR** position of meter to see if excessive reflected power (high VSWR) is causing a fault indication.	Troubleshoot transmission line/antenna subsystem.
	Also check output power level. Power settings less than 0.25 watt or much higher than 2 watts will cause a Fault indication.	Use RF LEVEL ADJUST on Cat. No. 476 to adjust output power to acceptable range.
	Check if PA heat sink is unduly hot.	Reduce ambient temperature.
	None of above conditions eliminates the fault indication.	Cat. No. 476
Cat. No. 457	Shows red whether all other modules are in place or all have been removed.	Cat. No. 457 + Cat. No. 458
	Changes from red to green when a particular module is removed.	Cat. No. 457 + offending module
	Power Supply is outside of tolerance, or line voltage low , or excessive ripple is present.	Check for low line voltage. Also check Power Supply test points (Section 3, Cat. No. 458 figure).
Summary Alarm LED	Only the **Summary Alarm** LED is red, with all other Status LEDs green	Cat. No. 458

Fig. 9.5 Transmitter Red STATUS Indicators (Sheet 1 of 2)

Red STATUS LED On	Comments	Replace Module(s)/ Remedies:
STATUS LED off on any module (neither green or red).	Indicator circuit malfunction, or one or more power supply voltages are absent on the module.	Offending module
Any module	There is no accompanying **Summary Alarm** indication.	Cat. No. 458 + offending module

Fig. 9.5 Transmitter Red STATUS Indicators (Sheet 1 of 2)

Cat. No. 453/473

Problems on the Cat. No. 453/473 can be partially diagnosed using the red LEDs mounted on the circuit board. These can be viewed with the module in place by removing the DSTL front subpanel. Refer to Figure 9.7 for LED locations.

Note Remove and then re-apply power to the DSTL unit before referring to the table below.

Flashing LED	Replace
DS706 (Cat. No. 453)	Main channel PROM (BE Part Number 84036) IC701
DS701 (Cat. No. 473)	Main channel PROM (BE Part Number 84089) IC704
DS710 (Cat. No. 453)	Aux channel PROM (BE Part Number 84037) IC708
DS705 (Cat. No. 473)	Aux channel PROM (BE Part Number 84089) IC714
DS711 (Cat. No. 453)	Cat. No. 453 + Cat. No. 452
Any other LED flashing	Cat. No. 453/473
No LEDs flashing	Cat. No. 453/473 + Cat. No. 452/472

Fig. 9.6 Additional Troubleshooting of the Encoder Module, Cat. No. 453/473

Note PROMs on the Encoder and Decoder modules (Cat. No. 453/473 and 463/483, respectively) must be of the same revision level. Be prepared to communicate the complete part number located on the top of the PROM to your distributor or to Dolby when requesting replacement PROMs. Both encoder and decoder PROMs will be sent if the current revision level is newer than those installed in your unit.

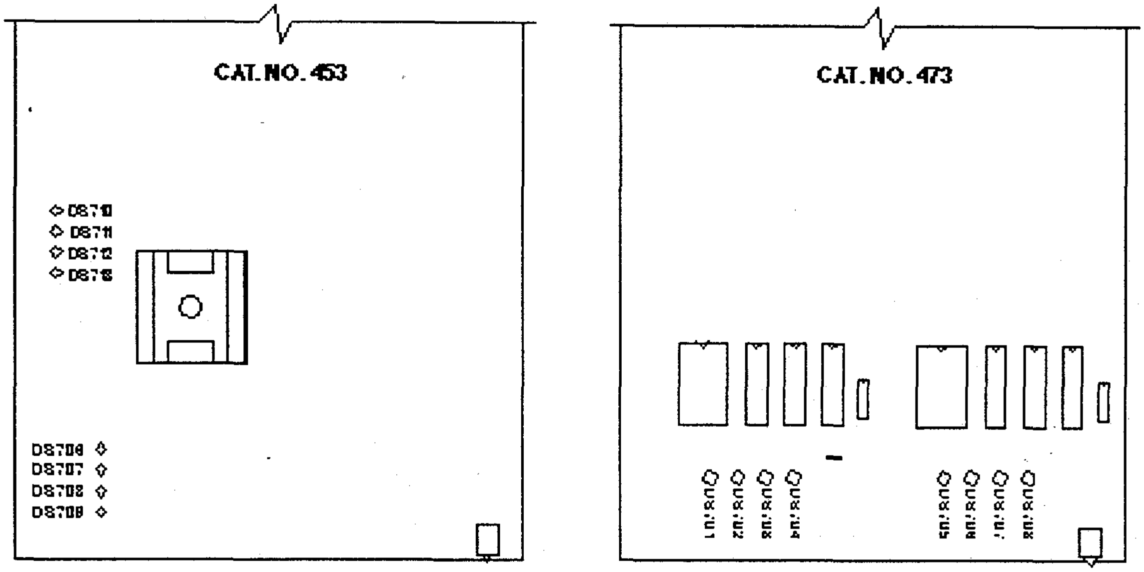


Fig. 9.7 Cat. No. 453/473 LED Locations

9.4.2 DP5502/5504 Receiver

Review Section 9.5.1 discussing testing of the Cat. No. 454/474 Modulator module on the DSTL transmitter. Also check that the transmitter is radiating RF and that there is adequate received signal strength.

The STATUS LED on the Digital Stereo Generator, Cat. No. 460, also has a flashing green indication that warns that test modes have been enabled. Check the positions of the toggle switches on the module. The test mode can be cleared by returning the toggle switches to their "normal" positions. See Sections 7 (Setup) and 8 (Test).

STATUS LED Red On	Comments	Replace Module(s):
Cat. No. 466/486	Synthesizer out of lock. Check that **S1, S2, S3** agree between transmitter and receiver. Push **RESET** button.	Cat. No. 464/484 + Cat. No. 466/486
Cat. No. 464/484		Cat. No. 464/484 + Cat. No. 466/486
Cat. No. 463/483	see below	Cat. No. 463/483

Fig. 9-8 Receiver Red STATUS Indicators (Sheet 1 of 2)

STATUS LED Red On	Comments	Replace Module(s):
Cat. No. 460		Cat. No. 460 + Cat. No. 463/483
Cat. No. 457	Shows red whether all other modules are in place or all have been removed.	Cat. No. 457 + Cat. No. 458
	Changes from red to green when a particular module is removed.	Cat. No. 457 + offending module
	Power Supply is outside of tolerance, or line voltage low, or excessive ripple is present.	Check for low line voltage. Also check Power Supply test points (Section 3, Cat. No. 458 figure)
Summary Alarm LED	Only the **Summary Alarm LED** is red, with all other Status LEDs green	Cat. No. 458
STATUS LED off on any module (neither green or red).	Indicator circuit malfunction, or one or more power supply voltages are absent on the module.	Offending module
Any module	There is no accompanying **Summary Alarm** indication.	Cat. No. 458 + offending module

Fig. 9-8 Receiver Red STATUS Indicators (Sheet 2 of 2)

Cat. No. 463/483

Problems on the Cat. No. 463/483 can be partially diagnosed using the red LEDs mounted on the circuit board. These can be viewed with the module in place by removing the DSTL front subpanel. Refer to the locations of these LEDs on Figure 9.10.

Note Remove and then re-apply power to the DSTL unit before referring to the table below.

Flashing LED	Replace
DS706 (Cat. No. 463)	Main channel PROM (BE Part Number 84038) IC701
DS701 (Cat. No. 483)	Main channel PROM (BE Part Number 84090) IC704
DS710 (Cat. No. 463)	Aux channel PROM (BE Part Number 84039) IC708
DS705 (Cat. No. 483)	Aux channel PROM (BE Part Number 84090) IC714
Any other LED flashing	Cat. No. 463/483
No LEDs flashing	Cat. No. 463/483 + Cat. No. 464/484

Fig. 9-9 Additional Troubleshooting of the Decoder Module, Cat. No. 463/483

Note PROMs on the Encoder and Decoder modules (Cat. No. 453/473 and 463/483, respectively) must be of the same revision level. Be prepared to communicate the complete part number located on the top of the PROM to your distributor or to Dolby when requesting replacement PROMs. Both encoder and decoder PROMs will be sent if the current revision level is newer than those installed in your unit.

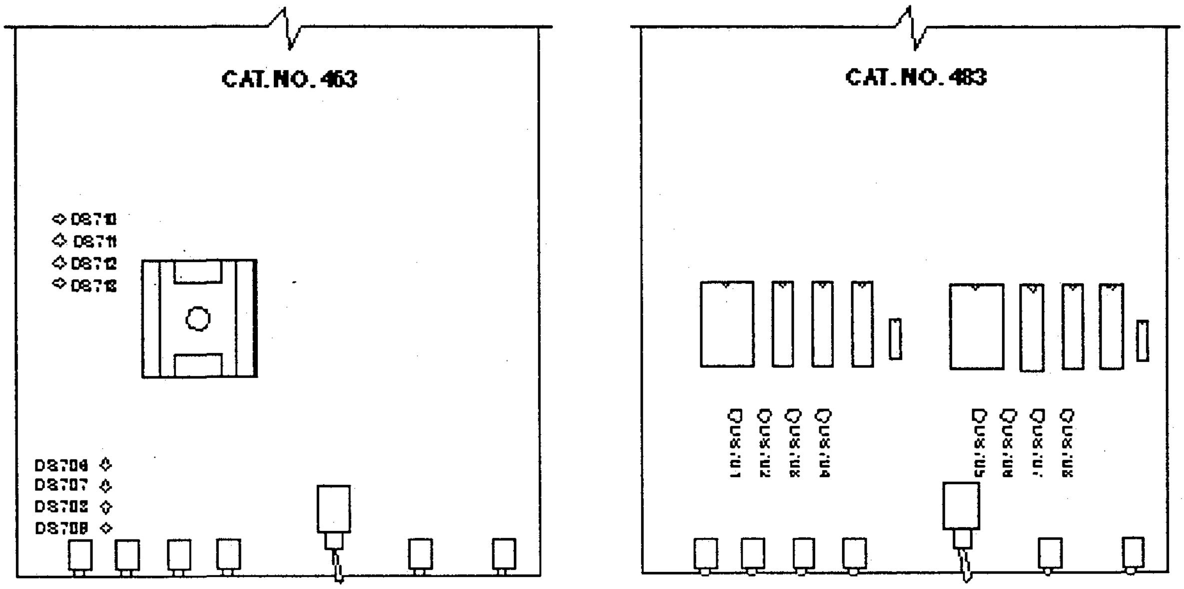


Fig. 9.10 Cat. No. 463/483 LED Locations

9.5 Transmitter - Receiver Interaction

(Transmitter problems that may show up at the receiver)

Exercise particular caution when interpreting the meaning of status indicators on the DP5502/5504 DSTL receiver (other than the Cat. No. 460 and power supply), because the absence of a transmitted signal or any data corruption in the transmitter that is not detected by its own **STATUS** LEDs may produce red **STATUS** LED indications in a correctly functioning receiver. The presence of red **STATUS** LEDs in the DP5502/5504 receiver requires an examination of the DP5501A/5503A transmitter for either red **STATUS** LEDs or correct operation of the Cat. No. 454/474 Modulator module.

9.5.1 Functional Test: Modulator Module, Cat. No. 454/474

Although the Cat. No. 454/474 Modulator module has a **STATUS** LED, an additional test is useful to verify proper operation. For this test, you will need an oscilloscope (10MHz minimum bandwidth) and 2 probes (BNC cables work best as probes due to their 200 - 300 pF capacitance), for I or Q, and trigger.

Procedure:

Connect the scope signal probe to the **I** or **Q** test point. Connect one ground lead of the two probes to the **ground** pin.

Connect the scope trigger probe to the **Clock** test point.

Set the oscilloscope settings to 50mV/div., 1 or 2 μs/div., DC coupling.

The oscilloscope is used to look at the **I** and the **Q** data waveforms while it is being triggered by the signal present on the **Clock** test point.

In normal operation, the waveform at the test point appears scrambled due to the fact that the decoder filter has yet to act upon the signal. As a result, the verification of proper operation is made more difficult. This difficulty is eliminated by use of the **Test** mode on the Cat. No. 454/474. When the module is put in this mode, the waveform will show a good differentiation between data levels at the sample intervals.

Place the **MODE** switch to **Test**. A properly working modulator in **Test** mode will produce the following waveform:

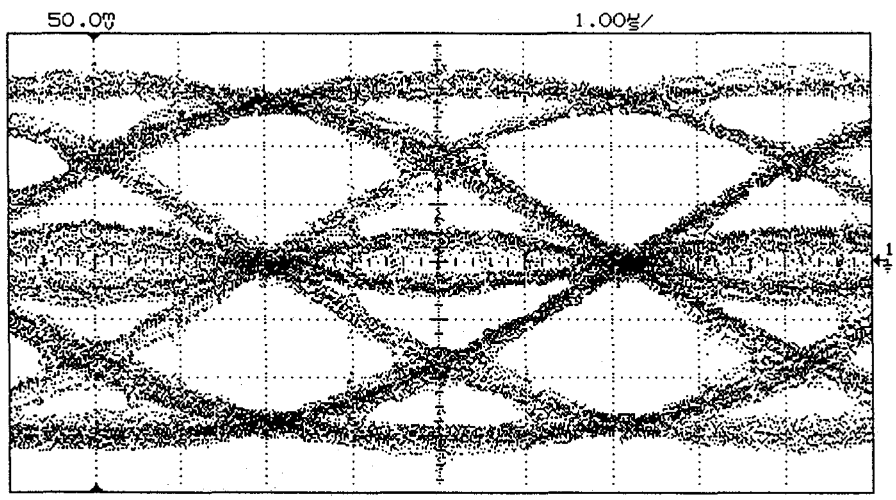


Fig. 9.11 Appearance of normal I or Q data pattern during test mode

If the waveform does not resemble the above figure, then the modulator is defective and should be replaced.

Note There may be clock glitches approximately every 500 nsec. These are an artifact of the test circuitry and should be ignored. They can be minimized by using scope cables that have 200 - 300 picofarads capacitance.

When the Cat. No. 454/474 is switched into the **Test** mode, the error rate will increase dramatically and the audio channels may mute at the receiver location. This is normal.

9.6 Isolating Defective Modules if No Red Status Indicators or Audio Mutes are Present

If the combination of both the DP5501A/5503A and DP5502/5504 are not operational and no status indicators are red, the following procedures may be useful:

9.6.1 Investigating the DP5501A/5503A Transmitter

Apply audio to the main, auxiliary, and voice (DP5501A) channels, then examine the meter for signal activity. Next, listen to the audio using the headphone jack to determine whether audio is present and undistorted.

If the meter or the headphones indicate an undistorted signal but no audio or distorted audio is present at the receiver output, then refer to the Encoder module, Cat. No. 453/473, troubleshooting instructions in Section 9.4.1

If either the meter shows no signal or the headphone monitor reveals distorted audio, replace the Cat. No. 452/472 A/D module.

9.6.2 Investigating the DP5502/5504 Receiver

Set the **MODE** switch to **Test** on the Audio Decoder module, Cat. No. 463/483, and verify the presence of 400 Hz audio on the main and auxiliary program outputs. The headphone monitor can be used for this test. If no audio is present, determine that there are no output shorts. If not shorted, examine the meter for a –12 dB reading on the main and auxiliary channels.

If there is a correct meter reading, then replace the D/A Converter module, Cat. No. 462/482. If there is no meter reading, then refer to the Decoder module, Cat. No. 463/483 troubleshooting instructions in Section 9.4.2.

If there is undistorted audio at the outputs, the problem is probably related to the DSTL transmitter or an unusual condition at the studio site.

If the Digital Stereo Generator, Cat. No. 460, is installed, troubleshooting is simplified somewhat:

If there is a problem with **both** audio and composite outputs, replace the Decoder module, Cat. No. 463/483.

If the composite output is normal, then replace the D/A module, Cat. No. 462/482.

If the audio outputs are normal but the composite output is bad, replace the Digital Stereo Generator, Cat. No. 460.

9.7 Interpreting Status Relay Closures

The fault status of all modules with **STATUS** indicators is summarized by the **Summary Alarm** LED on each DSTL unit, as well as by the rear panel **ALARM** relay closure. Certain other relay closures may confirm or independently highlight problems (either equipment faults or setup/operation errors). They are summarized below.

Note States described in the following table are associated with relay closures such that there is a circuit formed between the "Common" contact and the contact shown in the third column.

Relay	State	Short between Common (C) and	Description	Remedy
ALARM	T	NC contact	A fault has been detected in the unit that has also caused the **Summary Alarm** LED to light.	See Section 9.8 below "Faults Detected by STATUS LEDs"

Fig. 9-12 Status Relays (Sheet 1 of 3)

Relay	State	Short between Common (C) and	Description	Remedy
OPERATE	F	**NC** contact	The unit is no longer in operational mode, either because the front panel **MODE** switch has been placed into **REMOTE** or **STANDBY**; or a Hot Standby unit has taken this unit out of **Operate**. Certain faults (indicated by the **ALARM** closure & LEDs) have also caused the unit to switch out of **OPERATE.**	Return **MODE** switch to **OPERATE**. Consult with remainder of this section to troubleshoot reasons why switchover occurred; look for **Fault** LED indications. See Section 9.8 below "Faults Detected by STATUS LEDs"
IN REMOTE	T	**NO** contact	The front panel **MODE** switch has been placed in the **REMOTE** position to enable control by a Hot Standby unit or to enable operation by remote control.	If hot standby units are not present, or remote control is not desired, return **MODE** switch to **OPERATE**.

Fig. 9-12 Status Relays (Sheet 2 of 3)

Relay	State	Short between Common (C) and	Description	Remedy
AUDIO MUTE (receiver only)	T	NC contact	Excessive bit errors have caused the main program channels--Left (1), Right (1) (and composite, if Cat. No. 460 is installed)--to mute. Accompanied by **Summary Alarm.**	See Section 9.8 below.
AUX MUTE (receiver only)	T	NC contact	Excessive bit errors have caused the Aux channels to mute (and composite, if Left 2 and Right 2 are assigned to the Cat. No. 460 in the DP5504). Accompanied by **Summary Alarm.**	See Section 9.8 below.

Fig. 9-12 Status Relays (Sheet 3 of 3)

9.8 Faults Detected by STATUS LEDs

The following tables describe the various circuits that are monitored by **STATUS** LEDs on the modules of the DSTL transmitter and receiver.

9.8.1 Transmitter

Module Name	Cat. No.	Circuit(s) Monitored
Audio Encoder	453/473	5V power supply too low PLL out of lock Program not executing properly Not monitored: Output buffers Frequency divider circuit
Modulator	454/474	System clock PLL out of lock I or Q channel amplitude too low
Frequency Synthesizer/RF Power Amplifier	476	Main PLL out of lock Not monitored: Up converters RF buffers Output power outside limits *High spurious signals at output* Excessive VSWR High heatsink temperature ALC out of range Output power outside window Negative power supply absolute voltage too low Not monitored: Output buffers for meter/control Distortion problem causing excessive sidebands Low RF input power
Power Supply	457	Any power supply voltage outside window Excessive output voltage ripple **Note** The monitor circuit and LED are located on the Alarm / Control module, Cat. No. 458.
Alarm / Control module	458	Alarm logic not monitored. This circuitry drives the **SUMMARY ALARM** LED.

Fig. 9-13 Transmitter circuits monitored by **STATUS LEDs**

9.8.2 Receiver

Module Name	Cat. No.	Circuit(s) Monitored
Receiver	466/486	Main PLL out of lock AGC voltage is outside window RF power input is outside limits
Demodulator	464/484	Excessive pseudo-error voltage Not monitored: Demodulating logic after the ADCs Loss of carrier or bad input stages Loss of PLL clock
Audio Decoder	463/483	5V power supply too low PLL out of lock Program not executing properly Out of sync for the main channel Not monitored: Output buffers Frequency divider circuit Test tone generator
Digital Stereo Generator	460	5V power supply too low Program not executing properly Not monitored: Output DAC, filters, and buffers Switch operation
Power Supply	457	Any power supply voltage outside window Excessive output voltage ripple **Note** The monitor circuit and LED are located on the Alarm Module Cat. No. 458.
Alarm / Control	458	Alarm logic not monitored. This circuitry drives the **SUMMARY ALARM** LED.

Fig. 9-14 Receiver circuits monitored by **STATUS LEDs**

SECTION 10
SYSTEM DESCRIPTION AND BLOCK DIAGRAM

NOTE This section provides an overview of the DSTL system. For complete information, refer to the companion Technical Manual, available to registered users. You can order this manual by contacting Broadcast Electronics - Marti Facility.

10.1 DP5501A/5503A Transmitter

The Broadcast Electronics DSTL transmitter unit can be divided into two sections. One is the group of signal path modules and the other is the group of support modules.

The **signal path modules** transform the audio inputs into a data modulated microwave RF signal that is suitable for STL transmission purposes. The modules contained in this group are:

Cat. No. 452/472 (A/D Converter)
Cat. No. 453/473 (Audio Encoder)
Cat. No. 454/474 (Modulator)
Cat. No. 476 (Frequency Synthesizer/RF Power Amplifier)

The **support modules** supply the electrical power and provide alarm, control, and metering function for the rest of the unit. The modules contained in this group are:

Cat. No. 457 (Power Supply)
Cat. No. 458 (Alarm/Control)
Front sub-panel & meter assembly

10.1.1 Signal Path Modules Group

The system description of the signal path group is discussed in the direction of the signal flow.

Cat. No. 452 A/D Converter (DP5501A Only)

The signal processing begins when four analog audio signals from barrier strips TB110 and TB111 are accepted, gain adjusted, and A/D converted by the Cat. No. 452 module. This results in a main path data stream of 2 channels at 44.1 kHz sample rate with 16-bit word length PCM, an auxiliary data stream at a 16.5375 kHz sample rate with 16-bit word length PCM, and a voice data stream at a 8.26875 kHz sample rate with voice grade 8 bit μ-law representation. The converter clock signals are supplied from the Cat. No. 453 and the data signals are conveyed to that module. The detection of peak audio levels for the four channels is performed and sent as a DC level to the Cat. No. 458 module. A headphone monitor circuit also allows for listening to each of the input channels, just prior to analog to digital conversion.

Cat. No. 472 A/D Converter (DP5503A Only)

The signal processing begins when four analog audio signals from barrier strips TB110 and TB111 are accepted, gain adjusted, and A/D converted by the Cat. No. 472 module. This results in a main path data stream of 2 stereo pairs (4 channels) at 44.1 kHz sample rate with 16-bit word length PCM. The converter clock signals are supplied from the Cat. No. 473 and the data signals are conveyed to that module. The detection of peak audio levels for the four channels is performed and sent as a DC level to the Cat. No. 458 module. A headphone monitor circuit also allows for listening to either stereo pair, just prior to analog to digital conversion.

Cat. No. 453 Audio Encoder (DP5501A Only)

The Cat. No. 453 Audio Encoder module is the audio processing core of the DP5501A DSTL transmitter. It is designed to accept 3 data streams of audio data from the Cat. No. 452 A/D Converter module and convert them into a single data stream representing the combination of two channels of wide-band (15 kHz) stereo program, a single 7 kHz auxiliary channel (SCA), and a 3 kHz voice grade channel. Data for the auxiliary and voice channels is routed to the first DSP chip where it is converted into a coded representation having a data rate of 66.15 kbits/s for the auxiliary channel and 33.075 kbits/s for the voice grade channel. The data rate reduction is accomplished by the use of the G.721 standard voice compression algorithm for the voice channel and by transform coding techniques for the auxiliary and main channels. The auxiliary channel can also be configured to a "modem" mode, supplying standard digital voice service of 8-bit μ-law with a sample rate of 8.26875 kHz.

In a similar manner, data from the two wide-band channels are routed to a second DSP chip, where the corresponding data rate is reduced to 187.425 kbits/s per channel. Coded data from first DSP chip is passed to the second DSP chip, where it is combined with the 15 kHz data into a single formatted serial bit stream at 485.1 kbits/s, which in turn is sent to the Cat. No. 454 modulator module. The Cat. No. 453 module supplies the conversion and serial bit stream clocks via a PLL circuit working at 33.8688 MHz, which is locked to a reference crystal oscillator working at 15.5232 MHz. A module status determining circuit is included for trouble-shooting purposes.

Cat. No. 473 Audio Encoder (DP5503A Only)

The Cat. No. 473 Audio Encoder module is the audio processing core of the DP5503A DSTL transmitter. It is designed to accept 2 data streams representing 2 pairs of wideband (15 kHz) stereo program channels from the Cat. No. 472 A/D Converter module, and 2 channels of RS-232 serial data from a pair of 9 pin D connectors on the rear panel of the DP5503A. Data for one pair of audio channels is routed to a 24-bit DSP chip where it is converted into a coded representation having a data rate of 187.425 kbits/s per channel. One channel of RS-232 serial data is also sent to the DSP chip, where it is combined with the audio data into a single formatted serial bit stream at 396.9 kbits/s.

In the same manner, data from the remaining pair of program channels is sent to a second DSP chip, where it is combined with the second data channel into a similar bit stream. The bit streams from the two processors are then multiplexed into a single bit stream at a rate of 793.8 kbits/s, which in turn is sent to the Cat. No. 474 Modulator module. The Cat. No. 473 module supplies the conversion and serial bit stream clocks via a PLL circuit working at 11.2896 MHz, which is locked to a reference crystal oscillator working at 12.7008 MHz. A module status determining circuit is included for trouble-shooting purposes.

Cat. No. 454/474 Modulator

In the DP5501A, the Cat. No. 454 Modulator module accepts the 485.1 kbits/s data and clock output of the Cat. No. 453 Audio Encoder module. In the DP5503A, the Cat. No. 474 Modulator module accepts the 793.8 kbits/s data and clock output of the Cat. No. 473 Audio Encoder module. Alternately, either module can accept external RS-422 repeater inputs from rear panel connector J107. This data is split into two parallel paths (242.55 kbits/s in the Cat. No.454, 396.9 kbits/s in the Cat. No. 474), which are encoded for lock-state invariance and partial response signaling (PRS). The PRS-encoded data is fed to in-phase (I channel) and quadrature (Q channel) digital finite-impulse response (FIR) filters, which limit the bandwidth occupied by the transmitted data spectrum. After D/A conversion, the I and Q channel outputs feed two analog roofing filters to remove aliasing products generated by the FIR filter, and are sent to the Cat. No. 455 Transmit Frequency Synthesizer module. A module status-determining circuit is included for trouble-shooting purposes.

Cat. No. 476 Frequency Synthesizer/RF Power Amplifier

The Cat. No. 476 Frequency Synthesizer/RF power amplifier accepts the two digital modulation channels, I and Q, from the Cat. No. 454/474. These two digital channels are applied to a Q+I modulator. The modulator frequency is determined by a microprocessor controlled phase-lock-loop frequency synthesizer circuit. The circuit is programmed for the correct STL frequency by three rotary switches.

The signal from the Q+I modulator is applied to an RF gain stage and is attenuated by a variable RF attenuator circuit to provide the desired RF output level. This output level is adjustable between 1/4 and 2 watts by the RF LEVEL ADJUST potentiometer located on the front of the module. The signal then enters the ultra low IM distortion solid state power amplifier module which provides the required power gain. The module provides forward and reverse power detectors that send a DC signal to the Cat. No. 458 Alarm Control module for meter output. The RF output is switched on via a control line from the Cat. No. 458 module. The on board microprocessor monitors the status of several parameters for trouble-shooting purposes. The Cat. No. 476 circuitry can be grouped as follows: RF reference oscillator, a microprocessor controlled phase-lock-loop (PLL), frequency programming logic, RF modulator, a DC regulator, an RF power amplifier circuit, and forward/reflected power directional coupler circuits.

10.1.2 Support Modules Group

Cat. No. 457 Power Supply

The Cat. No. 457 Power Supply module generates four regulated DC voltages from the AC line. The supply is a conventional capacitor input filter type and is completely contained within the module itself (from the AC power entry module to regulated voltages output through the back plane connector). The supply is a high efficiency design achieved by the use of Schottky rectifiers and low drop-out voltage regulators. The voltages supplied are: +5 Volts at 2.0 Amps, +10 Volts at 3.3 Amps, and +/-15 Volts at 0.8 Amps each.

Cat. No. 458 Alarm/Control

The Cat. No. 458 Alarm/Control module acts as the control and monitoring center for the DSTL transmitter. It monitors the modules that provide status information to provide a Summary Alarm indication, and it controls whether the unit can be put into an operate status. This module is the interface between the front panel controls and the rest of the unit. There are four functions provided by this module:

The **alarm section** accepts information from other modules in the unit to provide summary alarm and operation status information.

The **control section** provides control signals to the Power Amplifier module. Control is modified by the status of modules, external remote control line, and the position of the **MODE** switch on the front panel. This also includes buffering of the "operate OK" line from the Cat. No. 456 Power Amplifier module.

The **meter section** selects the input to the front panel meter from 1 of 7 inputs and preconditions the signal to improve the peak response of the meter.

The **power supply monitor section** monitors the power supply voltages in the unit for both average level and the presence of ripple to provide the alarm function for the Cat. No. 457 Power Supply module.

10.2 DP5502/5504 Receiver

The Broadcast Electronics DSTL receiver can be divided into two sections. One is the group of signal path modules and the other is the group of support modules.

The **signal path modules** transform the received RF signal into 4 analog audio outputs and a composite signal. The modules contained in this group are:

Cat. No. 466/486 (Receiver)
Cat. No. 464/484 (Demodulator)
Cat. No. 463/483 (Audio Decoder)
Cat. No. 462/482 (D/A converter)
Cat. No. 460 (Digital Stereo Generator)

The **support modules** supply the electrical power and provide alarm, control, and metering function for the rest of the unit. The modules contained in this group are:

Cat. No. 457 (Power Supply)
Cat. No. 458 (Alarm/Control)
Front Sub-panel & Meter Assembly.

10.2.1 Signal Path Modules Group

The system description of the signal path group is discussed in the direction of the signal flow.

Cat. No. 466/486 Receiver

The Cat. No. 466/486 receiver module circuitry converts the RF signal from the antenna input, amplifies it, and frequency down-converts into a 70 MHz modulated signal suitable for demodulation by the Cat. No. 464/484 module. The Cat. No. 466/486 module includes the following sections: a RF preselector and amplifier, a frequency synthesizer (local oscillator) and mixer, an IF amplifier and AGC control, a DC regulator, and a module status-determining circuit, included for troubleshooting purposes. A DC level representing the RF received signal level is derived from the action of the AGC circuitry and sent to the meter selector section of the Cat. No. 458 Alarm/Control module.

The DSTL antenna signal is coupled through a four-pole ceramic block filter to the input matching network of the low-noise RF preamplifier. The preamplifier gain is provided by a dual gate GaAs FET. The preamplifier signal is then further filtered by an additional two-pole, ceramic block filter prior to being coupled to the down-converter (mixer). The down converted 70 MHz signal is then amplified with AGC control and sent to the Cat. No. 464/484 module.

The frequency synthesizer used as the local oscillator in the Cat. No. 466 is very similar to the synthesizer used in the Cat. No. 455 module. The correct STL frequency is determined by the phase lock loop within the Cat. No. 466 module.

Cat. No. 464/484 Demodulator

The Cat. No. 464/484 Demodulator module accepts the 70 MHz output of the Cat. No. 466/486 receiver module and down-converts this to two I- and Q-channel base-band signals. The clock and data contained in these PRS coded signals is regenerated and interleaved into a single 485.1 kbits/s (Cat. No. 464) or 793.8 kbits/s (Cat. No. 484) bit stream and sent to the Cat. No. 463/483 Audio Decoder module. This module contains the following components: input mixers and base-band filters, analog-to-digital converters and A/D bias circuits, a 70 MHz quadrature carrier generator and mixer drivers, a clock generator, recovery loop, and A/D timing loops, adaptive carrier phase and A/D bias control loops, an error rate estimator (pseudo-error monitor), error correction circuitry exploiting the inherent redundancy of PRS, a differential decoder and data interleaver, and a module status-determining circuit, included for trouble-shooting purposes. This module also sends a DC level representing the pseudo-error rate to the Cat. No. 458 module. The module also feeds an RS-422 output (J107) on the rear panel for external repeater use.

Cat. No. 463 Audio Decoder (DP5502 Only)

The Cat. No. 463 Audio Decoder module is the audio processing section that converts the single data stream containing the bit-rate reduction encoded data into standard PCM and digital voice formats. It is designed to accept a formatted serial bit stream and clock at 485.1 kbits/sec from the Cat. No. 464 Demodulator module. This bit stream represents the four encoded audio channels: two main channels of wide-band (15 kHz) stereo program, a single 7 kHz auxiliary channel (SCA), and a 3 kHz voice-grade channel. The auxiliary channel can also be configured to a "modem" mode supplying standard digital voice service of 8-bit μ-law with a sample rate of 8.26875 kHz.

This data is routed to the first DSP chip, where it is de-multiplexed into individual channels. The conversion and timing clocks for the Cat. No. 462 D/A Converter module and Cat. No. 460 Digital Stereo Generator module are generated from a 33.8688 MHz PLL that is locked to the incoming 485.1 kbits/sec data clock. Data representing the main channels are decoded into PCM audio by the first DSP chip, while encoded data for the auxiliary and voice channels is passed to a second DSP chip where it is decoded in a similar manner. The resulting audio data from both processors is then sent to the Cat. No. 462 D/A Converter module and the Cat. No. 460 Digital Stereo Generator module. A front panel switch selects either the normal or test mode of operation. In the test mode, encoded calibration data from memory is substituted for the serial output of the Cat. No. 464 to provide a calibrated output level for the main and auxiliary audio channels. A module status-determining circuit is included for trouble-shooting purposes.

Cat. No. 483 Audio Decoder (DP5504 Only)

The Cat. No. 483 Audio Decoder module is the audio processing section that converts the single data stream containing the bit-rate reduced audio data into standard PCM. It is designed to accept a formatted serial bit stream and clock at 793.8 kbits/sec from the Cat. No. 484 Demodulator module. This bit stream represents two pairs of wide-band (15 kHz) stereo program channels and two channels of RS-232 serial data.

This data is first demultiplexed into two 396.9 kbits/s data streams, each of which represents a stereo program pair and one of the data channels. Each of these data streams is routed to a separate 24-bit DSP chip, where they are decoded into PCM audio and RS-232 serial data. The resulting audio data from both processors is then sent to the Cat. No. 482 D/A Converter module and the Cat. No. 460 Digital Stereo Generator; the two RS-232 data channels are routed to a pair of 9 pin D connectors on the rear panel of the DP5504.

The conversion and timing clocks for the Cat. No. 482 D/A Converter module and Cat. No. 460 Digital Stereo Generator module are generated from a 11.2896 MHz PLL that is locked to the incoming 793.8 kbits/sec data clock. A front panel switch selects either the normal or test mode of operation. In the test mode, encoded calibration data from memory is substituted for the serial output of the Cat. No. 484 to provide a calibrated output level for the four audio program channels. A module status-determining circuit is included for trouble-shooting purposes.

Cat. No. 462 D/A Converter (DP5502 Only)

The Cat. No. 462 D/A Converter module accepts clock and data signals and provides four differential audio outputs with adjustable levels, available at the rear panel barrier strips TB110 and TB111. There are two high quality main channels using 16 bit stereo delta-sigma conversion at a 44.1 kHz sample rate, a medium quality auxiliary channel using 16 bit conversion at a 16.5375 kHz (auxiliary mode) or 8.26875 kHz (modem mode) sample rate, and a lower quality voice channel using an 8 bit μ-law codec at an 8.26875 kHz sample rate. The converter clock signals are supplied from the Cat. No. 463. The detection of peak audio levels for the four channels, before level adjustment, is performed in this module and sent as DC levels to the Cat. No. 458 module. A headphone monitor circuit also allows for listening to each of the output channels, just prior to level adjustment and the differential output circuitry. The output signals are routed through an operate/standby relay to the rear panel barrier strips. An operate enable line from the Cat. No. 458 controls this relay.

Cat. No. 482 D/A Converter (DP5504 Only)

The Cat. No. 482 D/A Converter module accepts clock and data signals and provides four differential audio outputs with adjustable levels, available at the rear panel barrier strips TB110 and TB111. There are two stereo pairs (4 channels) using 16 bit stereo delta-sigma conversion at a 44.1 kHz sample rate. The converter clock signals are supplied from the Cat. No. 483. The detection of peak audio levels for the four channels, before level adjustment, is performed in this module and sent as DC levels to the Cat. No. 458 module. A headphone monitor circuit also allows for listening to either stereo pair, just prior to level adjustment and the differential output circuitry. The output signals are routed through an operate/standby relay to the rear panel barrier strips. An operate enable line from the Cat. No. 458 controls this relay.

Cat. No. 460 Digital Stereo Generator

The Cat. No. 460 Digital Stereo Generator module is designed to produce an analog base-band stereo multiplex signal using the main channel (left and right) PCM audio data from the Cat. No. 463/483 Audio Decoder module. A single DSP chip performs the multiplex function in the digital domain, producing a composite PCM output signal at a sample rate of 176.4 kHz. The DSP chip allows control of the relative levels of the L-R and pilot signals, selection of mono or stereo operation, and built-in calibration and test operations. The composite output signal is converted to analog form by a 16-bit D/A converter, followed by a precision reconstruction filter. The resulting signal is then buffered and routed through an operate/standby relay to the rear panel composite output connector. An operate enable line from the Cat. No. 458 controls this relay. A module status determining circuit is included for trouble-shooting purposes.

10.2.2 Support Modules Group

Cat. No. 457 Power Supply

The Cat. No. 457 Power Supply module generates four regulated DC voltages from the AC line and is identical to the one used in the DSTL Transmitter. The supply is a conventional capacitor input filter type and is completely contained within the module itself (from the AC power entry module to regulated voltages output through the back plane connector). The supply is a high efficiency design achieved by the use of Schottky rectifiers and low drop-out voltage regulators. The voltages supplied are: +5 Volts at 2.0 Amps, +10 Volts (not used), and +/-15 Volts at 0.8 Amps each.

Cat. No. 458 Alarm/Control

The Cat. No. 458 Alarm/Control module acts as the control and monitoring center for the DSTL receiver. It monitors the modules that provide status information to provide a summary alarm indication and controls whether the unit can be put into an operate status. This module is the interface between the front panel controls and the rest of the unit. There are four functions provided by this module:

The **alarm section** accepts information from other modules in the unit to provide summary alarm and operation status information.

The **control section** provides control signals to the output relays on the Cat. No. 462/482 D/A and Cat. No. 460 Digital Stereo Generator modules. Control is modified by the status of modules, external remote control line, and the position of the mode switch on the front panel. This section also includes buffering of:

a) Mono mode control line -- From the rear panel, through the Cat. No. 458, to the Cat. No. 460.

b) Mono/Stereo status verification line -- From the Cat. No. 460 through the Cat. No. 458, to the front panel MONO yellow LED.

c) EXT operate OK line -- From the Cat. No. 456 through the Cat. No. 458, to the rear panel hot standby connector.

The **meter section** selects the input to the front panel meter from 1 of 7 inputs and preconditions the signal to improve the peak response of the meter.

The **power supply monitor section** monitors the power supply voltages in the unit for both average level and the presence of ripple to provide the alarm function for the Cat. No. 457 Power Supply module.

10.3 Backplane

The Backplane provides all of the power and signal interconnections between the various modules and external devices. In addition, it contains broadband electromagnetic filtering on all external input and output terminals. Jumpers for composite source selection in the DP5504 are also located on the backplane (see Section 3.4.2).

11.1 Repair / Exchange Order (RXO) Program

Due to the modular design of the DSTL, it is extremely easy to exchange defective or problematic modules with working modules supplied by Broadcast Electronics - Marti Facility. We call this exchange service the Repair/Exchange (RXO) program. The program is adminstered only by Broadcast Electronics. In emergency conditions, we can help you get through a catastrophic failure with a loan unit. Loan units are provided on an as-available basis.

When you request a replacement module from Broadcast Electronics, it will be processed as an *Advance Replacement.* When appropriate, we also process RXO transactions as *Exchange* or *Repair & Return.*

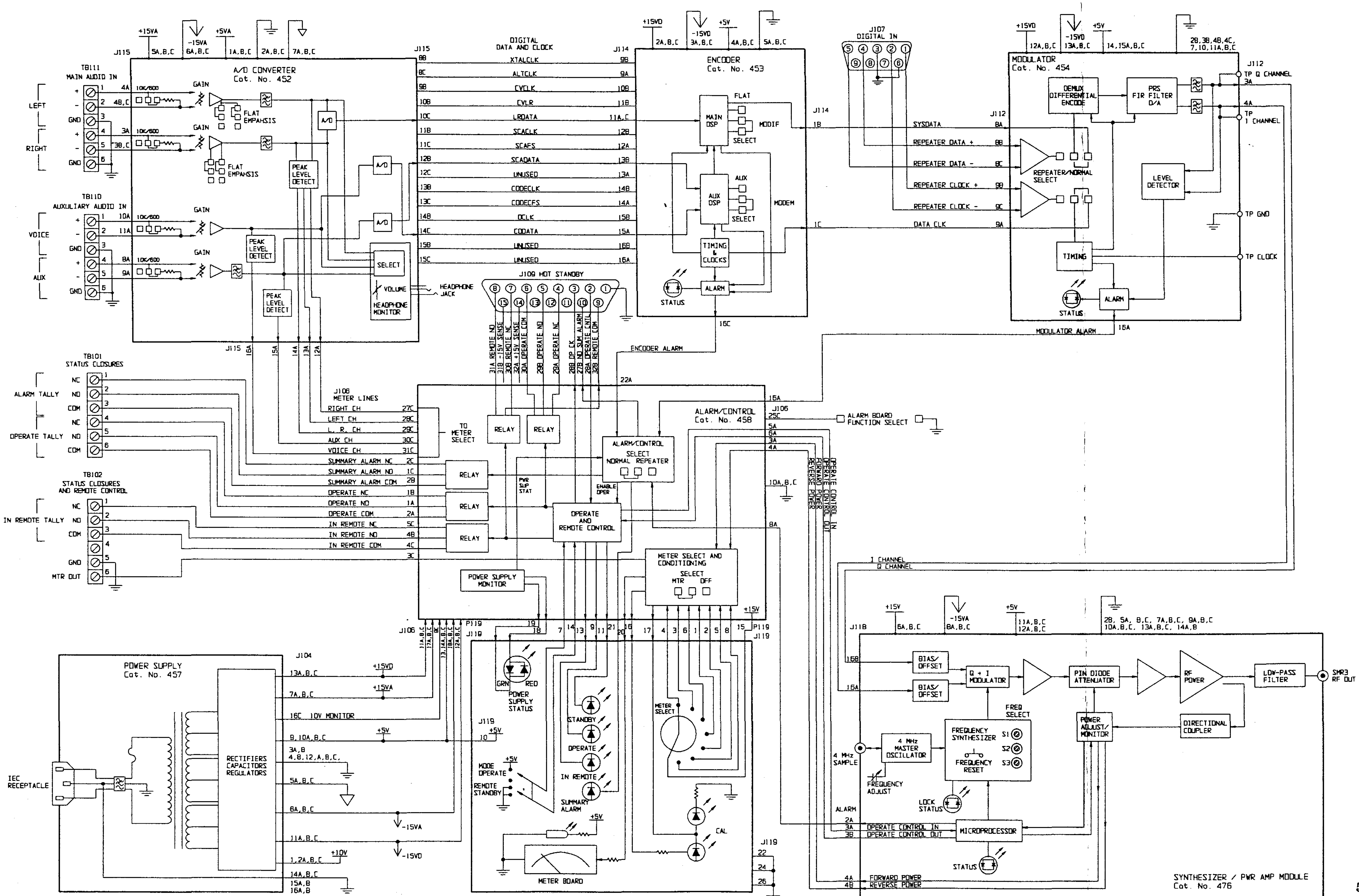
Advance Replacement

In an emergency, all modules available through the RXO program can be sent to you in advance of receiving your defective one. To assure prompt attention, please call with the **model number** and **serial number** of the unit, and the **Cat. No.** of the module that you wish to replace. We will send out the replacement module (freight prepaid, for warranty items) via overnight mail or, if you wish, counter-to-counter service.

Of course, timely return of the defective module is expected. Send freight prepaid in a timely manner to Broadcast Electronics - Marti Facility, 421 Marti Drive, Box 661, Cleburne, Texas 76033 **Attn.: RXO Program**. Use the same packing materials that were used to send the replacement module. Included with the good module is the ***RXO SHIP TO*** sheet. Please return the bottom portion of the sheet along with the defective module. There is a space for you to describe the problem. If the defective module is not returned within five weeks of the ship date of the replacement module, then we will bill you for it.

Exchange

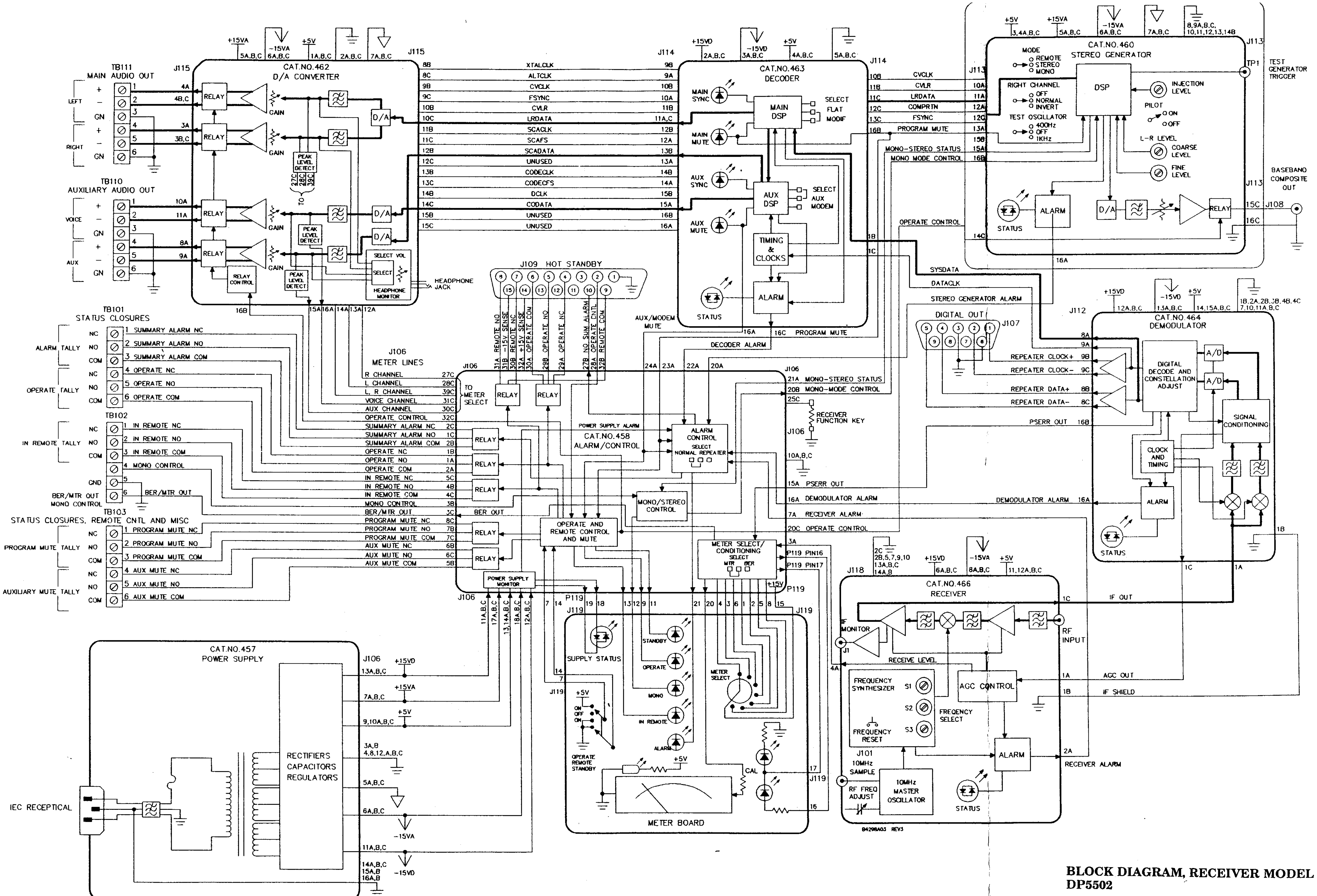
RXO transactions can also be processed as *Exchanges*. You send your defective module (freight prepaid) to Broadcast Electronics - Marti Facility. In exchange for it, we will send out a good module (freight prepaid, for warranty items). We typically send out the good module the day after receiving the defective one. After we repair that module it will enter our RXO inventory for future exchange transactions. The one sent to you is yours to keep. When possible, we try to exchange modules with similar serial numbers.



597-8100-9

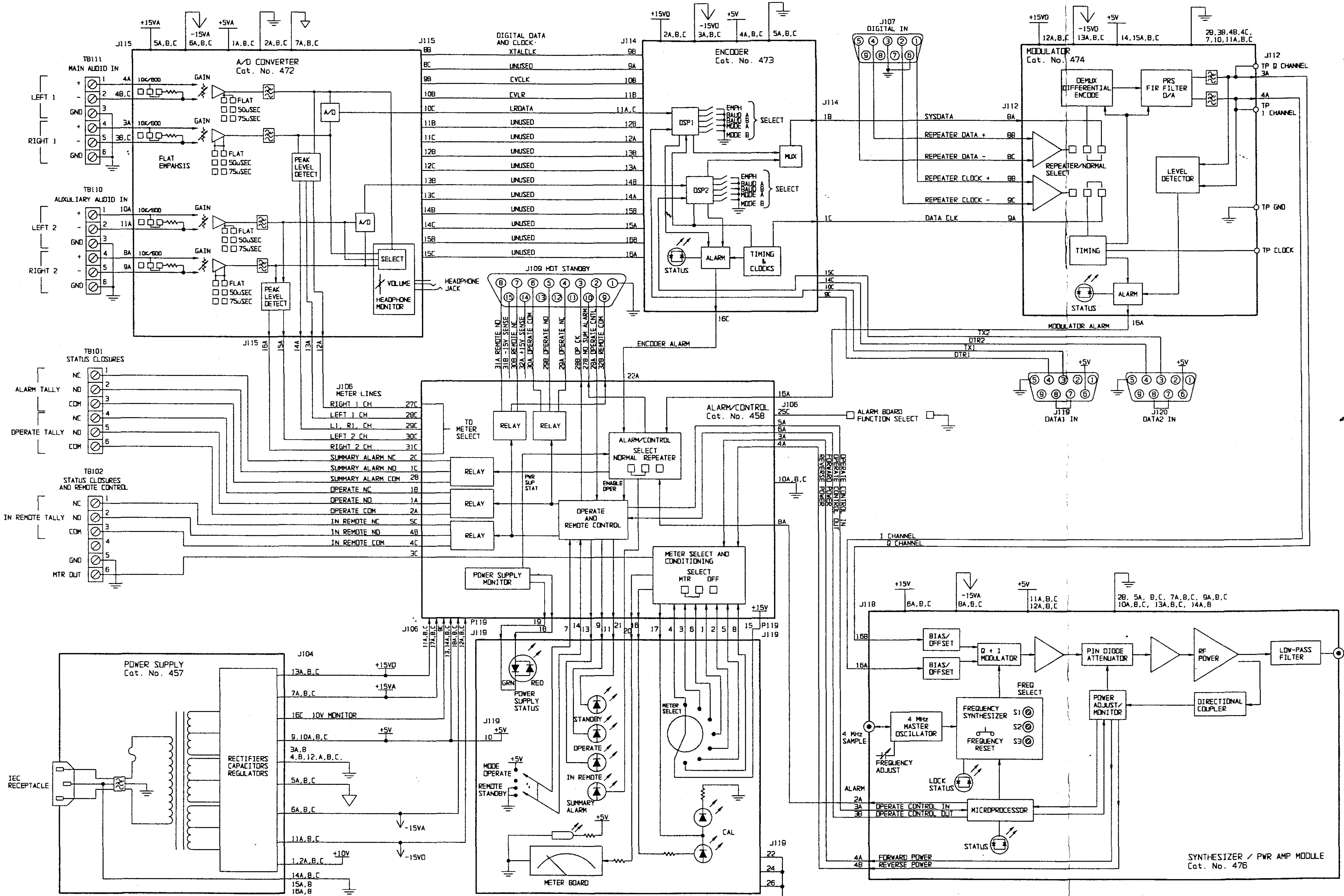
BLOCK DIAGRAM, TRANSMITTER MODEL DP5501A

(10-11/10-12)



BLOCK DIAGRAM, RECEIVER MODEL DP5502

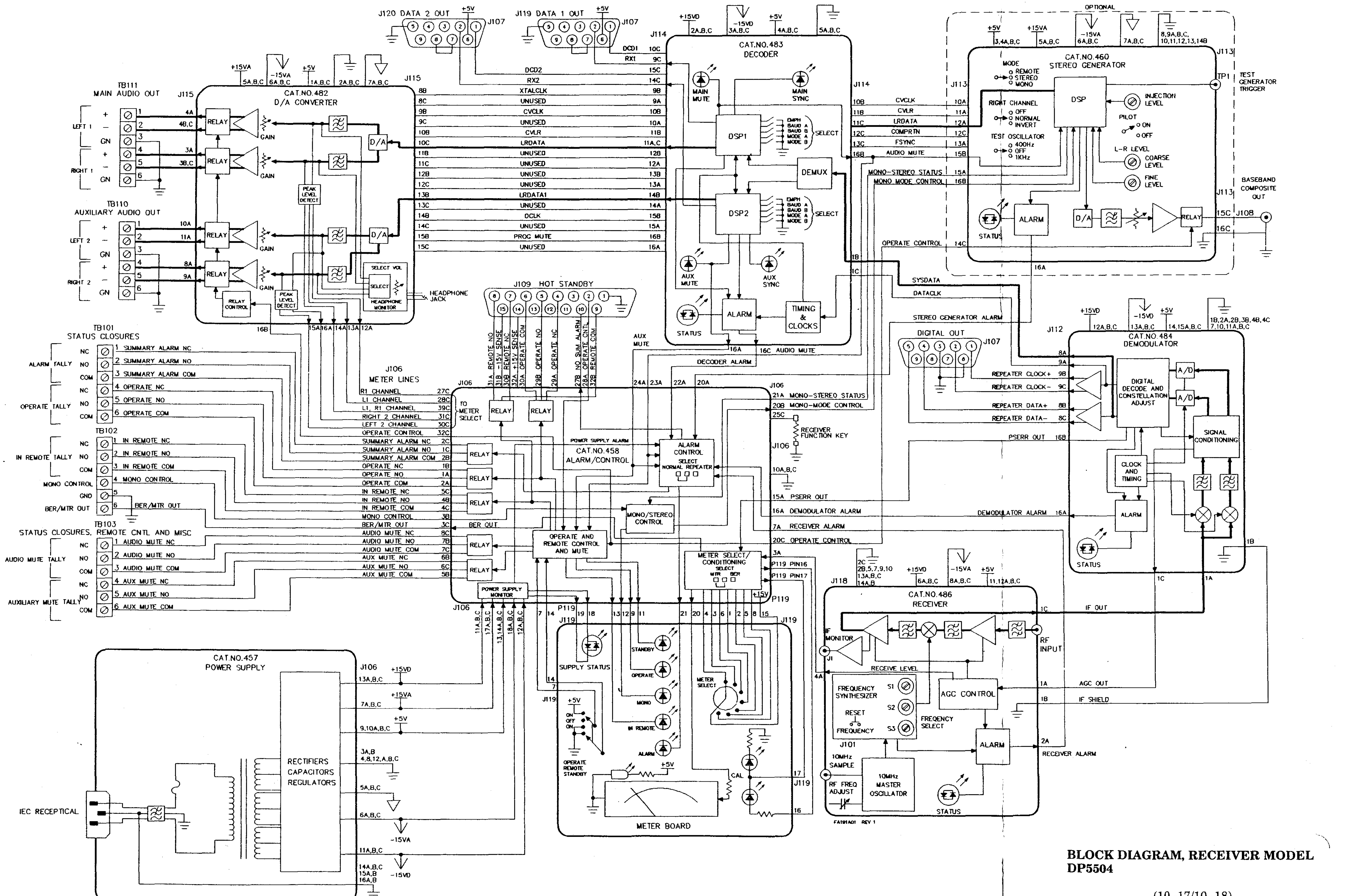
(10–13/10–14)



597–8100–10

BLOCK DIAGRAM, TRANSMITTER MODEL DP5503A

(10–15/10–16)



BLOCK DIAGRAM, RECEIVER MODEL DP5504

(10–17/10–18)

Repair & Return

When we receive obsolete items, like frames or seriously damaged equipment, we process them on a *Repair & Return* basis. The item is returned to you after repair. All repair work is under warranty for one year from date of shipment or for two years from the system purchase date, whichever is longer. Under emergency situations, we can help you get through a catastrophic failure with a loan unit.

Return Authorization

At present, no prior authorization is required to return defective items to Broadcast Electronics. When a failure involves a complete unit, however, we do request that you contact us first. We are always glad to help you localize your problem to a particular module before you send it in. We request that you include a description of the problem to help us expedite repairs, along with your address, phone number and a reference or PO number.

Warranty

All modules exchanged through the RXO program are under warranty for one year from date of shipment or for two years from the DSTL System purchase date, whichever is longer.

11.2 Field Repairs

We understand that for expediency, customers often prefer to attempt field repairs. Under certain conditions we will assist in the repair of items in the field with telephone support or by mailing parts. However, since we are not in a position to assess the troubleshooting and repair skills of our callers, a poor repair job may void the warranty. Such attempts may end up costing more and taking more time to resolve than if a module had been exchanged. Therefore, please take advantage of our RXO program.

CAUTION In order to avoid serious electrical shock or fire do not attempt to service this product unless you are qualified to do so.

11.3 Service Parts List

The following lists part numbers for items which are difficult to obtain locally and that are easily replaceable. They can be ordered directly from Broadcast Electronics - Marti Facility.

Description	Part Number	Manufacturer
Cat. No. 452/472		
Input level pot	12015	Bourns 3006P-1-103
3 position toggle switch (452)	52069	C&K 7203-M-D9-A-B-E
2 position toggle switch (472)	52145	C&K 7201-M-D9-A-B-E
Volume pot	12087	Clarostat P1097
Headphone jack	71205	Rean M203-03
Switch knob	64061	Rean C105-0F
Cap knob	64062	Rean T100-OF
Cat. No. 453/473		
Red/green LED	32047	Dialight 550-3006
Cat. No. 454/474		
2 position toggle switch	52073	Switchcraft J1010SD9C2QE
Yellow LED	32012	Dialight 550-0306
Red/green LED	32047	Dialight 550-3006
Cat. No. 476		
RF connector cable	83038	Broadcast Electronics
Red/green LED	32047	Dialight 550-3006
N connector w/ nut and washer	70136	CDI 8011-2SF
N connector nut	61225	Nut 5/8" Hex Thin
N connector washer	61224	Washer 5/8 Internal Star
Cat. No. 457		
Retention bracket	65140	Broadcast Electronics
Retention bracket screw	60096	M3x6mm pan pozidrive
Retention bracket washer	61091	M3 crinkle washer

Description	Part Number	Manufacturer
Cat. No. 462/482		
Output level pot	12015	Bourns 3006P-1-103
3 position toggle switch (462)	52069	C&K 7203-M-D9-A-B-E
2 position toggle switch (482)	52145	C&K 7201-M-D9-A-B-E
Volume pot	12087	Clarostat P1097
Headphone jack	71205	Rean M203-03
Switch knob	64061	Rean C105-0F
Cap knob	64062	Rean T100-OF
Cat. No. 463/483		
Red/green LED	32047	Dialight 550-3006
Yellow LED	32012	Dialight 550-0306
Green LED	32017	Dialight 550-0206
2 position switch	52073	Switchcraft J1010SD9C2QE
Cat. No. 464/484		
Red/green LED	32047	Dialight 550-3006
Cat. No. 466/486		
Red/green LED	32047	Dialight 550-3006
Cat. No. 460		
3 position switch	52127	C&K 7103-M-D9-A-B-E
2 position switch	52126	C&K 7101-M-D9-A-B-E
Hex switch	52123	Ecco 331035GS
Composite output pot	12015	Bourns 3006P-1-103
Meter Panel Bulkhead		
Bulkhead screws	60096	M3x6 Pan Posi
Bulkhead washers	61091	M3 Curved 6mm OD.
3 position toggle switch	52134	C&K 7203-M-D9-V3-B-E
6 position meter switch	52136	NKK MRA112
Switch knob	64063	Rean W215-OF
Cap knob	64064	Rean T210-OF
Green LED	32040	Dialight 521-9250
Yellow LED	32041	Dialight 521-9248
Red/green LED	32050	Dialight 521-9177

Description	Part Number	Manufacturer
Other		
Toggle switch covers	52135	C&K 7062-02-000
Jumpers	74087	Samtec SNT 100-BK-G
Back Panel 9-way Connector	70100	Cinch DE 9P
Connector Housing	75050	ITT Cannon DE121073-
154		
Back Panel 15 pin Connector	70099	Cinch DA 15P
Connector Housing	75053	Molex 82007-0131
Screws for barrier strips	60013	6-32x3/16 Slot Bind Hd
Front panel	DP5500-FP	Broadcast Electronics
Front panel handles (single and loop)	64069	Broadcast Electronics
Tweaker - transmitter	92002	Spectrol 8T000
Tweaker - receiver	92018	Johanson 8777
Tweaker clips	65143	Richco KKU-4-RT
Rack ears	63400	Broadcast Electronics
Repeater connector cable, 9 pin	83246	Belden 49900
Repeater comm cable, 15 pin	83247	Broadcast Electronics
110 Volt systems:		
Fuse 20mm (T2A) Transmitter	56027	Littelfuse 213.2.0
Fuse 20mm (T1A) Receiver	56016	Littelfuse 213.1.0
Power cord	92003	Belden 17250
240 Volt systems:		
Fuse 20mm (T1A) Transmitter	56016	Littelfuse 213.1.0
Fuse 20mm (T630mA) Receiver	56041	Littelfuse 213.630
Power cord	92004	Belden 17254

Models DP5501A And DP5503A Transmitter

Using the table in below, set the transmitter frequencies to your operating frequency, if the factory or distributor has not already done so. You may want to jot down the frequency and hex code in the space provided on the front panel.

Note: The frequency will not change until power has been re-cycled or the frequency **RESET** switch, has been pressed.

Frequency	S1	S2	S3
944.000	4	8	0
944.025	4	8	1
944.050	4	8	2
944.075	4	8	3
944.100	4	8	4
944.125	4	8	5
944.150	4	8	6
944.175	4	8	7
944.200	4	8	8
944.225	4	8	9
944.250	4	8	A
944.275	4	8	B
944.300	4	8	C
944.325	4	8	D
944.350	4	8	E
944.375	4	8	F
944.400	4	9	0
944.425	4	9	1
944.450	4	9	2
944.475	4	9	3
944.500	4	9	4
944.525	4	9	5
944.550	4	9	6
944.575	4	9	7
944.600	4	9	8
944.625	4	9	9
944.650	4	9	A
944.675	4	9	B
944.700	4	9	C
944.725	4	9	D
944.750	4	9	E
944.775	4	9	F

Frequency	S1	S2	S3
944.800	4	A	0
944.825	4	A	1
944.850	4	A	2
944.875	4	A	3
944.900	4	A	4
944.925	4	A	5
944.950	4	A	6
944.975	4	A	7
945.000	4	A	8
945.025	4	A	9
945.050	4	A	A
945.075	4	A	B
945.100	4	A	C
945.125	4	A	D
945.150	4	A	E
945.175	4	A	F
945.200	4	B	0
945.225	4	B	1
945.250	4	B	2
945.275	4	B	3
945.300	4	B	4
945.325	4	B	5
945.350	4	B	6
945.375	4	B	7
945.400	4	B	8
945.425	4	B	9
945.450	4	B	A
945.475	4	B	B
945.500	4	B	C
945.525	4	B	D
945.550	4	B	E
945.575	4	B	F

Appendix A-2

Transmitter

Frequency	S1	S2	S3
945.600	4	C	0
945.625	4	C	1
945.650	4	C	2
945.675	4	C	3
945.700	4	C	4
945.725	4	C	5
945.750	4	C	6
945.775	4	C	7
945.800	4	C	8
945.825	4	C	9
945.850	4	C	A
945.875	4	C	B
945.900	4	C	C
945.925	4	C	D
945.950	4	C	E
945.975	4	C	F
946.000	4	D	0
946.025	4	D	1
946.050	4	D	2
946.075	4	D	3
946.100	4	D	4
946.125	4	D	5
946.150	4	D	6
946.175	4	D	7
946.200	4	D	8
946.225	4	D	9
946.250	4	D	A
946.275	4	D	B
946.300	4	D	C
946.325	4	D	D
946.350	4	D	E
946.375	4	D	F
946.400	4	E	0
946.425	4	E	1
946.450	4	E	2
946.475	4	E	3
946.500	4	E	4
946.525	4	E	5
946.550	4	E	6
946.575	4	E	7
946.600	4	E	8

Frequency	S1	S2	S3
946.625	4	E	9
946.650	4	E	A
946.675	4	E	B
946.700	4	E	C
946.725	4	E	D
946.750	4	E	E
946.775	4	E	F
946.800	4	F	0
946.825	4	F	1
946.850	4	F	2
946.875	4	F	3
946.900	4	F	4
946.925	4	F	5
946.950	4	F	6
946.975	4	F	7
947.000	4	F	8
947.025	4	F	9
947.050	4	F	A
947.075	4	F	B
947.100	4	F	C
947.125	4	F	D
947.150	4	F	E
947.175	4	F	F
947.200	5	0	0
947.225	5	0	1
947.250	5	0	2
947.275	5	0	3
947.300	5	0	4
947.325	5	0	5
947.350	5	0	6
947.375	5	0	7
947.400	5	0	8
947.425	5	0	9
947.450	5	0	A
947.475	5	0	B
947.500	5	0	C
947.525	5	0	D
947.550	5	0	E
947.575	5	0	F
947.600	5	1	0
947.625	5	1	1

Appendix A-3

Transmitter

Frequency	S1	S2	S3
947.650	5	1	2
947.675	5	1	3
947.700	5	1	4
947.725	5	1	5
947.750	5	1	6
947.775	5	1	7
947.800	5	1	8
947.825	5	1	9
947.850	5	1	A
947.875	5	1	B
947.900	5	1	C
947.925	5	1	D
947.950	5	1	E
947.975	5	1	F
948.000	5	2	0
948.025	5	2	1
948.050	5	2	2
948.075	5	2	3
948.100	5	2	4
948.125	5	2	5
948.150	5	2	6
948.175	5	2	7
948.200	5	2	8
948.225	5	2	9
948.250	5	2	A
948.275	5	2	B
948.300	5	2	C
948.325	5	2	D
948.350	5	2	E
948.375	5	2	F
948.400	5	3	0
948.425	5	3	1
948.450	5	3	2
948.475	5	3	3
948.500	5	3	4
948.525	5	3	5
948.550	5	3	6
948.575	5	3	7
948.600	5	3	8
948.625	5	3	9
948.650	5	3	A

Frequency	S1	S2	S3
948.675	5	3	B
948.700	5	3	C
948.725	5	3	D
948.750	5	3	E
948.775	5	3	F
948.800	5	4	0
948.825	5	4	1
948.850	5	4	2
948.875	5	4	3
948. 900	5	4	4
948. 925	5	4	5
948. 950	5	4	6
948. 975	5	4	7
949.000	5	4	8
949.025	5	4	9
949.050	5	4	A
949.075	5	4	B
949.100	5	4	C
949.125	5	4	D
949.150	5	4	E
949.175	5	4	F
949.200	5	5	0
949.225	5	5	1
949.250	5	5	2
949.275	5	5	3
949.300	5	5	4
949.325	5	5	5
949.350	5	5	6
949.375	5	5	7
949.400	5	5	8
949.425	5	5	9
949.450	5	5	A
949.475	5	5	B
949.500	5	5	C
949.525	5	5	D
949.550	5	5	E
949.575	5	5	F
949.600	5	6	0
949.625	5	6	1
949.650	5	6	2
949.675	5	6	3

Appendix A-4

Transmitter

Frequency	S1	S2	S3
949.700	5	6	4
949.725	5	6	5
949.750	5	6	6
949.775	5	6	7
949.800	5	6	8
949.825	5	6	9
949.850	5	6	A
949.875	5	6	B
949.900	5	6	C
949.925	5	6	D
949.950	5	6	E
949.975	5	6	F
950.000	5	7	0
950.025	5	7	1
950.050	5	7	2
950.075	5	7	3
950.100	5	7	4
950.125	5	7	5
950.150	5	7	6
950.175	5	7	7
950.200	5	7	8
950.225	5	7	9
950.250	5	7	A
950.275	5	7	B
950.300	5	7	C
950.325	5	7	D
950.350	5	7	E
950.375	5	7	F
950.400	5	8	0
950.425	5	8	1
950.450	5	8	2
950.475	5	8	3
950.500	5	8	4
950.525	5	8	5
950.550	5	8	6
950.575	5	8	7
950.600	5	8	8
950.625	5	8	9
950.650	5	8	A
950.675	5	8	B
950.700	5	8	C
950.725	5	8	D
950.750	5	8	E
950.775	5	8	F
950.800	5	9	0
950.825	5	9	1
950.850	5	9	2
950.875	5	9	3
950.900	5	9	4
950.925	5	9	5
950.950	5	9	6
950.975	5	9	7
951.000	5	9	8
951.025	5	9	9
951.050	5	9	A
951.075	5	9	B
951.100	5	9	C
951.125	5	9	D
951.150	5	9	E
951.175	5	9	F
951.200	5	A	0
951.225	5	A	1
951.250	5	A	2
951.275	5	A	3
951.300	5	A	4
951.325	5	A	5
951.350	5	A	6
951.375	5	A	7
951.400	5	A	8
951.425	5	A	9
951.450	5	A	A
951.475	5	A	B
951.500	5	A	C
951.525	5	A	D
951.550	5	A	E
951.575	5	A	F
951.600	5	B	0
951.625	5	B	1
951.650	5	B	2
951.675	5	B	3
951.700	5	B	4
951.725	5	B	5

Appendix A-5

Transmitter

Frequency	S1	S2	S3
951.750	5	B	6
951.775	5	B	7
951.800	5	B	8
951.825	5	B	9
951.850	5	B	A
951.875	5	B	B

Frequency	S1	S2	S3
951. 900	5	B	C
951.925	5	B	D
951.950	5	B	E
951.975	5	B	F
952.000	5	C	0

Models DP5502 And DP5504 Receiver

Using the table in the following text, set the receiver frequency to your operating frequency. The frequency is set at the factory. You may want to jot down the frequency and hex code in the space provided on the front panel.

Note: The frequency will not change until power has been re-cycled or the frequency **RESET** switch, has been pressed.

Frequency	S1	S2	S3
944.000	0	7	C
944.025	1	7	C
944.050	2	7	C
944.075	3	7	C
944.100	4	7	C
944.125	5	7	C
944.150	6	7	C
944.175	7	7	C
944.200	8	7	C
944.225	9	7	C
944.250	A	7	C
944.275	B	7	C
944.300	C	7	C
944.325	D	7	C
944.350	E	7	C
944.375	F	7	C
944.400	0	0	D
944.425	1	0	D
944.450	2	0	D
944.475	3	0	D
944.500	4	0	D
944.525	5	0	D
944.550	6	0	D
944.575	7	0	D
944.600	8	0	D
944.625	9	0	D
944.650	A	0	D
944.675	B	0	D
944.700	C	0	D
944.725	D	0	D
944.750	E	0	D
944.775	F	0	D
944.800	0	1	D

Frequency	S1	S2	S3
944.825	1	1	D
944.850	2	1	D
944.875	3	1	D
944.900	4	1	D
944.925	5	1	D
944.950	6	1	D
944.975	7	1	D
945.000	8	1	D
945.025	9	1	D
945.050	A	1	D
945.075	B	1	D
945.100	C	1	D
945.125	D	1	D
945.150	E	1	D
945.175	F	1	D
945.200	0	2	D
945.225	1	2	D
945.250	2	2	D
945.275	3	2	D
945.300	4	2	D
945.325	5	2	D
945.350	6	2	D
945.375	7	2	D
945.400	8	2	D
945.425	9	2	D
945.450	A	2	D
945.475	B	2	D
945.500	C	2	D
945.525	D	2	D
945.550	E	2	D
945.575	F	2	D
945.600	0	3	D
945.625	1	3	D

Appendix A-2

Receiver

Frequency	S1	S2	S3
945.650	2	3	D
945.675	3	3	D
945.700	4	3	D
945.725	5	3	D
945.750	6	3	D
945.775	7	3	D
945.800	8	3	D
945.825	9	3	D
945.850	A	3	D
945.875	B	3	D
945.900	C	3	D
945.925	D	3	D
945.950	E	3	D
945.975	F	3	D
946.000	0	4	D
946.025	1	4	D
946.050	2	4	D
946.075	3	4	D
946.100	4	4	D
946.125	5	4	D
946.150	6	4	D
946.175	7	4	D
946.200	8	4	D
946.225	9	4	D
946.250	A	4	D
946.275	B	4	D
946.300	C	4	D
946.325	D	4	D
946.350	E	4	D
946.375	F	4	D
946.400	0	5	D
946.425	1	5	D
946.450	2	5	D
946.475	3	5	D
946.500	4	5	D
946.525	5	5	D
946.550	6	5	D
946.575	7	5	D
946.600	8	5	D
946.625	9	5	D
946.650	A	5	D

Frequency	S1	S2	S3
946.675	B	5	D
946.700	C	5	D
946.725	D	5	D
946.750	E	5	D
946.775	F	5	D
946.800	0	6	D
946.825	1	6	D
946.850	2	6	D
946.875	3	6	D
946.900	4	6	D
946.925	5	6	D
946.950	6	6	D
946.975	7	6	D
947.000	8	6	D
947.025	9	6	D
947.050	A	6	D
947.075	B	6	D
947.100	C	6	D
947.125	D	6	D
947.150	E	6	D
947.175	F	6	D
947.200	0	7	D
947.225	1	7	D
947.250	2	7	D
947.275	3	7	D
947.300	4	7	D
947.325	5	7	D
947.350	6	7	D
947.375	7	7	D
947.400	8	7	D
947.425	9	7	D
947.450	A	7	D
947.475	B	7	D
947.500	C	7	D
947.525	D	7	D
947.550	E	7	D
947.575	F	7	D
947.600	0	0	E
947.625	1	0	E
947.650	2	0	E
947.675	3	0	E

Appendix A-3

Receiver

Frequency	S1	S2	S3
947.700	4	0	E
947.725	5	0	E
947.750	6	0	E
947.775	7	0	E
947.800	8	0	E
947.825	9	0	E
947.850	A	0	E
947.875	B	0	E
947.900	C	0	E
947.925	D	0	E
947.950	E	0	E
947.975	F	0	E
948.000	0	1	E
948.025	1	1	E
948.050	2	1	E
948.075	3	1	E
948.100	4	1	E
948.125	5	1	E
948.150	6	1	E
948.175	7	1	E
948.200	8	1	E
948.225	9	1	E
948.250	A	1	E
948.275	B	1	E
948.300	C	1	E
948.325	D	1	E
948.350	E	1	E
948.375	F	1	E
948.400	0	2	E
948.425	1	2	E
948.450	2	2	E
948.475	3	2	E
948.500	4	2	E
948.525	5	2	E
948.550	6	2	E
948.575	7	2	E
948.600	8	2	E
948.625	9	2	E
948.650	A	2	E
948.675	B	2	E
948.700	C	2	E

Frequency	S1	S2	S3
948.725	D	2	E
948.750	E	2	E
948.775	F	2	E
948.800	0	3	E
948.825	1	3	E
948.850	2	3	E
948.875	3	3	E
948.900	4	3	E
948.925	5	3	E
948.950	6	3	E
948.975	7	3	E
949.000	8	3	E
949.025	9	3	E
949.050	A	3	E
949.075	B	3	E
949.100	C	3	E
949.125	D	3	E
949.150	E	3	E
949.175	F	3	E
949.200	0	4	E
949.225	1	4	E
949.250	2	4	E
949.275	3	4	E
949.300	4	4	E
949.325	5	4	E
949.350	6	4	E
949.375	7	4	E
949.400	8	4	E
949.425	9	4	E
949.450	A	4	E
949.475	B	4	E
949.500	C	4	E
949.525	D	4	E
949.550	E	4	E
949.575	F	4	E
949.600	0	5	E
949.625	1	5	E
949.650	2	5	E
949.675	3	5	E
949.700	4	5	E
949.725	5	5	E

Appendix A-4

Receiver

Frequency	S1	S2	S3
949.750	6	5	E
949.775	7	5	E
949.800	8	5	E
949.825	9	5	E
949.850	A	5	E
949.875	B	5	E
949.900	C	5	E
949.925	D	5	E
949.950	E	5	E
949.975	F	5	E
950.000	0	6	E
950.025	1	6	E
950.050	2	6	E
950.075	3	6	E
950.100	4	6	E
950.125	5	6	E
950.150	6	6	E
950.175	7	6	E
950.200	8	6	E
950.225	9	6	E
950.250	A	6	E
950.275	B	6	E
950.300	C	6	E
950.325	D	6	E
950.350	E	6	E
950.375	F	6	E
950.400	0	7	E
950.425	1	7	E
950.450	2	7	E
950.475	3	7	E
950.500	4	7	E
950.525	5	7	E
950.550	6	7	E
950.575	7	7	E
950.600	8	7	E
950.625	9	7	E
950.650	A	7	E
950.675	B	7	E
950.700	C	7	E
950.725	D	7	E
950.750	E	7	E

Frequency	S1	S2	S3
950.775	F	7	E
950.800	0	0	F
950.825	1	0	F
950.850	2	0	F
950.875	3	0	F
950.900	4	0	F
950.925	5	0	F
950.950	6	0	F
950.975	7	0	F
951.000	8	0	F
951.025	9	0	F
951.050	A	0	F
951.075	B	0	F
951.100	C	0	F
951.125	D	0	F
951.150	E	0	F
951.175	F	0	F
951.200	0	1	F
951.225	1	1	F
951.250	2	1	F
951.275	3	1	F
951.300	4	1	F
951.325	5	1	F
951.350	6	1	F
951.375	7	1	F
951.400	8	1	F
951.425	9	1	F
951.450	A	1	F
951.475	B	1	F
951.500	C	1	F
951.525	D	1	F
951.550	E	1	F
951.575	F	1	F
951.600	0	2	F
951.625	1	2	F
951.650	2	2	F
951.675	3	2	F
951.700	4	2	F
951.725	5	2	F
951.750	6	2	F
951.775	7	2	F

Appendix A-5

Receiver

Frequency	S1	S2	S3
951.800	8	2	F
951.825	9	2	F
951.850	A	2	F
951.875	B	2	F
951.900	C	2	F

Frequency	S1	S2	S3
951.925	D	2	F
951.950	E	2	F
951.975	F	2	F
952.000	0	3	F

Introduction

To the owners and operators of broadcast stations, time is money and STL outage time is lost money. So the major objective of performing STL path analysis is to create a reliable, trouble free system that has negligible outage time. The responsibility for this activity usually falls on the Chief Engineer. The purpose of this section of the manual is to aid the Chief Engineer, or his designate, in the preliminary design[1] of a Digital-Studio-Transmitter-Link (DSTL) using the Broadcast Electronics DP5500 system.

B.1 Fade Margin

The measure of robustness of an aural DSTL system can be expressed by its fade margin. A fade margin is the difference, in dB, between the magnitude of the signal received at the receiver's input terminals and the amount of signal required for a given level of performance. The level of performance used in any given analog STL system is a particular signal-to-noise ratio (SNR) that is typically 50 or 60 dB. Since in digital audio systems the ultimate SNR can exceed 90 dB, a new level of performance is established. However, since the DSTL is based on psychoacoustic principles, SNR measurement results can be misleading. In most digital communications systems, the level of performance is expressed as the Bit Error Rate (BER)[2]. The digital fade margin number then reflects the amount of degradation that can be experienced in the transmission path before the system performance degrades below the desired BER reference point (Typically 10^{-4}). As will be shown later, system fade margin requirements are directly related to the amount of tolerable outage time. The resulting fade margin depends on the STL equipment, path length, complexity and local weather conditions. Longer STL systems are exposed to more weather systems, are susceptible to more fades from a variety of causes, and therefore require a larger fade margin as is shown in Table 1.

Path Length	Path Length	Fade Margin
5 miles	8.05 km	5 dB
10 miles	16.1 km	7 dB
15 miles	24.14 km	15 dB
20 miles	32.2 km	22 dB
25 miles	40.2 km	27 dB
30 miles	48.3 km	30 dB

Table 1 NAB recommended fade margin versus path length.[3]

[1] A much more detailed path analysis should be performed by a broadcast engineering consultant or professional engineer.
[2] Ratio of the number of incorrectly received bits to the number of bits received.
[3] *NAB Engineering Handbook*, 8th Ed., National Association of Broadcasters, 1992.

An example of the relationship between system reliability and outage time is given in Table 2. From this table and a station's billing rate it is possible to calculate the revenue lost due to STL outage time.

RELIABILITY (%)	OUTAGE TIME (%)	OUTAGE TIME PER YEAR (Avg.)	MONTH (Avg.)	DAY (Avg.)
0	100	8760 hours	720 hours	24 hours
50	50	4380 hours	360 hours	12 hours
80	20	1752 hours	144 hours	4.8 hours
90	10	876 hours	72 hours	2.4 hours
95	5	438 hours	36 hours	1.2 hours
98	2	175 hours	14 hours	29 minutes
99	1	88 hours	7 hours	14.4 minutes
99.9	0.1	8.8 hours	43 minutes	1.44 minutes
99.99	0.01	53 minutes	4.3 minutes	8.6 seconds
99.999	0.001	5.3 minutes	26 seconds	0.86 seconds
99.9999	0.0001	32 seconds	2.6 seconds	0.086 seconds

Table 2 Relationship between system reliability and outage time[4]

One preferred method of deciding the required fade margin of an STL system is to use the mathematical expression below for calculating outage time:

$$\textit{Outage Time} = 1.3 \cdot a \cdot b \cdot f \cdot D^3 \cdot 10^{(-F/10)} \text{ (minutes/year)}$$

where: a = terrain constant (¼ smooth to 4 mountains)
b = climate constant (¼ dry to ½ humid)
f = frequency in GHz
D = distance in miles
F = fade margin in dB

A nomograph that summarizes several of these calculations, for a 950 MHz STL system for typical terrain and temperate climates, is shown in Figure 1. For paths over very smooth terrain, water or in humid climates the chart may not apply. Charts for these conditions can be developed using the formula given above.

It should be mentioned that often, in digital systems, the outages may be so brief that they may not be detected by the average listener. It may therefore be possible to have a perceived lower amount of outage time than actually experienced.

[4]*Engineering Considerations For Microwave Communications Systems,* AG Communication Systems, Northlake, IL, 1989.

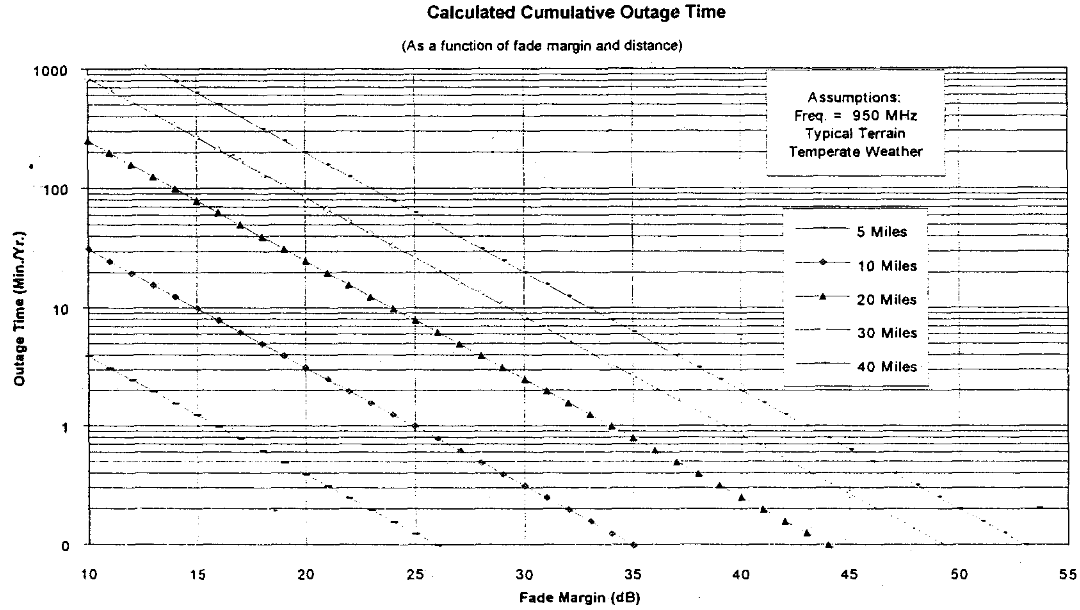


Figure 1 Typical Outage Time Nomograph

B.2 Path Calculations[5]

The prediction of the fade margin performance of a system is determined by performing a series of path calculations.

Path calculations start with the calculation of the DSTL transmitter output power in dBm. Then the loss or gain of each element in the RF transmission path, expressed in dB, is simply added in series from the transmitter's output until the signal reaches the receiver's input connector. For example, the figure below shows that as the transmitter signal leaves the output connector its level is reduced by the cable losses and then further by connector losses before it reaches the antenna. Then the signal is increased by the gain of the transmit antenna. Free space loss is then experienced over the path to the receiver antenna.[6] This is followed by the gain of the antenna at the reception site and then connector and cable loss before reaching the receiver's input port. The following paragraphs describe the calculation of the output power in dBm and then each possible gain and loss in a typical, single hop, DSTL system are discussed. Many of the sections, such as transmission lines, connectors and antennas apply to both the transmitter and the receiver legs of the path.

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[5]Contact Broadcast Electronics - Marti Facility for DSTL path calculations.

[6]There may be additional path loss for certain geographic areas that experience thermal inversion weather patterns and/or high humidity. In addition, there may be additional loss due to refraction around obstacles or multipath reflections. The reader is directed to the reference *Engineering Considerations For Microwave Communications Systems*, AG Communication Systems, Northlake, IL, 1989 for more information on these topics.

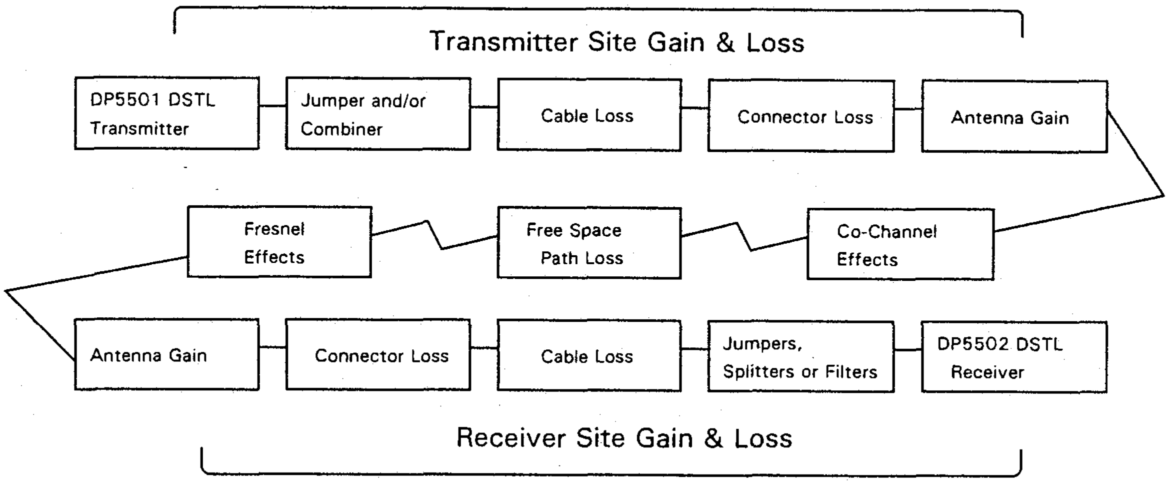


Figure 2 DSTL system block diagram for path loss calculations.

B.3 dBm Expression Of Power

As mentioned in the previous paragraph, it is convenient in path calculations to first convert transmitter power in watts to dBm where

$$dBm = 10 \cdot \log[P(Watts) \cdot 1000]$$

Some typical values of dBm for various output power levels are given in Table 3.

Power (Watts)	Power (dBm)	Power (Watts)	Power (dBm)	Power (Watts)	Power (dBm)
0.10	20.00	1.50	31.80	6.50	38.10
0.20	23.00	2.00	33.00	7.00	38.50
0.30	24.80	2.50	34.00	7.50	38.80
0.40	26.00	3.00	34.80	8.00	39.00
0.50	27.00	3.50	35.40	8.50	39.30
0.60	27.80	4.00	36.00	9.00	39.50
0.70	28.50	4.50	36.50	9.50	39.80
0.80	29.00	5.00	37.00	10.00	40.00
0.90	29.50	5.50	37.40	10.50	40.20
1.00	30.00	6.00	37.80	11.00	40.40

Table 3 Some dBm values for various power levels in Watts.

B.4 Transmitter Site Loss

B.4.1 Transmission Line Loss

Jumpers

In a typical installation, the first loss experienced in the STL path is the result of a jumper cable. This flexible jumper or "pig-tail" (three feet or less) is connected between the DSTL transmitter and the main cable, a low-loss, large diameter cable that is used to feed the transmit antenna. These jumpers, often constructed out of flexible, but higher loss RG-8A/U or 213A/U cable with male Type-N connectors, ease installation and prevent transmitter connector damage and provide strain relief. There is usually jumper between the antenna and the main cable as well. The center conductors of the Type-N connectors are susceptible to bending and breaking. In addition, excessive vertical plane torque on equipment with Type-N connectors, caused by heavy cables, can result in internal equipment failure.

Should jumper cables be used, take care to select cables that will not contribute to intermodulation and related problems. To this end, RG-8/U and similar cables should not be used. Rather, flexible, solid outer conductor shields cables should be used.

Antenna Transmission Line

The main loss, from the transmitter to the antenna, in a simple DSTL system is the loss experienced in the main transmission line to the antenna. The most frequent transmission line impedance for STL applications is 50 ohms and the standard connector used is the type-N connector. The selection of a transmission line typically depends on the impact that it has on the total path calculation. The lower the losses required to obtain a usable installation, the more expensive and exotic the transmission line becomes. Transmission line loss is primarily a function of the conductor loss and secondarily a function of dielectric loss. The larger the diameter of the transmission line, the lower its conductor loss will be. It is not unusual to use 1-5/8 inch diameter transmission line in systems that have long cable runs to the antenna. Dielectric loss becomes more significant at higher frequencies. The order of increasing dielectric loss, by dielectric type, is: air, foam, and then solid dielectric materials.

Finding the amount of cable loss is easy as most manufacturer's data sheets usually express the line loss in dB/100 feet (or 100m) of length at 1 GHz. Table 4 gives the attenuation of several common types of transmission lines at 950 MHz. It also should be pointed out that these figures are for 20°C and may need to be adjusted for other temperatures. Consult the transmission line supplier's data sheet if your environmental factors include temperature extremes.

Cable Type	Loss per 100 feet (dB)	Loss per 100 meters (dB)
Standard Foam Dielectric		
1/2 inch diameter	3.0	9.1
7/8 inch diameter	2.0	6.1
1-5/8 inch diameter	1.4	4.3
Low-loss Foam Dielectric		
1/2 inch diameter	2.4	7.3
7/8 inch diameter	1.4	4.3
Air Dielectric		
1/2 inch	2.7	8.2
7/8 inch	1.4	4.3
1-5/8 inch	0.7	2.1
Standard Coaxial Cable		
RG-8/U	8.5	26
RG-218/U	3.8	11.6

Table 4 Transmission line attenuation at 950 MHz for common transmission lines.

Since large inflexible transmission line is used for the main run, most installations use smaller gauge jumpers as strain relief. Therefore the main run should have type-N female connectors installed at both ends. Transmission line manufacturers usually have a full line of mounting hardware and attaching hardware to secure the transmission line to towers. Where possible avoid splices or multiple connections as they all introduce loss. Every dB saved contributes to increased fade margin.

B.4.2 Other Transmitter Site Loss

In more complicated STL systems a variety of additional devices may be placed into the signal path between the transmitter and the antenna to solve some unique problems experienced at that particular site. The sum of these additional losses should be accounted for in the path calculations.

Isolators — Tower Isolators or "Iso-couplers"

Many STL systems share the same tower with the antennas of other services. In these installations iso-couplers may be used to prevent ground-loops from occurring in the STL equipment. Center DC-Block, Outer DC-Block, Center and Outer DC-Block devices might be substituted to accomplish the same purpose. These devices typically add 0.5 dB of attenuation to the 950 MHz signal.

Isolators — Ferrite Circulators/Isolators

In certain locations, where there are many other transmitters, it is common to use a ferrite directional isolator to present a matched impedance at the transmitter output. Excessive VSWR will degrade the intermodulation distortion (IMD) performance or damage the transmitter. The worst case VSWR condition would be a RF open or short which reflects 100 percent of the transmitted power less twice the loss between the transmitter and the disturbance point. Sometimes external cavities or additional filters are used for the same purpose. These additional losses have to be accounted for in the path calculations. But, since the DP5501A/5503A has its own built-in isolator and low-pass filter these additional transmission line components, and their associated loss, is not generally necessary.[7]

Transmitter Combiners (Also applies to Receiver Splitters)

In more involved installations where more than one STL or DSTL system is used to transmit more than one RF signal through the same antenna, a power combiner is required. Bear in mind that the combiner, and splitter at the receiver site, introduce a loss of approximately 3.5 decibels and your path calculations should include provision for these losses. When selecting combiners be sure that there is adequate isolation between input ports. This will prevent the RF signal from one transmitter from being introduced to the output of the other transmitter. Such an interaction can cause intermodulation products that would result in the transmission of spurious products and in severe cases, may cause damage to the transmitter.

[7]The DP5501/5503 contains circuitry to monitor the forward and reverse power. Because of the location of this circuitry, with respect to the internal isolator, high levels of energy entering the output connector below 1 GHz may cause the internal power meter to read incorrectly.

Where bi-directional transmission in the same frequency band is necessary, duplexers can be used. In such installations a transmitter output and receiver input are connected to the same antenna terminal. These devices also introduce loss and must be accounted for in your path calculations. Another consideration to be mindful of is to ensure adequate spacing between the transmit and receive frequencies. For the Broadcast Electronics DSTL system, we recommend a spacing of 1 MHz provided that there is at least 40dB of isolation.

In dual mono systems, a passive transmitter combiner or coupler is used to allow two STL transmitters to operate on a single antenna. Because of the improved performance and capability of DSTL systems over dual mono system, DSTL systems will usually not need combiners. However, for four channel operation with two DP5501A's a combiner might be used in a single antenna application.

Filters and Cavities, Combiners

In environments where there may be unusually high RF fields, STL's including the DSTL may require the use of external filtering to attenuate these signals. Depending on the circumstances, a bandpass or band-reject type of filter could be used. Band-reject or notch filters are ideally suited when very specific frequencies are present and can benefit from being notched out. If potential interfering RF fields are of a more broad band nature, a bandpass filter that admits the frequencies of the STL band might be a more practical solution. These filters come in a variety of complexities. Better performance is achieved by using multiple stages.

Diplexers are filters that separate two or more frequency bands to separate receivers or transmitters. Typically they are used to frequency isolate transmitters and receivers from one another. Diplexers also may be used to frequency multiplex the use of the transmission line. These losses must be accounted for in the path analysis.

Hot Standby

Some STL installations incorporate a hot standby unit to allow for automatic switching to a secondary transmitter in the event that the primary transmitter goes off the air. The additional loss of this unit (typically no more than about 0.5 dB loss through a RF coaxial switch) should be included.

B.4.3 Connector Loss

Unless corrosion or debris is present, the loss in connectors is mostly due to impedance mismatch. Avoid cheap connectors that may not have adequate plating to resist weathering effects. When properly designed and properly installed, connectors should have no loss. However, for most path calculations it can be assumed that at 950 MHz, each connector contributes approximately 0.2 dB loss and 0.3 dB loss if the connectors are a right-angle type. Most path calculations take into account 1 dB of connector loss for the overall system.

B.5 Transmitter Site Gains

B.5.1 Power Amplifiers

Although an external power amplifier can improve the system gain, they are very expensive. Amplifiers used with DSTL systems must be class-A designs that have very high third-order intercepts[8] to avoid spreading of the DSTL signal into adjacent channels. In addition, take care to insure that undesirable interference does not occur to other users of the same (co-channel) or adjacent channels.

Because of the very high expense associated with linear power amplifiers, it is probably cheaper to use lower loss transmission line and higher gain antennas than to add more power.

B.5.2 Antennas

The choice of transmit and receive antennas is the most important consideration after the selection of the DSTL radio.

By far, the largest influence that one can make on fade margin is in the selection of antennas. Antenna gain is the variable most used in radio link design to satisfy path loss and fade margin requirements. The more gain that is required the more expensive and large the antenna becomes. Table 5 lists the 950 MHz gain in dBi (dB over an isotropic radiator) for various parabolic antenna diameters.

The larger the antenna, the more directional it becomes. In areas of spectrum congestion, that may be of significant importance. Antennas today are graded class A or class B in terms of their ability to meet requirements established to suit areas of spectrum congestion. Partial parabolic antennas may not exhibit a narrow enough beam width in certain orientations for areas that require FCC Category A antennas.

[8]Third-order intercept is an amplifier linearity figure of merit. Typically the intercept, expressed in dBm, is 10 dB higher than the 1 dB compression point.

Where environmental factors play a role, such as zoning restrictions, high winds or other severe weather, antenna choices may vary. A grid style antenna for example, is a better choice in areas of high wind. De-icers or radomes may be needed in areas of severe cold weather.

Prior to mounting the antenna it will be necessary to decide what polarization to select. Contact should be made with either the local frequency coordinating committee or your frequency neighbors to find out the polarization of signals nearest yours. Opposite polarization should be selected to gain the advantages of cross polarization attenuation of undesired signals. Alternatively you can experimentally determine the best polarization by rotating your antenna, if that is possible, into the best position.

Be sure to plan for a secure mounting method for the antenna, whether it be positioned on a tower, the side of a building or other structures. When mounted outdoors, allow sufficient margin for the effects of wind and the weight of snow or ice. Also allow for provision for mounting the transmission line.

Antenna Type	Gain at 950 MHz	Antenna Beam width
Antenna Size (feet / meters)	(dBi)	(degrees)
Grid Parabolic		
4/1.2	18.9	18
5/1.5	21.0	15
6/1.8	22.0	12
8/2.4	25.0	9
10/3.0	27.0	7
12/3.7	28.5	6
15/4.6	30.0	5
Solid Parabolic		
4/1.2	18.7	18
6/1.8	22.3	12
8/2.4	24.8	9
10/3.0	26.7	7
12/3.7	28.3	6
15/4.6	30.2	5
Paraflector®[9]	20.1	12V/24H[10]

Table 5 Typical isotropic gain for various antennas at 950 MHz.

[9]Paraflector is a registered trademark of the Scala Electronic Corporation.
[10]The Paraflector beam width is polarization dependent.

B.5.3 Free-space Path Loss

Free space path loss is the attenuation that results from a decrease in antenna effective area as the antennas are separated in distance. This loss can be calculated from the following equation or approximated from Table 6 for 950 MHz installations. Path loss for other frequencies and distances may be calculated from the following equation:

$$\alpha_{db} = 36.6 + 20\log f(MHz) + 20\log d(miles)$$

$$\alpha_{db} = 32.4 + 20\log f(MHz) + 20\log d(km)$$

Distance (Miles)	Distance (Kilometers)	Loss (dB)	Distance (Miles)	Distance (Kilometers)	Loss (dB)	Distance (Miles)	Distance (Kilometers)	Loss (dB)
1	1.6	96.2	16	25.7	120.2	31	49.9	126.0
2	3.2	102.2	17	27.4	120.8	32	51.5	126.3
3	4.8	105.7	18	29.0	121.3	33	53.1	126.5
4	6.4	108.2	19	30.6	121.7	34	54.7	126.8
5	8.0	110.1	20	32.2	122.2	35	56.3	127.0
6	9.7	111.7	21	33.8	122.6	36	57.9	127.3
7	11.3	113.1	22	35.4	123.0	37	59.5	127.5
8	12.9	114.2	23	37.0	123.4	38	61.1	127.8
9	14.5	115.2	24	38.6	123.8	39	62.8	128.0
10	16.1	116.2	25	40.2	124.1	40	64.4	128.2
11	17.7	117.0	26	41.8	124.5	41	66.0	128.4
12	19.3	117.7	27	43.4	124.8	42	67.6	128.6
13	20.9	118.4	28	45.1	125.1	43	69.2	128.8
14	22.5	119.1	29	46.7	125.4	44	70.8	129.0
15	24.1	119.7	30	48.3	125.7	45	72.4	129.2

Table 6 Free Space Path Attenuation at 950 MHz.

B.5.4 Absorption

The free space loss calculated above does not include the loss caused by the atmospheric absorption or precipitation between the transmit and receive antennas of a DSTL system. All microwave signals are attenuated, to some degree, by the presence of rain, snow and fog. Losses depend upon the frequency and amount of moisture in the path. At 950 MHz the signal attenuation due to water vapor is less than 0.001 dB/mile and can usually be ignored.

B.5.5 Fresnel Contours and Obstacles

The main beam of the STL transmitting and receiving antenna is formed by the interaction of the direct radiation of the antenna and the reflected radiation from the ground plane.

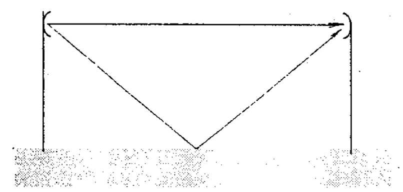


Figure 3 Direct And Reflected Paths

Since the antennas are directional, radiation at or near the center of the beam is more important than that occurring at azimuth angles away from the center. From this, it follows that there is an elliptical shaped area (with the major axis in the direction of the main beam, as shown in Figure 4) in which the ground must be level, clear of obstructions and smooth if the main beam is to be formed without appreciable distortion.

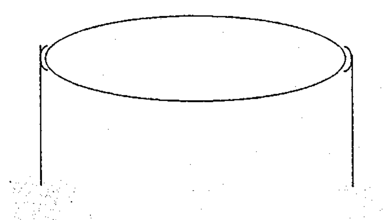


Figure 4 Ellipsoidal Fresnel Zone

The ellipse dimensions corresponding to the first Fresnel contours are vertical and horizontal profiles between the transmit and receive antennas, and describe the region in front of the antenna in which direct and reflected radiation differ in phase by 180^{o} or less.

$$F_{nr} = 2280\sqrt{\frac{n \cdot A \cdot B}{f_o \cdot P}} \text{ (feet)}$$

Where: n = Fresnel number

A= Distance from the transmit antenna to the obstruction

B= Distance from the obstruction to the receive antenna

fo= Frequency (MHz)

P= Total path length

As the STL antenna beam passes close to an object in the path as shown in Figure 5, some energy is reflected by that object, causing a variation in the received signal level. The STL frequency, the distance from the center of the STL beam, the distance to the obstruction and the distance to the ends of the STL path determine where the nulls and the peaks of the signal will occur.

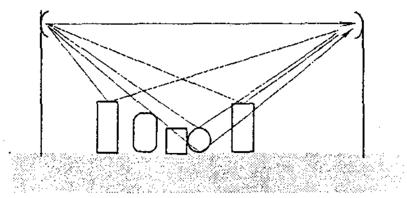


Figure 5 Multiple Reflections From Objects

The distance from the STL beam center, as it passes over an obstruction as shown in Figure 6, to the obstruction is measured in units of Fresnel zones. As described earlier, the first Fresnel zone (F1) is the surface of points along the path in which the total distance between the endpoints is exactly ½ wavelength longer than the direct end-to-end path. It is important to note that a clearance of at least 0.6F1 of the first Fresnel zone is required to maintain minimum free-space loss. When less than 0.6F1 clearance is present, the STL beam is considered to be obstructed. When the object is at or outside the first Fresnel zone, the distance from the object to the center of the STL beam is measured in actual Fresnel zones.

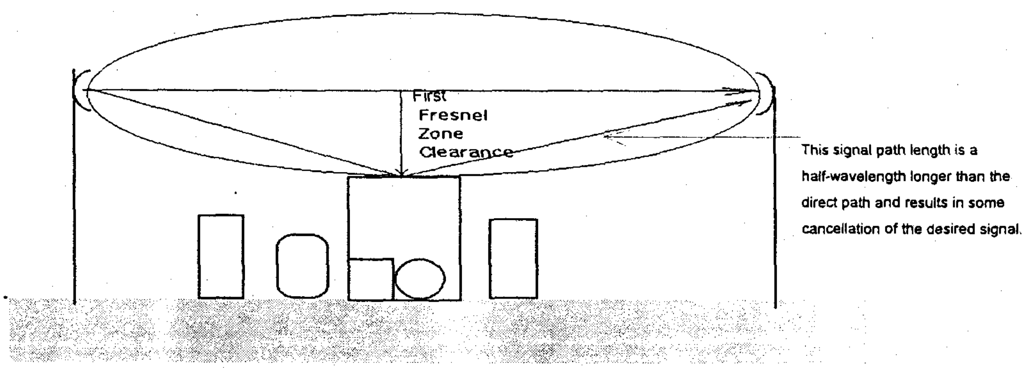


Figure 6 Fresnel Zone Clearance

Figure 7 illustrates a typical computer generated STL path profile used to determine earth-Fresnel zone interaction.

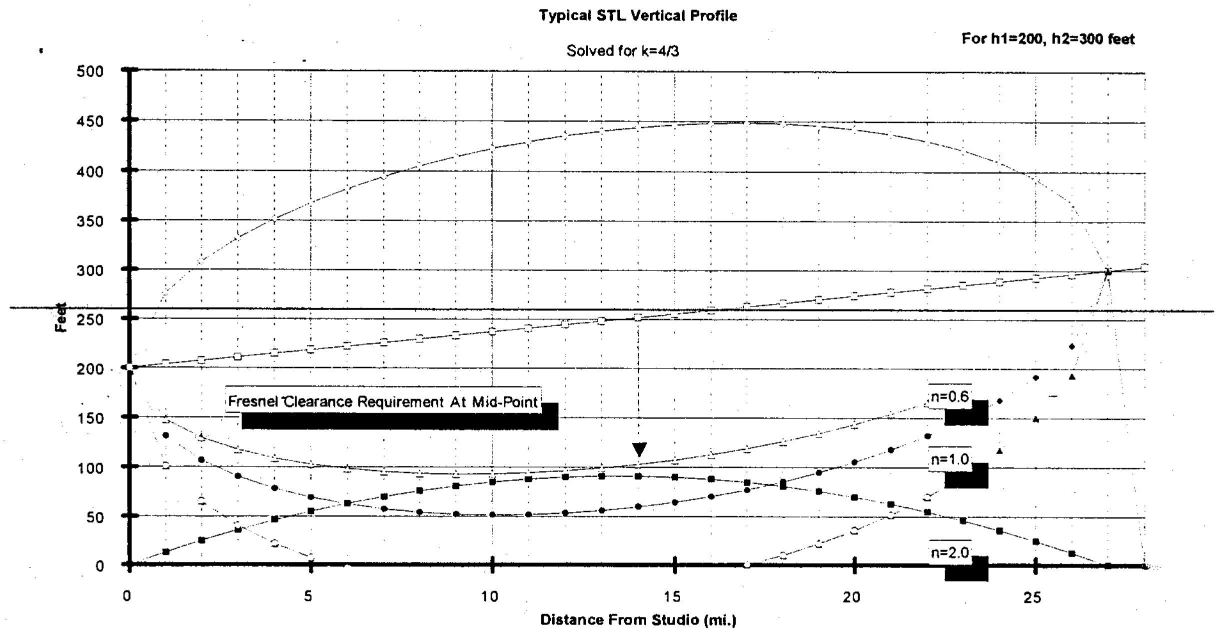


Figure 7 Path Profile Example

The type of obstruction and its shape determine the magnitude of the variation. If the obstruction is sharp in form, it can be anticipated that some refraction of the desired signal will occur over this obstruction. This diffraction is referred to as knife-edging and the loss is not overly significant. As most obstructions are not sharp in form, however, it is advisable to add some additional, loss possibly as much as 10 or 15 dB.

B.6 Receiver Site Gains

As with the transmit signal, the received signal level is increased by the gain of the receive antenna. All of the comments related to the transmit site also apply here. One additional point should be made about the receiving antenna. Since this antenna may be in a location that is the receiving site for many additional STLs, the antenna beamwidth, as well as its polarization, become important factors. A narrow beamwidth antenna and cross polarization are additional tools that can be used to filter out undesired co-channel and adjacent channel signals. Table 7 gives the calculated beamwidth for common 950 MHz parabolic antennas.

Gain and Beamwidth of 950 MHz parabolic antennas			
Antenna	Antenna	Gain	3dB
Diameter	Diameter		Beamwidth
(ft.)	(m.)	(dBi)	(degrees)
4	1.22	19.00	19.34
6	1.83	22.52	12.89
8	2.44	25.02	9.67
10	3.05	26.96	7.73
12	3.66	28.54	6.45
15	4.57	30.48	5.16

Table 7 Gain and Beamwidth of common STL parabolic antennas

B.6.1 Low-noise Preamplifiers

Antenna-mounted, low-noise preamplifiers can provide additional system gain. If the selected preamplifier does not have any preselection, it must exhibit good overload performance. Otherwise, it might be susceptible to strong undesired signals. Where the amplifier is "phantom" powered over the transmission line, bias insertion units with DC blocks must be used to isolate the RF and DC paths.

B.7 Receiver Site Loss

The loss mechanisms in the received signal path follow those in the transmit chain but in reverse order. There is the loss of the main cable followed by jumpers between the main transmission line and the receiver and the antenna and the main transmission line. As the main cable enters the receiver site it may terminate in an isolator, filter or another jumper before entering the receiver. As with the transmitter, each of these losses must be subtracted from the signal as it progresses.

In multiple receiver systems there may be a power divider or splitter to feed the multiple receivers from one antenna.

B.7.1 Filters and Cavities

In certain locations where there are strong pocket pager or other undesired signals (i.e. image frequencies) band-reject cavities, or notch filters, may be used to prevent receivers with bi-polar front-ends from overload and IMD. These losses must be added to the received loss. The DP5502/5504 has a very high dynamic range GaAs FET front-end that is much more tolerant of undesired signals. Notch filters and cavities should not be needed in most applications.

Diplexers may be used to frequency multiplex the use of the main transmission line. These losses should be included in the "other receiver loss" category.

B.7.2 Hot Standby

Some STL installations incorporate a hot standby unit to allow automatic switching to a secondary receiver in the event that the primary receiver goes off the air. The additional loss of the splitter contained within this unit, typically 3.5 dB, should be included in the other receiver loss category.

B.8 Required Signal Strength

The required, or desired, signal strength of an STL receiver is often expressed in µV, dBµV or dBm. The formula below and Table 8 give the relationship between dBm and microvolts in a 50 ohm system.

$$\mu V = \sqrt{50 \cdot 10^{\frac{Pi_{dBm}+90}{10}}}$$

$$dB\mu V = 107 + Pi_{dBm}$$

where: Pi= the receiver input power in dBm

	dBm	Microvolts	dBm	Microvolts	dBm	Microvolts	dBm	Microvolts
	-105	1.3	-89	7.9	-73	50.1	-57	315.9
	-104	1.4	-88	8.9	-72	56.2	-56	354.4
	-103	1.6	-87	10.0	-71	63.0	-55	397.6
	-102	1.8	-86	11.2	-70	70.7	-54	446.2
	-101	2.0	-85	12.6	-69	79.3	-53	500.6
DP5502 Mute Point	-100	2.2	-84	14.1	-68	89.0	-52	561.7
	-99	2.5	-83	15.8	-67	99.9	-51	630.2
	-98	2.8	-82	17.8	-66	112.1	-50	707.1
	-97	3.2	-81	19.9	-65	125.7	-49	793.4
DP5504 Mute Point	-96	3.5	-80	22.4	-64	141.1	-48	890.2
	-95	4.0	-79	25.1	-63	158.3	-47	998.8
	-94	4.5	-78	28.2	-62	177.6	-46	1120.7
	-93	5.0	-77	31.6	-61	199.3	-45	1257.4
	-92	5.6	-76	35.4	-60	223.6	-44	1410.9
	-91	6.3	-75	39.8	-59	250.9	-43	1583.0
	-90	7.1	-74	44.6	-58	281.5	-42	1776.2

Table 8 Conversion from dBm to microvolts.

B.8.1 Analog Systems

As the desired service of the DSTL system is to deliver the program audio to the transmitter site, a measurement of program signal-to-noise ratio (SNR) seems more appropriate than quieting figure or other receiver sensitivity measurement. It is customary to use 60 dB SNR. With improving noise floors of program sources, transmitters, and other components of the program chain, this figure should be increased to 70 dB or even higher.

B.8.2 Digital Systems

The measure of robustness of an aural 950 MHz DSTL system can be expressed by its fade margin. Fade margin is the difference, in dB, between the magnitude of the signal received at the receiver's input terminals and the amount of signal required for a given level of performance. The given level of performance used to be a particular signal-to-noise ratio. Since in digital audio systems, based on psychoacoustic principles, the SNR results are misleading, a different measure of performance is used. In a digital STL system the given level of performance is a Bit-Error-Rate (BER) of 10^{-4}. The fade margin reflects the amount of degradation that can be experienced in the transmission path before the system performance degrades below the reference point. The fade margin for requirements are a function of system length, complexity and local weather conditions. As shown in Figure 1 earlier in the appendix, longer STL paths are more susceptible to fade conditions due to weather and therefore require a larger fade margin. The system reliability is directly related to the calculated fade margin.

B.9 Interference — Other Signals

B.9.1 Co-channel

A signal that has the same carrier frequency as the desired channel. A co-channel signal (analog or digital) must be at least 20 dB below the desired signal to achieve optimum DSTL performance.[11]

B.9.2. Adjacent Channel

A signal that is located one channel away on either side of the desired channel. Depending on whether the signal is mono, stereo or digital the frequency offset maybe 250, 300 or 500 kHz away from the desired signal.

B.9.3 Alternate Service - Pagers, etc.

Beware of high power paging signals at 931 MHz often located near, or on the same tower as, the DSTL receiver. Besides overloading the DSTL receiver, these transmitters may splatter their sideband energy into the aural DSTL band. While the DSTL receiver is highly immune to interference use of external filtering discussed earlier may be necessary in some installations.

In other STL bands, there may also be sources of high-level RF energy.

[11]Some refer to this as a desired-to-undesired ratio of 20dB.

B.10 Worksheet Summary

The worksheet below provides a means to log and compute the factors necessary to determine the fade margin of the DSTL system. Most system designs are started with an assumed antenna size. After the fade margin has been computed for the selected antennas, the figure is reviewed to see if it meets the outage requirements. If the fade margin is not adequate, the next larger antenna or lower loss cables should be chosen. In extremely long paths or locations that have severe weather conditions, a multi-hop system may have to be considered.

R-F Link Analysis For DSTL Applications

Station Call Sign:	State:	Chief Engineer:				DP5500 System #			Rev. 5g
City:		Planner:				2 or 4 Channel:		2	
Tel. No.	N+	Lat. (--)	S-	W+	Long. (|)	E-	Hop #	Elev.	Ant. Hgt.
FAX. No.	Deg.	Min.	Sec.	Deg.	Min.	Sec.	of #	Ft.	Ft.
Studio Location Coordinates									
Transmitter Location Coordinates									
CASE #1 Dolby DP5500 DSTL System	#VALUE!	#VALUE!		#VALUE!	#VALUE!		#VALUE!	#VALUE!	
Frequency in MHz =				Bearing =		degrees			
Path Length in Statute Miles =			#VALUE!	Reverse Bearing =			degrees		
First Fresnel zone distance in feet =				Path Length in Kilometers =					
Transmitter Power (Watts)				First Fresnel zone distance in meters =					
	Stage	(dB)	(dBm)	NOTES:		Select One From Each Category			
Transmitter Output Power	1			Length(Ft.)	Size (Dia.)	Type(A-C)	A	B	C
Transmitter Line Loss	2						Std. Foam	Low-Loss	Air
Transmitter Connector and Jumper Loss	3						Number of		
Transmitter Antenna Gain	4						Grid	Solid	ParaFlector
Other Transmitter Loss	5						Filter	Isolator	Combiner
Free Air Path Loss	6								
Other Path Loss	7								
Receiver Antenna Gain	8						Grid	Solid	ParaFlector
Receiver Connector and Jumper Loss	9						Number of Jumpers =		
Receiver Line Loss	10						Std. Foam	Low-Loss	Air
Other Receiver Loss (-Gain)	11			Loss =			Filter	Isolator	Splitter
Receiver Input Power	12					Antenna PreAmp Gain =			0
Receiver Input Voltage (microvolts)									
Receiver Thermal Noise Power (No preamp)				Receiver Bandwidth in MHz =					
RF Gain Margin				Receiver Noise Figure in dB =					
Signal Required for BER=10^(-4)				Antenna ERP =			Watts		
Fade Margin for BER=10^(-4)				Est. Outage Time =			min./yr.		

Table 9 Sample worksheet for path calculations.

Once most of the data has been obtained, the actual design or engineering of the path is straightforward. The resulting system will be transparent to the listener and trouble-free.

Example of a completed path analysis worksheet:

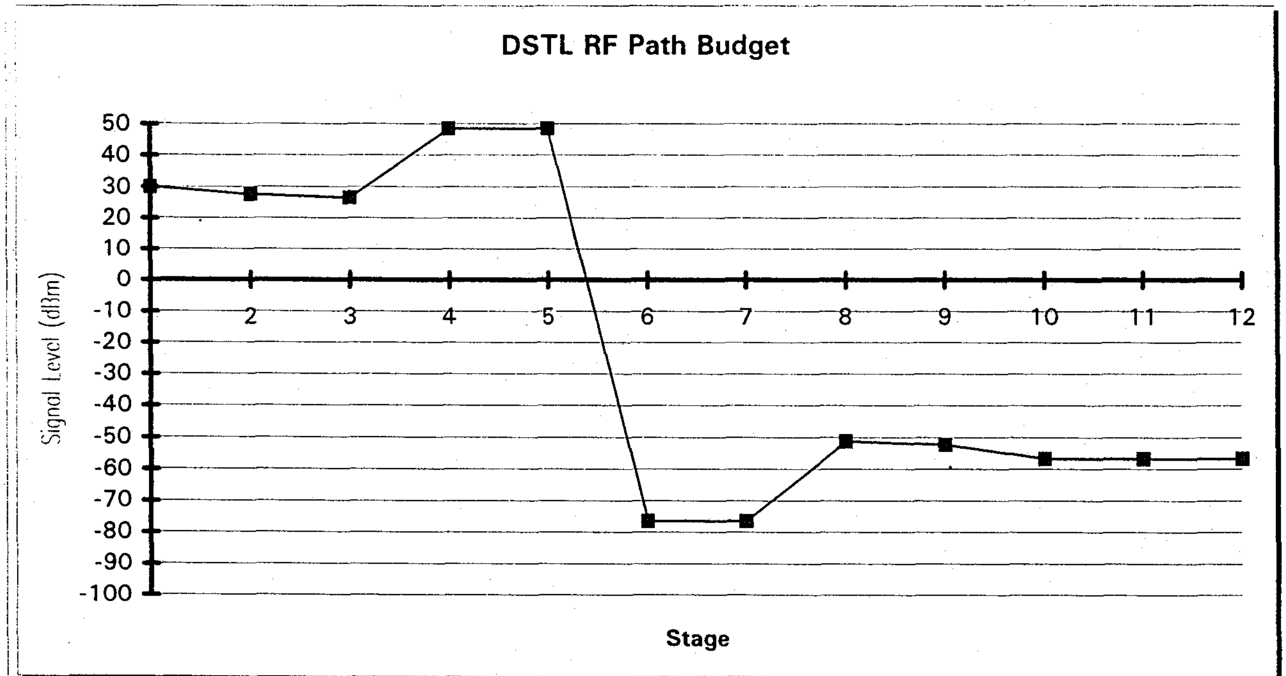
R-F Link Analysis For DSTL Applications

Station Call Sign: KZZZ	State:	Chief Engineer:				DP5500 System #			Rev. 5g
City: Anytown, CA		Planner:				2 or 4 Channel:		2	
Tel. No. 123-456-7890	N+	Lat. (--)	S-	W+	Long. (|)	E-	Hop #	Elev.	Ant. Hgt.
FAX. No.	Deg.	Min.	Sec.	Deg.	Min.	Sec.	of #	Ft.	Ft.
Studio Location Coordinates	41	39	17	70	17	4	1	100	170
Transmitter Location Coordinates	41	16	12	70	10	80	1	400	300

CASE #1 DP5500 DSTL System	41.65	41.27		70.28	70.18889	0.10	-0.983014
Frequency in MHz =		950.000		Bearing =	169.42 degrees		
Path Length in Statute Miles =		27.00	27.00175	Reverse Bearing =		349.42 degrees	
First Fresnel zone distance in feet =		192.19		Path Length in Kilometers =		43.45	
Transmitter Power (Watts)	1.00			First Fresnel zone distance in meters =			58.58

	Stage	(dB)	(dBm)
Transmitter Output Power	1		30
Transmitter Line Loss	2	2.6	27
Transmitter Connector and Jumper Loss	3	1.0	26
Transmitter Antenna Gain	4	22.0	48
Other Transmitter Loss	5	0.0	48
Free Air Path Loss	6	124.8	-76
Other Path Loss	7	0.0	-76
Receiver Antenna Gain	8	25.0	-51
Receiver Connector and Jumper Loss	9	1.0	-52
Receiver Line Loss	10	4.4	-57
Other Receiver Loss (-Gain)	11	0.0	-57
Receiver Input Power	12		-57
Receiver Input Voltage (microvolts)	324		
Receiver Thermal Noise Power (No preamp)			-110
RF Gain Margin		53.6	
Signal Required for BER=10^(-4)			-100
Fade Margin for BER=10^(-4)		43.2	

NOTES:		Select One From Each Category			
Length(Ft.)	Size (Dia.)	Type(A-C)	A	B	C
185	0.875	B	Std. Foam	Low-Loss	Air
			Number of Jumpers =		1
	6.0	A	Grid	Solid	ParaFlector
			Filter	Isolator	Combiner
	8.0	A	Grid	Solid	ParaFlector
			Number of Jumpers =		1
315	0.875	B	Std. Foam	Low-Loss	Air
Loss =			Filter	Isolator	Splitter
		Antenna PreAmp Gain =			0
Receiver Bandwidth in MHz =		0.50			
Receiver Noise Figure in dB =		6.6			
Antenna ERP =		69.3	Watts		
Est. Outage Time =		0.56	min./yr.		



B.11 References

Reference Data for Engineers: Radio, Electronics, Computer, and Communications, 7th Edition, Howard W. Sams & Co., Inc., Indianapolis, 1985.

Engineering Considerations For Microwave Communications Systems, AG Communication Systems, Northlake, IL, 1989.

NAB Engineering Handbook, Seventh Edition, National Association of Broadcasters, Washington, DC., 1985.

NAB Engineering Handbook. Eighth Edition, National Association of Broadcasters, Washington, DC., 1992.

Benson, K. Blair, *Television Engineering Handbook*, McGraw-Hill, New York, 1985

C.1 Repeater Operation: Minimum Frequency Spacing

Certain installations may require the use of repeaters. Examples of such installations include especially long paths or unavoidable obstructions between the studio and the transmitter site or, in some cases, frequency coordination problems.

A significant benefit of the DSTL system is that repeater functions can be performed in the digital domain so that signal degradation is non-existent. Configuring a DSTL system for repeater operation does not require any special modifications to the units themselves, only the relocation of jumpers on both units. Connectors are provided on the rear panel to interconnect a receiver and transmitter used in repeater operation. As with duplex operation, because of the presence of a transmitter and receiver in the same location, a minimum frequency spacing of 1 MHz is recommended to avoid receiver problems such as blocking from the local transmitter. See the next section for instructions on constructing a cable for repeater operation.

Please note that both the DSTL transmitter and receiver can be ordered with modules omitted for dedicated repeater operation. On the transmitter, these include the omission of the Cat. Nos. 452/472 and 453/463 since, in repeater operation, additional audio signals cannot be combined with the digital signal coming from the repeater receiver. On a DSTL receiver intended only for repeater operation, the Cat Nos. 462/482 and 463/483 are omitted. However, the omission of these boards prevents the ability to recover the audio signal at the repeater site. If these boards are retained, the audio signals are available. In addition, if the Cat. No. 483 is retained in the DP5504, data channel signals are also available at the repeater site. When a system is ordered for repeater operation, repeater cables are shipped with the units (see Section 4.3.8 for jumper configurations).

C.2 Repeater Cables

In applications where the DSTL system is installed in a repeater configuration, a DP5502/5504 receiver is connected to a DP5501A/5503A transmitter by a 9-pin D connector cable between connectors **J107**. This cable supplies two pairs of differential TTL lines for both data and clock signals in addition to ground lines. A special 15-pin D connector cable is also used to ensure proper communication from the receiver to the transmitter. This cable should be installed between connectors **J109** (Hot Standby) as labeled. Following cable installation, place the **MODE** switch of the DSTL repeater transmitter in the **REMOTE** position, and place the **MODE** switch of the repeater receiver in the **OPERATE** position.

Introduction

Broadcast Electronics offers the Model HS1 DSTL transmitter hot standby unit and Model HS2 DSTL receiver hot standby unit for installations requiring hot standby capabilities. These products provide full-featured control and flexibility in these applications, including interface capability to analog STLs.

For less stringent applications, DSTL transmitters can be switched manually, since personnel are likely to be present at the studio. For semi-automated switching of DSTL receivers at the transmitter site, the following section describes a master/slave wiring scheme.

D.1 Not using a Hot Standby Unit – Master / Slave Configuration.

It is possible to configure two DP5502/5504 receivers in a master-slave configuration without the use of a HS2 hot standby unit. This is done by setting the mode switch of the "master" unit to **OPERATE**, while setting the mode switch of the slave unit to **REMOTE**. This creates a situation where the slave unit only goes into the "operate" state if the master unit is forced into the "standby" state due to a detected malfunction. The following interconnection allows the user to arbitrarily select one (and only one) unit as the master unit.

Receiver "A" and "B" should be connected together with 100% shielded cable using male 15 pin D connectors wired as follows:

Unit "A"		Unit "B"
pin 1	-------	pin 7
pin 2	-------	pin 4
pin 4	-------	pin 2
pin 7	--------	pin 1

Short pins 6 and 9 together at each connector J109. Do not connect the cable to these pins. Care should be exercised to keep any unshielded wires to a short length. Connect this cable to connectors J109.

D.2 Using a Hot Standby Unit from Another Manufacturer

In applications involving existing hot standby units or systems, it may be possible to adapt the DSTL to those systems. An adapter cable employing a male 15 pin D-sub connector, a 100% shielded cable, and the appropriate connector for the existing hot standby system should be constructed. The following table is the pin-out and function of the STL's hot standby connector J109, which will assist you in determining how to best interface the units, and to assist in making a cable. Care should be exercised to keep any unshielded wires to a short length.

DP 5501A/2/3/4 J109 Pin	Signal Description
1	Ground pin to chassis.
2	Operate control, when unit is remote shorting this line to ground puts the unit in the operate mode.
3	**DP5501A/5503A Transmitter only.** This line is pulled to 5 volts through a 4.7 kohm resistor when the unit is radiating RF, otherwise it is pulled to ground.
4	"Operate" tally relay connection **open** to pin 6 when the unit is in the operate mode.
5	"Operate" tally relay connection **shorted** to pin 6 when the unit is in the operate mode.
6	The common connection to the "operate" tally relay.
7	"In remote" tally relay connection **open** to pin 9 when the unit is set for remote operation.
8	"In remote" tally relay connection **shorted** to pin 9 when the unit is set for remote operation.
9	The common connection to the "in remote" tally relay.
10	"No Summary alarm" signal. This line is pulled to ground when the Summary Alarm is inactive (normal operation). Otherwise it is open.

J109 Pin	Signal Description
11	Not used
12	Not used
13	Not used
14	+ 15 volt supply with 1 kohm resistor in series.
15	- 15 volt supply with 1 kohm resistor in series.

D.3 Wiring Considerations

Transmitters

The audio input cables to the left, right, auxiliary, and voice channels are connected in parallel to the terminal blocks of both STL units. The cables should be 100 % shielded twisted pair cables suitable for connection to #6 screw terminal blocks. Care should be exercised to keep any unshielded wires to a short length.

For DP5503A installations, the input cables to the Data 1 and Data 2 channels are connected in parallel to the 9-pin D data connectors of both units, in conformance with EIA standard RS-232.

Receivers

Systems employing antenna redundancy have each antenna feed connected to each receiver unit. Otherwise the RF input for both units is obtained by using an N connector power splitter; in this way the single RF input is equally distributed between the two units.

The audio output cables from the left, right, auxiliary, and voice channels from both units should be connected in parallel. They should be 100 % shielded twisted pair cables suitable for connection to #6 screw terminal blocks. Care should be exercised to keep any unshielded wires to a short length.

For DP5504 installations, the output cables from the Data 1 and Data 2 channels from both units should be connected in parallel, in corformance with EIA standard RS-232.

The composite outputs should be connected to 50 ohm, 100 % shielded coaxial cables with BNC connectors at each end. Both outputs from the two units should be connected together with a "Tee" connector.

APPENDIX E
GLOSSARY OF TERMS

9-QPRS. 9-level Quadrature Partial Response Signaling. Modulation method that maps digital data into 9 different points in level and phase. This provides a spectrum efficiency higher than schemes such as QPSK, which are less complex. But 9-QPRS is not so complex as to compromise the robustness of the DSTL system.

AC-2®. Dolby Laboratory's method of audio coding used by the DSTL system to effectively reduce the data rate of digital audio data, resulting in a practical degree of RF spectrum efficiency without compromising audio quality. For further information, consult Appendix F.

Adjacent Channel. The next higher or lower occupied radio frequency band.

AES/EBU. A world-standard digital interface specification for electrical interconnection of digital audio systems. The DSTL system will, as a future option, have the capability to accept AES/EBU connections.

BER. Bit Error Rate. Specified in errors per number of bits received in a transmission system. For example, a BER of 10^{-6} means that one bit in one million may be erroneous. The DSTL system's high degree of immunity to errors means that audible degradation is just barely perceptible when the BER is 10^{-4}: one bit in 10,000. The DSTL system mutes when the BER exceeds 10^{-3}.

CAL. During setup, LED indicators assist in calibrating input levels to provide 12 dB of headroom before digital clip occurs in the analog to digital conversion on the Cat. No. 452/472 module. Not used when all processing occurs before the DSTL system.

Carrier/Noise Ratio. The ratio in dB of the mean carrier power to the mean noise power.

Class A amplifier. A type of amplifier that is biased such that the output voltage is continuously conducting throughout the electrical cycle (i.e. 360 degrees).

Co-channel. Occupying the same channel.

Constellation. A signal state vector diagram representation of a modulated signal (i.e. 9 states for 9-QPRS).

DAB. Digital Audio Broadcasting. A concept for broadcasting digital audio to radio listeners. The DSTL system, because of its modular nature, can evolve to accommodate the future needs of DAB.

Digital Clip. The hard overload characteristic of a digital audio channel, corresponding to the maximum possible digital representation of a coded audio signal. The DSTL System incorporates safety limiters to avoid this condition.

DSP. Digital Signal Processing. Dedicated "computer on a chip" that specializes in performing extremely complicated digital processes—such as that needed for Dolby AC-2® audio coding and implementation of a digital stereo generator—on the DSTL.

Error Rate. see BER

Fade Margin. Difference between calculated or actual received signal strength and minimum signal strength needed for a receiver to demodulate the received signal. When performing path analysis calculations, an adequate margin will ensure that operational integrity will be sustained despite factors that may degrade the received signal strength, such as atmospheric effects and interference.

The presence of a co-channel interferor, and to a lesser degree, strong adjacent channel signals, will degrade the minimum signal strength needed for proper receiver operation, thus degrading the fade margin. These factors need to be considered when analyzing a path.

For further information, please consult Appendix B.

Form C. Definition of relay contact configuration. Form C contacts allow access to the Normally Open, Normally Closed, and Pole (Common) contacts.

Fresnel Zone. A locus of points between the transmit and receive antennas that represent a surface where a reflected signal could cause destructive interference. See Appendix B.

Fresnel Zone Clearance. The radii from the direct path to edge of the Fresnel Zone. See Appendix B.

GaAs FET. Gallium Arsenide Field Effect Transistor. These transistors are very linear microwave amplifying devices. The DSTL uses GaAs FET low-noise devices in the receiver and high-power, high-linearity, devices the power amplifier.

Headroom. The difference between the peak level at clipping and the average peak level.

High Intercept. The intercept point, measured in dBm, is a figure of merit for intermodulation product suppression. The higher the intercept point, the better the IM suppression. Therefore, a **high intercept** is desirable. The intercept point is the point at which the fundamental response and the third order spurious response curves intersect.

Hot Standby. The automatic ability to switch over to a backup unit in the event of an alarm condition.

I,Q. The data signals from the modem module to the frequency synthesizer module. They represent digital data in quadrature.

IF Mon. A test port on the Cat. No. 466/486 module in the DP5502. This monitor point is typically (-46 dBm), 11 dB below the IF output from the receiver module.

Linear Amplification. Occurs when an amplifier is operating within its small signal gain region of its power transfer function curve. Typically, the small signal region is 3 to 10 dB below the 1 dB compression point.

Master/Slave Hot Standby. A backup strategy that defines the primary DSTL as the Master and the Backup unit as the slave. In this strategy, operation will return from the Slave to the Master once the alarm condition is reset. See Appendix D.

MAX (L,R). Maximum of the Left or Right signal.

MODE. Status of the DSTL unit can be: Operate, Standby, or Remote.

Overshoot Limiter. A signal processing circuit that prevents the input signal from causing modulation overshoots.

Path Analysis. A study of the expected propagation gains, losses and prediction of the expected signal strength. See Appendix B.

Pre-emphasis. Signal conditioning whereby the high-frequency content of the input signal is emphasized prior to transmission.

QPSK. Quadrature or quaternary phase shift keying.

QPRS. Quadrature partial response system.

R CHAN. On the digital stereo generator, Cat No. 460, a switch to enable modification of the right channel audio (or test oscillator) signal to facilitate crosstalk and separation measurements.

Repeater, Digital. In a terrestrial digital microwave transmission system, a repeater system is employed to extend the transmission path or re-route the transmission path due to interference, since a microwave radio system requires a "line of sight". A repeater system typically consists of a receiver and transmitter and is either a regenerative or non-regenerative type. A regenerative repeater is when the RF signal goes through the complete modulation-demodulation-regeneration process. A DP5501A/5503A and DP5502/5504 operated in repeater mode is of this class. A non-regenerative repeater heterodynes the RF signal to IF and is then upconverted and transmitted. This process bypasses the demodulation-modulation action.

RF Mon. A monitor port which is provided for the purpose of connecting a spectrum analyzer.

RXO. Repair Exchange Order. Broadcast Electronics procedure for exchanging defective modules, minimizing field-repair hassles and down-time. Replacement modules are stocked at Broadcast Electronics - Marti Facility. For further information see Section 11.

Sample Rate. On the DSTL system, the sample rate is 44.1 kHz, the same as CDs, which is rapidly becoming the primary signal source in radio broadcasting.

SCA. Subsidiary Communications Authorization (USA).

Signal Adaptation. Spectral compensation applied in the Cat. No. 453/473 Encoder and Cat. No. 463/483 Decoder modules to optimize AC-2 coder performance when the program audio is pre-emphasized.

Summary Alarm. A status indicator that summarizes the status of all the modules in a DSTL unit.

Sync. On the Audio Decoder module, Cat. No. 463/483, the L,R Sync LEDs indicate that the decoder has synchronized to the program audio data from the Cat. No. 464/484 Demodulator. The Aux Sync LED provides a similar indication for the Aux and Voice channel data on the Cat. No. 463.

TEST OSC. On the digital stereo generator, Cat. No. 460, a convenient, digital, oscillator can be introduced into the signal path to simplify stereo generator setup and Proof of Performance testing (separation, crosstalk).

Ultralinear. This refers to extremely low distortion products such as intermodulation distortion generated by an amplifier or other circuit.

VSWR. Voltage Standing Wave Ratio. An indicator of reflected power at the transmitter RF output. Excessive amounts are indicative of problems with antennas, transmission line, or connectors. Although the DSTL will withstand infinite VSWR (100% reflection of the transmitted power), the **Summary Alarm** will be triggered when VSWR exceeds approximately 12%, giving an early warning of potential transmission problems.

Dolby® Technical Papers

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RF Design Considerations in the Development of a High-Spectral Efficient, Multi-Channel, All-Digital STL

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RF DESIGN CONSIDERATIONS IN THE DEVELOPMENT OF A HIGH-SPECTRAL EFFICIENT, MULTI-CHANNEL, ALL-DIGITAL STL

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ABSTRACT

An all digital approach to the design of an aural studio-transmitter-link (STL) for the 944-952MHz band imposes several unique requirements on the radio frequency (RF) design of the transmitter and receiver portions of the system. The FM radio designs that have served the broadcasters' STL needs so well in the past are no longer adequate. Where high performance and spectrum efficiency are the primary needs only new approaches in transmitter and receiver RF designs will suffice for the digital-studio-transmitter-link (DSTL™).

This paper describes the RF technologies behind the design of the radio-frequency portion of the DSTL. These include the use of gallium-arsenide microwave monolithic integrated circuits (MMIC), power devices, dielectric and surface-acoustic wave (SAW) filters, multiple p-i-n diode arrays, and high dielectric-constant stripline circuits.

INTRODUCTION

The heart of the aural 950 MHz STL link for many years has been the tried and true analog FM radio. This classical approach to the broadcasters' needs, until recently, has served the industry well. Now with spectrum congestion in all the major markets there is talk of Category A antennas[1], compatible sharing of spectrum and segment-allocation schemes[2]. At the same time the radio listening audience has come to realize the benefits of digital audio performance and are now demanding the benefits of this technology in their radio entertainment.

One totally different approach to the congestion/performance problem is a spectral efficient radio. A spectral efficient radio could mean lower power for the same or higher system fade margin, lower occupied bandwidth or a combination of these two specifications. A DSTL radio with a 250 kHz occupied bandwidth could double the capacity of the current STL band while also improving both the signal-to-noise performance (figure 1) and the system fade margin (table 1) by a considerable amount.

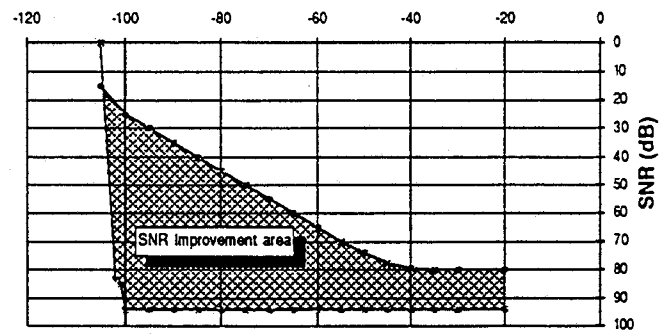

Fig. 1 DSTL SNR Improvement Area

There are a variety of analog ways to build a spectrum-efficient radio with analog technology; single-side band (SSB), narrow-band frequency modulation (NBFM), etc. However, none of these approaches provides the necessary audio fidelity required by the marketplace. Then there are the digital modulation techniques such as pulse code modulation (PCM) that provide the high-fidelity performance, but are inefficient users of the available spectrum.

Recently digital audio coding technology[3,4] has provided a new tool for developing spectral-efficient modulation techniques. A family of digital audio encoders and decoders that allows high quality stereo audio to be transmitted efficiently through existing digital data channels over terrestrial, wired or wireless, or satellite links are available today.

When one of these digital audio coding technologies is combined with a judicious choice of digital modulation a new performance standard for STL service is created. Hence the DSTL. The benefits of the DSTL are:

- Wide audio bandwidth
- High signal-to-noise ratio
- No crosstalk
- Degradation-free multiple hops
- Constant audio SNR during substantial fades
- Higher system gain (greater fade margin)
- Lack of background chatter
- No phase distortion
- Encryption against pirates

To successfully implement the DSTL concept into a workable approach it is first necessary to develop the desired carrier signal, perform the modulation and amplify the result to a usable signal level. The first two functions are performed in the DSTL's exciter module and the latter in the power amplifier module.

The key to maintaining the spectral-efficiency of the audio coding technology and modulation format is processing those signals through highly linear stages. If during any of the RF modulation, up-conversion or amplification processes the signal experiences any form of odd-order non-linearity it will corrupt the occupied bandwidth.

Figure 2 illustrates the three technologies that are integrated into the DSTL system.

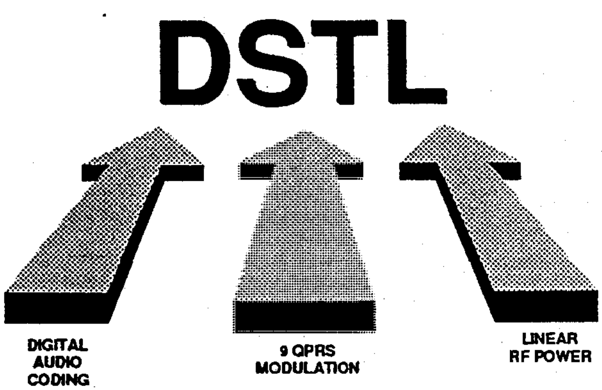

Fig. 2 DSTL Technology Base

RF DESIGN CONSIDERATIONS

Transmitter Design

The DSTL transmitter, as shown in Figure 3, is comprised of an A/D, DSP, Digital Modulator, Exciter, Power Amplifier and Power Supply modules. The A/D module provides the multi-channel analog-to-digital conversion. The DSP board contains the Dolby AC-2 data compression technology. The Modulator board performs the digital 9-QPRS encoding. The Exciter board performs the RF frequency synthesis and RF modulation functions. The Power Amplifier performs the amplification and power control functions. Special consideration is given to the linear power amplification required in an all-digital radio system.

In addition, the transmitter has an Alarm Module which monitors the status of all of the other modules and the power supply.

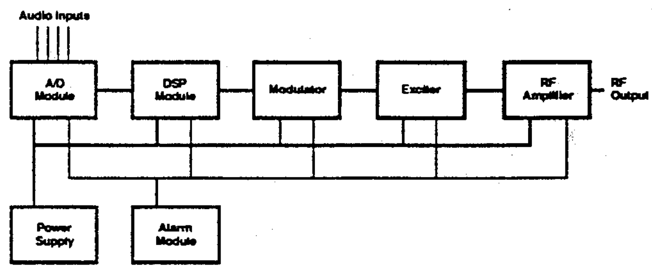

Fig. 3 DSTL Transmitter Block Diagram.

Spectrum Efficiency and Output Power Capability

Proposed spectrum mask requirements would need a 250 kHz occupied bandwidth. For this requirement Dolby chose to utilize AC-2 encode technology combined with 9-QPRS modulation which provides a spectral efficiency of 2 bits/s/Hz.

Estimated receiver performance and path length calculations determined that, for most applications, only a 1 Watt output would be necessary to provide adequate fade margins.

In order to maintain the spectral efficiency of the modulation it was further determined that the power output stage would have to have a third-order intermodulation intercept point of +60 dBm[5]! Linear microwave power takes on new meaning in the DSTL radio.

Exciter Design

The various blocks in the exciter module are shown in Figure 4.

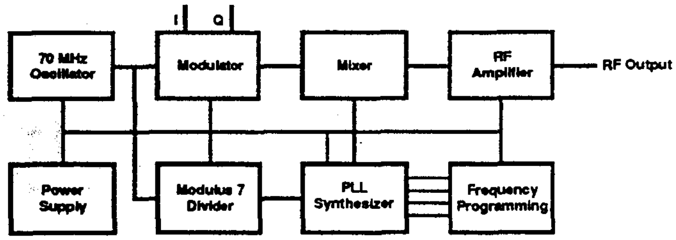

Fig. 4 DSTL Exciter Module Block Diagram

Master Oscillator

A temperature stabilized 70 MHz oscillator is used as the source for the reference signal for the frequency synthesizer and the carrier for the modulation. The reference signal is processed by a fixed modulus-7 divider to provide a 10MHz signal for the phase-locked-loop (PLL) integrated circuit (IC). The other 70 MHz signal is modulated by the I and Q channels from the modem and then up converted to the 944-952 MHz band in a later part of the exciter.

Field-programmable Frequency Synthesizer

If this radio is to be used in an area that plans to take advantage of its two-for-one spectral-efficient capability it must have the flexibility of being frequency agile. A field-programmable frequency synthesizer approach gives the DSTL transmitter and receiver frequency agility in order to manage circumstantial frequency relocation.

The CMOS PLL IC has an internal reference divider that is programmed to divide the 10 MHz input signal, from the modulus-7 divider by another factor of 400. This division, when used with a dual-modulus prescaler, allows the DSTL to be frequency programmed in 25 kHz steps anywhere in the 944-952 MHz U.S. STL band. User programming is via switches located on the front panel of the frequency synthesizer.

The emitter-coupled logic (ECL) dual-modulus, 128/129 prescaler, translates the 1014-1022 MHz voltage controlled oscillator (VCO) signal into a lower frequency signal that the CMOS PLL integrated circuit can handle.

Audio Source Coding Technology

Dolby AC-2 coder technology provides the two high-quality 15 kHz channels, a 7 kHz auxiliary and a 3 kHz voice/modem channel to the DSP module. In this module 16-bit, 44.1 k samples/sec bit-rate reduction is achieved. By using a low time delay implementation of Dolby AC-2 data compression, less than 250 kHz of STL bandwidth is required with approximately 8 msec. time delay in the main audio channels.

9-QPRS Modulation

Spectrum efficiency is achieved by the use of QPRS signaling, at 70 MHz, in conjunction with a system cosine filter that results in partial response signaling as disclosed by Todd.[6]

Up-conversion

The 9-QPRS modulated 70 MHz signal is up-converted by a high-side injection, passive, double balanced mixer. This mixer and its drive level were chosen to produce the lowest possible third-order intermodulation distortion. Broadband resistive terminations are used on all ports of the mixer to properly terminate the image and spurious frequencies.

Dielectric Filters

In order to filter out the undesired frequency products from the up-converter, a ceramic-block dielectric filter is used. The filters operate, in theory, similar to microwave quarter-wavelength interdigital-line comb filters constructed from round rods. These, however, are constructed out of a material that has a very high dielectric constant (Er range from 20-100) and low loss tangent. It is not uncommon for these structures to have unloaded Qs greater than 5,000.

Unlike the machined metal construction of older microwave filters the dielectric is machined and then plated. This results in a very cost-effective, physically small (7 mm x 9 mm x 27 mm) filter with less than 2 dB insertion loss.

Two- and four-pole versions of these filters are used in both the DSTL exciter and the receiver designs.

RF Power Amplifier Design

The heart of the power amplifier is a stripline sub-assembly. This sub-assembly exhibits over 50 dB of linear power gain. Since amplifier load mis-matches can impact the IM performance of the last stage an isolator is an integral part of the sub-assembly. Following the isolator is a six-port directional coupler. Part of this structure is used as a 30 dB monitor port for the transmitter. Another part of the six-port coupler is used to allow for customer setting of the output power and to maintain the power level constant over temperature and aging. An ALC loop has been designed into the power amplifier. An output filter keeps the harmonic content of the transmitter greater than 70 dB below the nominal 1 Watt output signal.

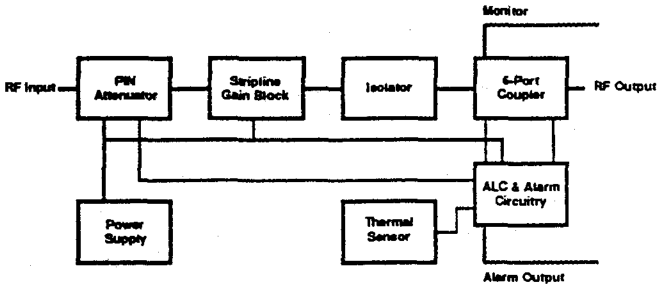

Fig. 5 DSTL Power Amplifier Module Block Diagram

Third-order Intercept Requirements

As mentioned earlier, the 9-QPRS modulation format requires a ultra-linear power amplifier to preserve the spectrum-efficiency. To ensure that the amplifier intermodulation products are greater than -60 dBc the third order intercept performance must be close to +60 dBm.

There were three different amplifier design approaches that were investigated: "Pre-distortion", "Feed-forward" and "Back-off". Pre-distortion did not lend itself to the need for long term stability over time and temperature. Although significant improvements in distortion cancellation have been reported and achieved using feed-forward techniques, a design using this approach also exhibited time and temperature effects that were difficult to control. In addition the feed-forward technique requires an additional side-chain amplifier and had high component and labor costs.

Upon first investigation the back-off approach would appear to be too costly because of the amount of raw input power and thermal dissipation required. However, further investigation indicates that if ultra-linear devices are used in the amplification stages the input power and heat problems are manageable.

GaAs MESFET Design

Due to their many non-linear mechanisms[7] bi-polar silicon devices clearly do not have the linearity required for DSTL applications. Prior to the availability of microwave Gallium-Arsenide (GaAs) metal-gate field-effect (MESFET) power transistors, high intercept amplifiers had to be constructed using feed-forward or pre-distortion techniques. As mentioned, both of these techniques have gain and phase stability problems that result in complex and costly support circuitry.

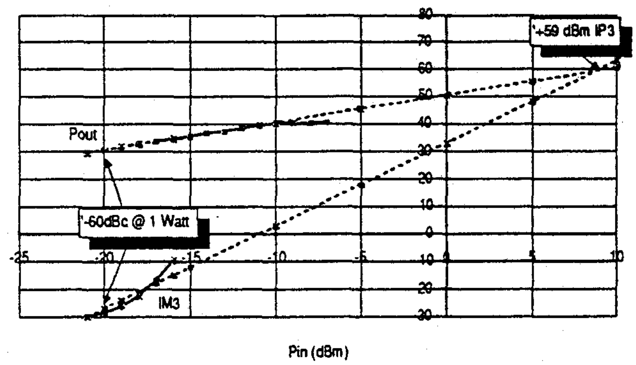

Fig. 6 Measured DSTL Third-order Intercept

Today, linear Class-A GaAs FETs are available with 20 Watt performance up and into the C-band frequency region.

Because of their inherent higher gain compared to available bi-polar transistors only three stages of power amplification are required. Figure 7 illustrates the gain distribution that is realized by each device. The gain numbers reflect the intrinsic forward gain of the device and the gain realized by providing the optimum input, interstage and output impedance match.

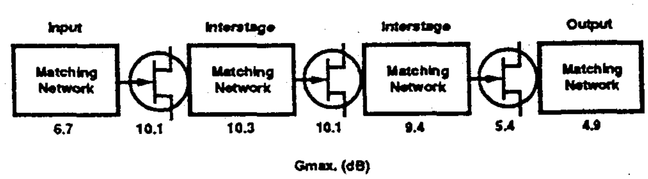

Fig. 7 DSTL Power Amplifier Gain Distribution

Impedance Matching Structures

To provide the device impedance transformation, DC block function and some degree of filtering a unique tri-plate, broadside-coupled quarter-wavelength stripline resonator structure was developed. These structures were optimized for bandwidth and manufacturing yield by the use of Touchstone software.[8]

Ceramic Loaded Teflon Substrate

Conventional microstripline design techniques were avoided because of their large physical size and spurious radiation problems. The final power amplifier design was realized by using totally enclosed three-layer stripline. The stripline structures were realized using high-dielectric constant, e.g. 10.5, microwave, ceramic loaded Teflon, material.

The above mentioned material and stripline structure combine to provide the advantages of being physically small, wideband and capable of handling high power. This amplifier is constructed without the need for expensive discrete blocking capacitors or spring wound coils. The structure is thermally stable and provides its own EMI shielding.

Monitor, Filter and Control Circuitry

The power amplifier is activated by an OPERATE signal from the transmitter front panel. This operate signal is processed by a logic circuit which monitors all of the module's operating voltages. This circuitry prevents the power amplifier module from being activated in the event of the loss of any power source which might damage the devices in the power amplifier sub-assembly.

Part of this control circuitry also monitors the power amplifier sub-assembly temperature. The circuitry is designed to disable the power amplifier in the event that there is an over temperature condition.

Samples of the forward and reverse output signals are coupled via part of the six-port directional coupler to biased Schottky-barrier diode detectors. Signal processing circuitry monitors the detected forward power signal and adjusts the input p-i-n attenuator to keep the output forward power level constant.

The detected reverse power signal is compared to the forward power signal to determine if high VSWR conditions exist. When the VSWR exceeds 3:1 the circuitry initiates an alarm signal which in turn activates the alarm LED on the module and a summary alarm LED on the transmitter's front panel.

Performance

The spectrum mask of the completed amplifier at the 1 Watt output level is shown in the figure below.

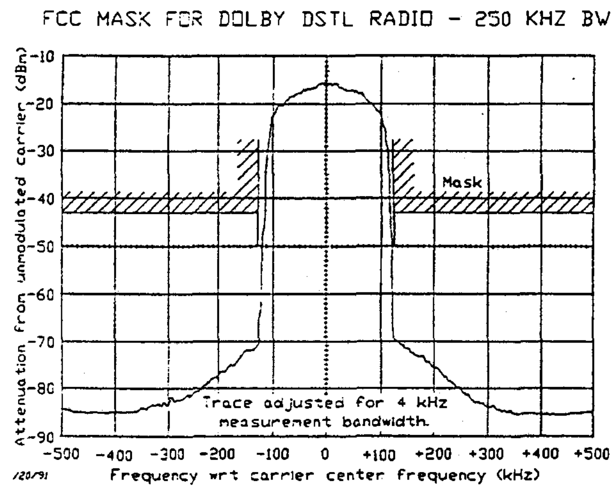

Fig. 8 Occupied Spectrum Mask of DSTL

All IM products are well below the unmodulated carrier by 60 dB. Since the power amplifier is so linear, harmonics of the unit are greater than -70 dBc. Very little output filtering is required for normal amplification.

Receiver Design

Description - The receiver is comprised of a Receiver/ Synthesizer, Modem, DSP, D/A, Alarm and Power Supply Modules. A digitally synthesized FM Stereo Baseband Generator module is also available.

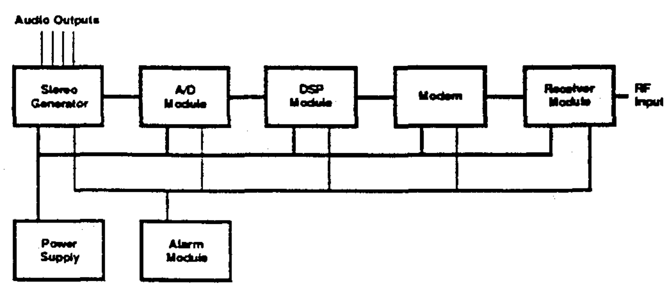

Fig. 9 Receiver Block Diagram

Front-end Design

Preselector considerations and design

In many metropolitan areas the most common STL receiver locations are fraught with a variety of undesired high level signals. In addition to multiple STL signals, there are quite often pocket pager, mobile and other services just below and above of the 944-952 MHz STL band. Often, these signals are too close to the STL band or too high in amplitude for the receiver front-end to handle.

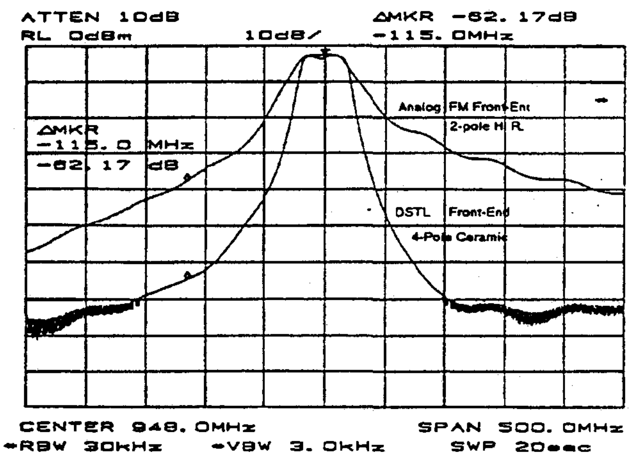

Fig. 10 Receiver Selectivity Curves

The above figure demonstrates the receiver front-end selectivity of a typical analog FM STL receiver and the DSTL receiver. As can be seen from the curves even a high amount of selectivity will still allow some out-of-band energy to get through to the RF pre-amplifier stage. The only way to prevent this stage from producing IM is to select amplifier and mixer stages with moderate gain and high third-order intercept capability. Even then it will be necessary to balance the gain distribution with the right amount of AGC action.

Third-order Intercept Requirements

In the DSTL receiver the RF pre-amplifier device is a GaAs MMIC which was chosen to have a good noise figure and high third-order intercept to prevent the creation of IM products. The majority of the front-end selectivity follows this stage where the filter's insertion loss has lower impact on the noise figure. Since no active mixers could be located with high third-order intercepts, i.e. greater than 18 dBm, a passive mixer was chosen to perform the frequency conversion function.

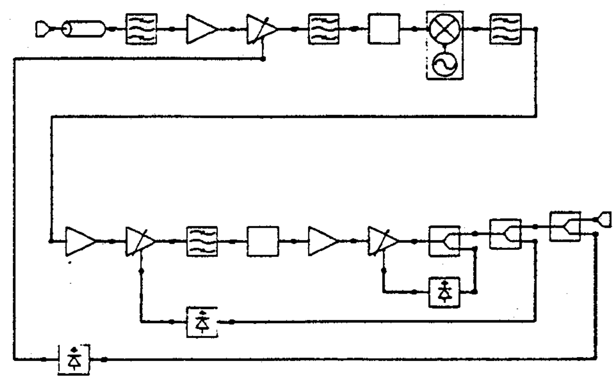

Fig. 11 DSTL Receiver AGC Block Diagram

IF Design

Third-order Intercept Requirements

Like the transmitter chain, the receiver must be highly linear. Critical to over all performance of the DSTL system is low IM distortion. Each stage of the 90 dB gain IF chain has been designed to maximize it's third-order IM intercept-point and control the amount of gain.

Surface-Acoustic-Wave Filtering

IF selectivity is provided by a surface-acoustic-wave filter. This filter has a 3 dB bandwidth of 1 MHz and exhibits 60 dB of alternate channel attenuation. Its small physical size and good temperature stability make it an ideal choice for this application.

AGC Considerations and Implementation

Another key part of maintaining the IM distortion in the receiver chain is to provide the correct amount of interstage AGC action. Improper front-end AGC action can result in either front-end IM or degradation in the receivers noise figure. Improper AGC action in the IF section can result in undesirable clipping resulting in a high bit-error-rate.

The following figure demonstrates the DSTL receiver's AGC characteristic over a -120 to -20 dBm range of input signals. The figure indicates that the RF signal level in each stage is controlled to keep the generation of IM as low as possible.

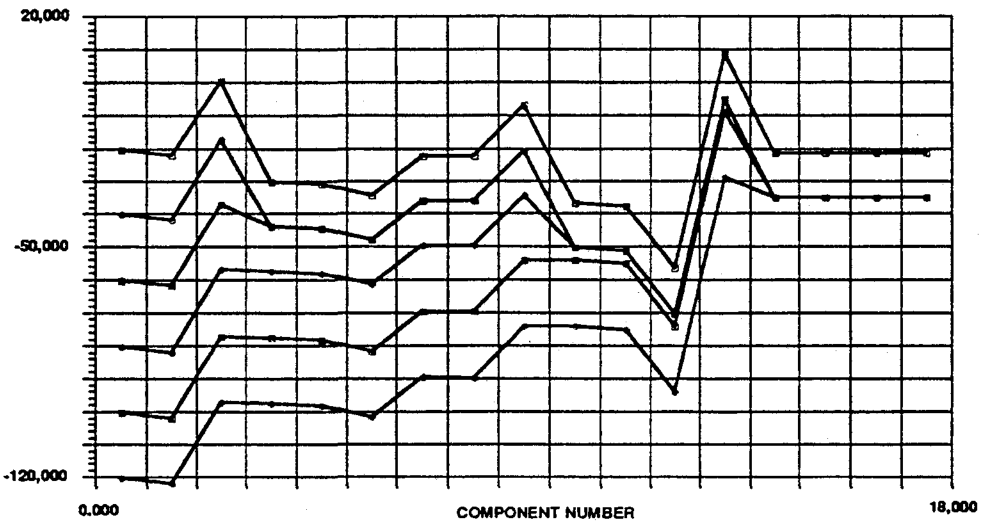

Fig. 12 AGC Budget Analysis for Various Input Levels

MECHANICAL DESIGN CONSIDERATIONS

Unique Requirements

Need for Modularity

Rapid technological changes can make a product obsolete if it has no ability to be up-graded. With the advances in audio, digital signal processing power, and direct digital frequency synthesis it was felt that a modular approach would provide the user with the flexibility over years of ownership. In addition, the modular approach provides service and exchange benefits should they ever be needed.

Thermal Design Aspects

From the RF electronic packaging point of view the most difficult problem is thermal management. The power amplifier module dissipates close to 30 Watts of heat in its normal mode of operation. Most RF amplifier design engineers are used to junction temperature values of 200°C in bi-polar devices. In order to meet the DSTL's desired MTBF performance the channel temperature of the GaAs power FETs must be keep well below their 175°C maximum operating point. Data provided by the device manufacturer indicated that the typical thermal resistance is about 70% of the given maximum values. From these device ratings and the amplifier's operating efficiency the heat sink and thermal interface requirements were determined.

The resulting heat sink design runs the full length of the power amplifier module and results in a device MTBF of over 750,000 hours at +70°C ambient.

ELECTROMAGNETIC-MAGNETIC INTERFERENCE CONSIDERATIONS

A product design that takes into account EMI considerations from conception will have fewer of those problems in the product launch cycle and over its operating life. Interference and susceptibility from either the RF, digital circuitry or external environmental fields have the potential of reducing a product's performance. To preclude this possibility, all of the modules developed for the DSTL contain EMI suppression. All entry and exit lines of the RF modules contain RF filtering and are shielded.

SUMMARY

It was once thought that all digital radios required more bandwidth than analog radios. As shown, when the optimum audio coding technology is combined with the proper choice of digital modulation and RF technology this is no longer true. The multi-channel DSTL radios will now start to replace their less spectrum efficient analog predecessors.

These new technologies will also have an impact on other future high-spectral efficient digital radios.

ACKNOWLEDGMENTS

The author wishes to acknowledge the efforts of Edmond Chu in the fabrication and testing of the RF circuitry.

SAMPLE PATH LENGTH AND FADE MARGIN COMPARISONS

		DSTL	Analog FM		Notes
LOSS					
	Path	-122	-122	dB	20 mi. / 32.2 km.@ 950MHz
	Transmission Line	-5.6	-5.6	dB	400 ft. / 122m. (7/8" foam)
	Connectors	-4	-4	dB	Total
	Others			dB	
TOTAL SYSTEM LOSSES		**-131.6**	**-131.6**	**dB**	
GAIN					
	Transmitter Power	30	35	dBm	DSTL 1W. / FM 7 W.
	Transmit Antenna	15	15	dB	6 ft. grid parabolic
	Receive Antenna	15	15	dB	6 ft. grid parabolic
	Others			dB	
TOTAL SYSTEM GAINS		**60**	**65**	**dBm**	
TOTAL SYSTEM LOSSES		**-131.6**	**-131.6**	**dB**	
TOTAL SYSTEM GAINS		**60**	**65**	**dBm**	
Received Signal Strength		-71.6	-66.6	dBm	
Desired Signal Level		-95	-66.9	dBm	
Fade Margin		**23.4**	**0.3**	**dB**	DSTL @10^-4 BER. FM 100 uV for 70 dB SNR

Table 1

REFERENCES

[1] R. Rudman, "Sharing the crowed spectrum." *Broadcast Engineering,* Jan. 1991.
[2] R. Balonis, "Living by the rules," *Broadcast Engineering,* Mar. 1989.
[3] G. Davidson, L. Fielder, and M. Antill. "High-Quality Audio Transform Coding at 128 kbits/s." *Proceedings of Int. Acoustics, Speech and Signal Processing,* April 1990.
[4] G. Davidson, L. Fielder and M. Antill, "Low-Complexity Transform Coder for Satellite Link Applications", *J. of the Audio Eng. Soc.* Preprint#2966, Sept. 21-25, 1990.
[5] R. Bell, "Class-A STL RF Power Amplifier Analysis," Report prepared for Dolby Laboratories, May 16, 1989.
[6] C. Todd, "NTSC Compatible Digital Modulation for TV Sound," *NAB Proceedings,* 44th NAB Engineering Conference, 1990.
[7] R. Bell, and R. Clark, "Elimination of Cross-Modulation in CATV Amplifiers," *21st NCTA Convention Record,* April 1972.
[8] Touchstone is a trademark of EEsof.

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AC-2: A Family of Low Complexity Transform Based Music Coders

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AC-2: A FAMILY OF LOW COMPLEXITY TRANSFORM BASED MUSIC CODERS

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Two high-quality data rate reduction music coders from a family of TDAC transform based coders are discussed. An overview of the psychoacoustic principals used in their design is given and their limitations discussed. The use of psychoacoustics and DSP technology are combined to yield a low complexity approach to music coding. Issues of complexity, word length requirements, and memory usage are examined for both general-purpose DSP and custom IC implementations.

0 INTRODUCTION

The use of data rate reduction coders for digital audio applications shows great promise for a large variety of storage and transmission applications. Since Compact Disc digital audio employs a data rate greater than 1.4 Mbits/sec., this type of digital audio has been limited only to areas that can maintain a high data rate. Fortunately, the development of high quality data rate reduction technology for music applications has changed this situation. Now lower data rates may be used for audio in radio and television broadcast, computer hard disk storage, and telephone line connections. This paper will describe two coders from the Dolby AC-2 family, developed for different applications, that have the desired characteristics of data rate reduction, excellent sound quality, and computational simplicity.

The need to reduce the data rate for the practical application of digital audio into many areas has resulted in much work in the field of data rate reduction for music, as typified by Brandenburg et al. [1990], Johnston [1988], Schroeder et al. [1987], Stoll and Dehery [1990], Davidson et al. [1990], and Fielder [1989]. The fundamental approach of these techniques is to divide the audible frequency range into sub-bands which approximate auditory critical bands. Crucial elements in the design of these coders are the bit allocation and quantization schemes in which perceptually relevant sub-bands are identified, and the appropriate fraction of the available bit rate assigned to their representation. Many of these algorithms require a great deal of processing power to perform the frequency division and quantization operations (e.g., multiple DSP chip implementations for a single audio channel). Furthermore, they all extrapolate published models of human hearing and masking to a broader class of signals than those upon which the models were based.

This paper builds on the work described by Davidson et al. [1990] and Fielder [1989] which described 15 kHz bandwidth coders with resultant data rates between 128 and 192 kbits/sec. per channel. The two coders described here have 20 kHz bandwidth, require less than one programmable DSP chip to implement one stereo pair, and possess excellent sound quality. In particular, one coder, which will be called the low-delay coder, achieves excellent subjective and objective quality at 4:1 compression, exhibits robust tandem coding performance (i.e., where a number of encode/decode processes occur in series) and has a coding/decoding delay less than 9 msec. This low-delay feature is essential for applications requiring that announcers monitor their own coded voice signals. The other coder trades coding delay for a lower bit-rate (6:1 compression) and will be called the moderate delay coder. The coding systems described here can be applied for either 44.1 k or 48 ksamples/sec., however the remaining discussions will center on 48 ksample/sec. results.

A general overview of the psychoacoustics of masking as it effects the design of data rate reduction music coder technology will be given. Next, the details of the two coding systems resulting from this psychoacoustic examination will also be presented. Issues of implementation will also be discussed and the use of 24-bit and 16-bit DSP chips will be examined and processor speed/memory requirements determined. The use of custom DSP chips will also be considered. It will be shown that the two systems described are quite low in complexity while at the same time providing excellent sound quality.

1 APPLICATION OF PSYCHOACOUSTIC MODELS TO CODER DESIGN

The basis of all good rate reduction music coders is the application of the psychophysics of the human auditory system. As a result, a discussion of the present state of knowledge in this area is essential for the understanding of coders of this type. Masking effects for simple signals will be extended to the development of the filter bank design and quantization technology used in music coders. It will be seen that the targeted application will greatly influence the way the psychoacoustic principals are utilized. Next, these principles will be extended to more complex signals and discussed for AC-2 coding. An indication of the effectiveness of the AC-2 coding system in controlling the amount and frequency characteristics of the errors due to the reduction of word-lengths for data rate reduction will be given by a spectral comparison between both coder's performance and frequency characteristics of auditory masking.

1.1 Critical-Band Model of Hearing

Central to the development of a workable model of the auditory system is the critical-band concept and its relationship to the masking characteristics of the ear. The critical-band model of the human auditory system was first developed by Fletcher [1940] to explain why masking experiments showed that signals covering a frequency range less than a certain threshold bandwidth produced the same masking and detection properties as other signals with smaller bandwidths. The fundamental approximation is that the ear acts as a multi-channel real-time analyzer with varying sensitivities and bandwidths throughout the audio range. Despite the intrinsic simplicity of the model, it has been shown to be very enduring. Effective data rate reduction coders for music rely heavily on this model.

The critical-bandwidth represents the minimum frequency bandwidth resolvable for masked signals. For example, the masking of a low level error signal caused by a larger level tone nearby in frequency is maximal and continues at a constant level until the frequency separation between them exceeds this bandwidth. Detection of a signal component takes place based on the entire energy within a critical-bandwidth, whether it is tonal in nature, noise-like, or a combination of the two. Later workers have further refined this concept; Zwicker et al. [1957] examined this resolution bandwidth via various detection and masking experiments. Later Zwicker [1961] established 24 fixed critical-bands over the 20 Hz - 15 kHz frequency range.

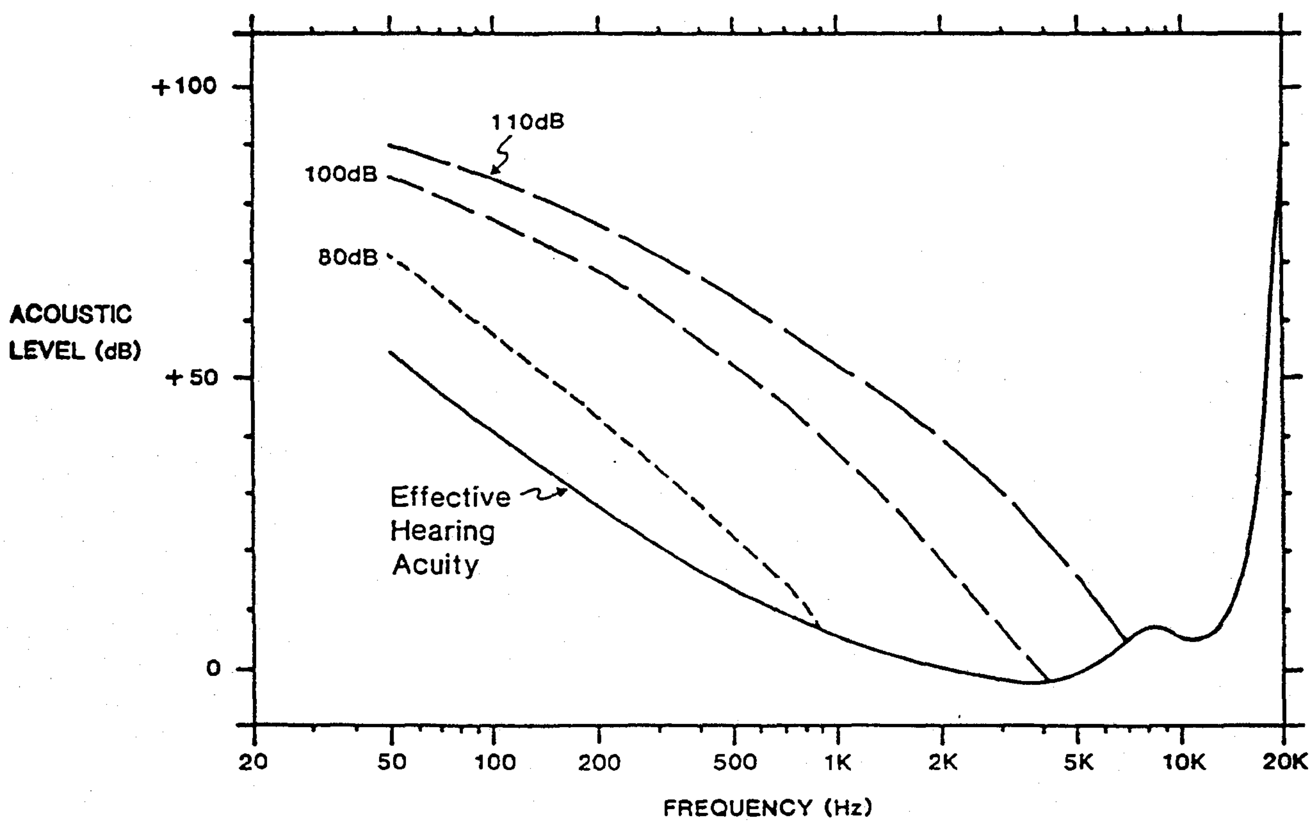

Fig. 1 100 Hz Masked Threshold Curves (0 dB = 20 micropascals)

1.2 The Use of Single Tone Masking Curves

Information on the masking effect of signal components is available primarily for single tones or bands of noise. As a result, coder design depends greatly on principles derived from these simple masking experiments. These typically generate masking curves of single high level component masking the presence of another smaller component and are quite useful because they can be used to derive an upper bound on the levels of permissible error signals due to the data rate reduction process. Since the masking effect varies significantly depending on whether the large level component or masker is tone-like or noise-like in character, the more demanding situation of sinewave masking curves are shown in Figures 1, 2, and 3. The figures present various 1/3 octave hearing thresholds when subjects are subjected to various levels of 100 Hz, 500 Hz, and 4 kHz sinewave maskers, as described by Fielder [1987]. For more information on the variation of the masking effect for tonal or noise signals, see Ehmer [1959].

The most appropriate way to examine masking phenomena is to perform a spectrum analysis based on critical-bandwidths. Since critical-band analyzers are not common, a good approximation can be made with the use of 1/3 octave bands; see Fielder [1987] for further details. These spectral analyses of masking are then used as a basis for the design of the coder filter bank structures and the methods to reduce the bit rate via word-length reduction.

The first observation from Figures 1-3 is that masking is generally greatest at the masker's frequency. This indicates that the coder design should concentrate error energy directly adjacent to the signal frequency. The next property the figures have in common is that the masking effect slowly decreases with increasing frequency separation, if the smaller signal is higher in frequency than the masker. The masking effect for signals at a 70 dB acoustic level may extend only a few octaves upward in frequency while higher level situations may produce six upward octaves of significant masking.

Looking at masking of signals lower in frequency than the masker shows a very different situation. For these signals, the masking effect falls off much more quickly. This is particularly evident for frequencies between 500 Hz - 2 kHz when evaluated in a dB per Hz fall-off from the masker frequency; in this frequency region the slope can be as steep as 100 dB per 350 Hz below 500 Hz (i.e., 90 dB/octave) and drop as deep as 40 dB within 1/2 octave. This rapid decrease in masking for components lower in frequency than the masker has significant

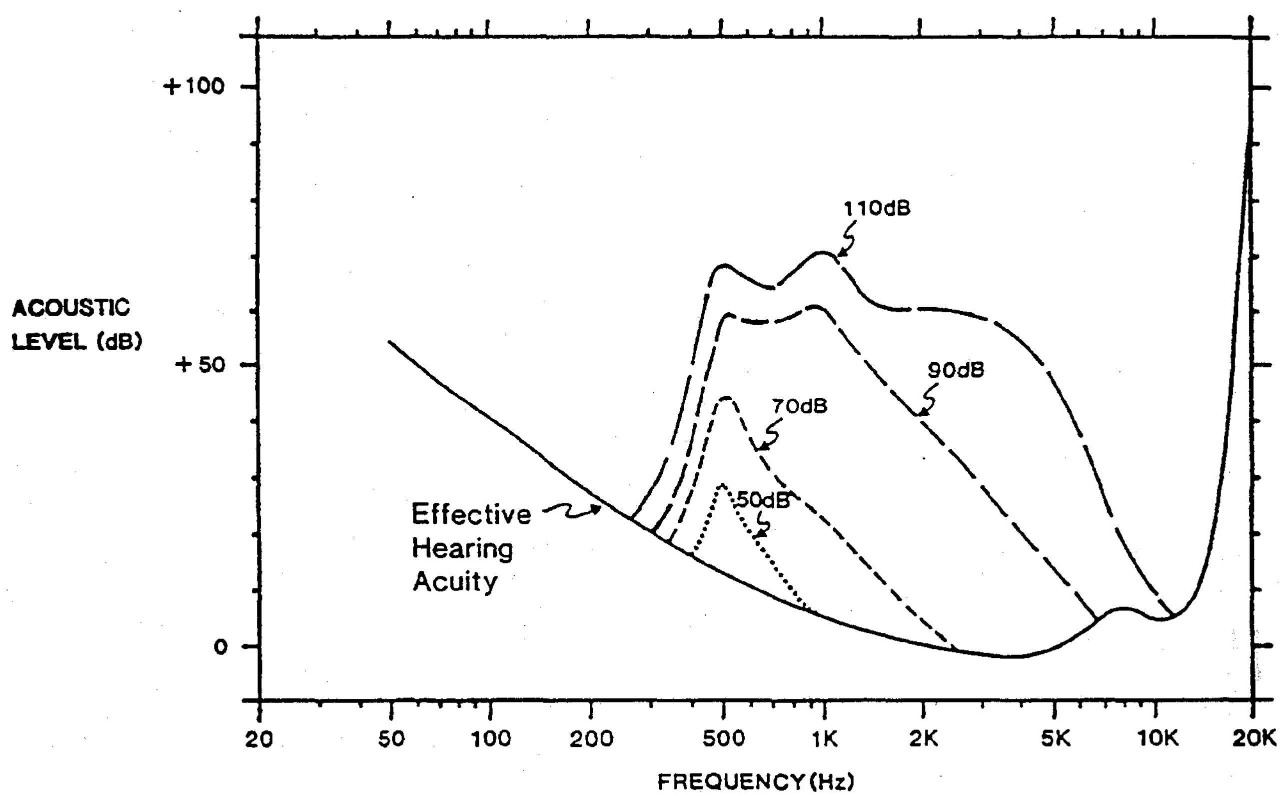

Fig. 2 500 Hz Masked Threshold Curves (0 dB = 20 micropascals)

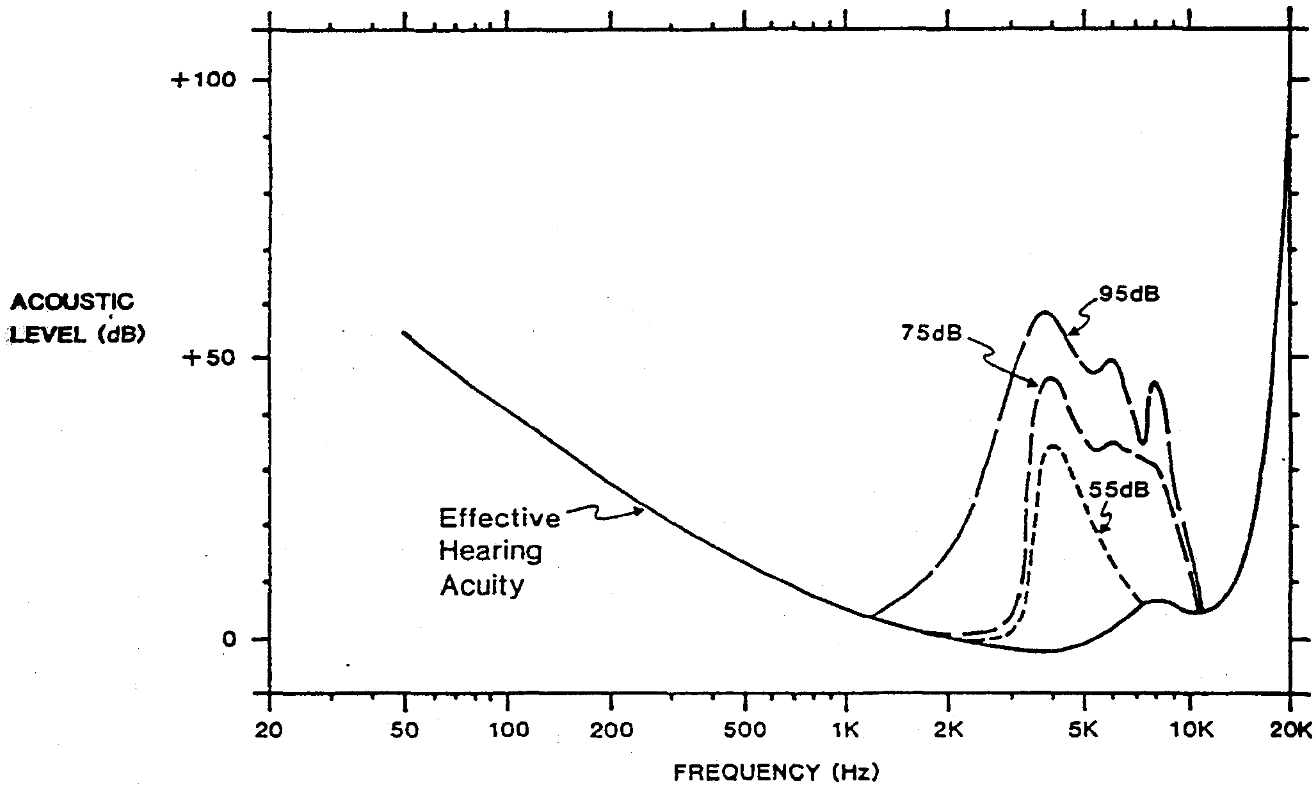

Fig. 3 4 kHz Masked Threshold Curves (0 dB = 20 micropascals)

consequences in coder design, and has been one of the primary reasons that data rate reductions of 4:1 or greater have awaited the practical availability of powerful DSP architectures which can practically implement the necessary complementary filter structures with sharp frequency characteristics that are suitable for music coders.

The differences in the masking characteristics versus frequency are also significant. In Figure 1 the masked threshold falls off only for frequencies above 100 Hz. The upward frequency fall-off in masking above 100 Hz is rapid on a dB per Hz basis, with a slope that is as much as 100 dB per 400 Hz. In the case of 100 Hz masking curves, it is important to note that a ratio of as much as 100 dB may be necessary between the 100 Hz masker and a resultant error component, if the error is to be inaudible. This means that any filter bank used by a rate reduction coder is most effective if its ultimate attenuation spans this 100 dB range. The masking curves of 100 Hz are typical for masking situations for maskers at or below 200 Hz.

The masking curves for 500 Hz, depicted in Figure 2, show a different situation. In this case there is a rapid reduction of downward frequency masking of up to 100 dB per 360 Hz, while having a much slower reduction at higher frequencies. In addition, high sound levels between 90-110 dB cause a very large masking effect at the second harmonic, causing the masking effect to be significantly extended upward in frequency. These 500 Hz curves are typical for the masking properties of midrange signals in the 500 Hz - 2 kHz region. Although not shown, at 2 kHz the slope of the masking curves have only 1/2 - 1/3 the slope of masking curves at 500 Hz, but the total fall-off has increased to 60 dB.

Figure 3 shows masking that is typical for high frequency signals. Masking for lower frequency error components falls off fast but not as fast a dB per rate as in the case of midrange signals. However, the total may exceed 70 dB for maskers at 8 kHz and above. As in the case of midrange signals, upward frequency masking reduces slowly with frequency but covers a more extended frequency range.

1.3 Temporal Masking and Time vs. Frequency Trade-Off

Sinewave masking experiments and the shape of masking curves derived from them indicate the requirements for the filter bank of a low bit rate coder under steady-state signals. Another requirement is the accommodation of human auditory characteristics during transient events. Although the frequency resolution for steady state sinewave signals is extremely sharp, the characteristics of auditory masking for transient events involves time resolutions on the order of a few milliseconds. The temporal characteristics of masking are important because the filter banks used for data rate reduction coders can disperse error signals in time. This spreading occurs because of the fundamental trade-off between temporal and frequency resolution of filters. For this reason, filter bank design typically involves a trade-off between these conflicting goals.

Just as in the case of the frequency characteristics of auditory masking under steady state signal conditions, there is a basic asymmetry in the characteristics of temporal masking. The masking of small signal components occurring during in time before a masker (i.e., backward masking) is substantially less than the forward masking effect in which the same small signals occur after the masker. Backward masking remains strong for about 4 milliseconds and disappears for time separations larger than 10's of milliseconds, while forward masking lasts approximately ten times as long. For further information on the temporal masking characteristics of the ear, see Carterette and Friedman [1978]. The temporal resolution characteristics of a filter bank used for data rate reduction of music signals should maximize the masking effect so that the largest data rate reduction induced errors are tolerated by the ear. Since a transient event can occur anywhere within the effective time window of a particular filter, this argues strongly for filter banks with time resolutions less than 4 milliseconds.

1.4 Filter Bank Design and Auditory Masking

The filter bank of a coder is the primary element that allows rate reduction to occur with minimal audible consequences. It does that by confining the error temporally and spectrally in such a way as to allow the greatest errors to occur. This spectral and temporal confinement must satisfy the following conditions. First, the ideal filter bank should have a frequency selectivity less than one critical band in any part of the audio band, have a fall-off rate of 100 dB per 360 Hz, with an ultimate rejection of 100 dB, and finally, have a temporal spreading effect of less than 4 msec. A filter bank which is easy and efficient to implement is also desirable. Unfortunately, the attainment of all the previously mentioned goals is extremely difficult and a compromise is necessary. As a result, further discussion will concentrate on the compromises and results for the low and moderate time delay AC-2 coders.

The design of the AC-2 coding technology is strongly influenced by the desire to keep the implementation as low in complexity as possible, while preserving coder effectiveness. For this reason, the AC-2 coders use Time Domain Aliasing Cancellation (TDAC), as developed by Princen and Bradley [1986]. This transform has the computational complexity advantages of an FFT and has excellent frequency selectivity characteristics. Unfortunately, the resultant filter bank is constant bandwidth, rather than having the varying bandwidths of the auditory system. This disadvantage of the TDAC can be overcome by approximating the nonuniform bandwidths of the human auditory system by grouping transform coefficients together to form sub-bands with bandwidths approximately that of the auditory system.

Consider first the TDAC filter bank for the moderate time delay coder, useful in applications where a low data rate is more important than low time delay. In this case the transform length is chosen to be 512 samples, which is found to be the best compromise between frequency and temporal selectivity. The resultant filter bank has a frequency selectivity that is sufficient for most of the audio band, while at the same time having a time resolution on the order of 10 msec. This compromise is acceptable since limitations in the temporal or spectral resolution are minor and can be greatly improved by a quantization process that allocates additional data to mitigate the increased audibility of errors during transient circumstances.

The other AC-2 coder is targeted for applications were low time delay is important, such as disk based storage applications requiring fine time resolution editing or for broadcast applications where an announcer may listen to the transmitted signal as a verification of proper system operation. Monitoring of the transmitted voice signal is problematic for the announcer if the time delay is too long, because it interferes with the cognitive process of speaking. The time delay at which speech difficulties begin to occur is not well defined, but 10 msec. appears to be a reasonable compromise, see Gilchrist [1990] for more details. The transform block length for this coder is set at 128 samples by this requirement and the resultant encode/decode delay is 8 msec.

This restriction in the block length has important consequences in the coder design because it moves the filter bank temporal-frequency resolution trade-off away from the optimal compromise. As a result, the frequency resolution is inadequate for masking the error signals for frequencies below 3 kHz. Insufficient frequency selectivity translates to either reduced audio quality or increased data rate. For this reason, this coder uses a higher data rate of 192 kbits/sec per channel. The time resolution of the system is 2.7 msec. and the resultant coder has excellent performance under transient conditions.

The loss of frequency selectivity to satisfy time resolution or computational complexity issues is very important in coder design. Figure 4 demonstrates this point by comparing the filter bank selectivity of three filter banks used in music coders to that of a masking curve for a 100 dB S.P.L. 1 kHz sinewave. This masking curve for 1 kHz was chosen since it is nearly a worst case for the selectivity requirements of a single tone situation. Both

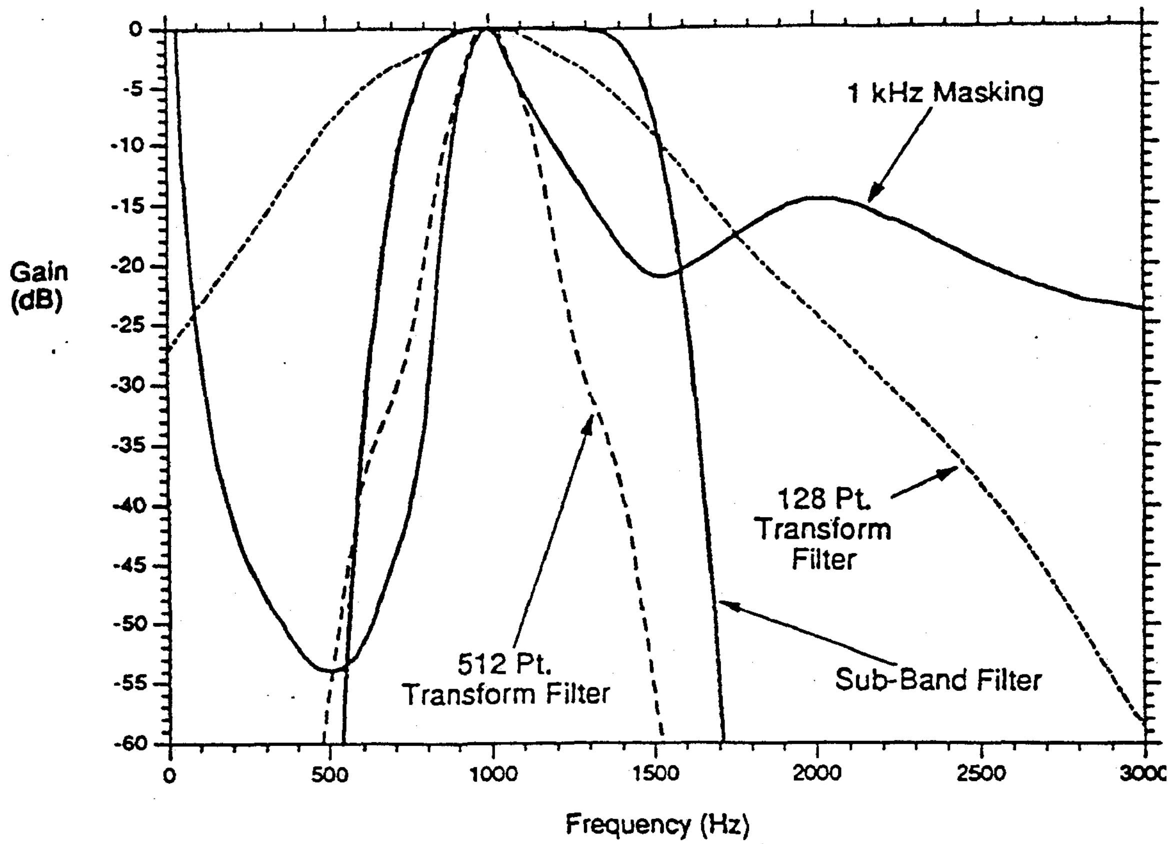

Fig. 4 Comparison Between Various Filter Banks and 1 kHz Human Auditory Selectives

filter banks used in the two AC-2 coder are shown, and in addition, a typical uniform bandwidth sub-band filter band, having 750 Hz bandwidth, is included.

Examination of Figure 4 shows that none of the filter banks presented have ideal frequency selectivity when compared to this most demanding requirement of the human auditory system. The consequences of this fact is that all the coders implemented with these filter banks must either have a higher data rate than ideal or have lower sound quality. Inspection of this figure shows that the moderate delay version of AC-2 has the selectivity closest to that required, implying that little additional data rate is required to preserve sound quality. Next in selectivity is the uniform sub-band filter; the sharpness of the filter roll-off is excellent but it has the limitation that the filter's bandwidth is too wide for low frequency and midrange signals. This lack of frequency selectivity will result in quantization error that is spread over a wide frequency range (i.e., 550-1500 Hz) and must be accommodated by an increase in data rate. This increase in data rate results in an additional word-length requirement because the overall level of the error must be lowered until all of its spectrum lies below the masking curve. Finally, the short time delay AC-2 filter bank frequency selectivity is considered. In this case, additional data rate is seen to be required to mitigate the insufficient frequency selectivity of the low time delay filter bank. This, along with the desire for excellent multi-generation sound quality results in a data rate for this coder of 192 kbits/sec.

In conclusion, the examination of sinewave masking shows that the frequency selectivity of the moderate delay AC-2 is somewhat less than the worst case condition of 1 kHz masking. This indicates that its computationally efficient filter bank does not significantly limit the performance. The low time delay AC-2 coder selectivity is examined and shown to be too broad for use in the lowest possible data rate system. Fortunately, this increase in data rate is modest because the selectivity of the human auditory system is poorer than this filter bank over most of the audio band (i.e., 4 kHz - 20 kHz). One additional benefit of the short time delay AC-2 coder is that it possesses a temporal resolution substantially below that at which either forward or backward masking effects occur. The disadvantage of having too wide a filter bank bandwidth was demonstrated by the 750 Hz sub-band filter example.

1.5 Extension to Complex Signals

The use of simple stimuli masking models has determined the basic requirements of frequency and time resolution. This is done because there is not a widely accepted model of hearing for more complex signals. Unfortunately, real music signals are complex, so coder design must extend these simple masking models to the complex conditions of music signals. In the case of the AC-2 coding systems, simple stimuli masking principles are extended in a very conservative manner. Although many coding systems adaptively allocate most of the available data rate in a signal dependent manner to produce errors that are just below predicted masking, this was found to be an unnecessarily aggressive approach for applications with data rates at or above 128 kbits/sec.

The conservative approach of the AC-2 coder family is as follows: The appropriate TDAC transform filter bank is first combined with a trial quantization process that has a fixed number of bits assigned to each band, which are adjusted to simultaneously satisfy the masking requirements of simple and complex signals. Once this fixed allocation scheme is properly adjusted for optimal audible effect, a modest amount of the data responsible for this representation of the audio signal is removed and replaced by a smaller amount of adaptively allocated data, resulting in 20% or less data of this type. The advantage of the largely non-adaptive nature of most of the data is that problems in the extension of simple masking models are not nearly as serious as in the case of coders that have a more adaptive allocation strategy. This prevents serious audible mistakes from occurring: in fact the audible performances of the AC-2 coders without any adaptive bits are quite good.

This method of extension to more complex signals is evaluated and optimized by both objective and subjective means. This includes comparison of computed noise spectra with psychoacoustic masking threshold data, and conducting A:B listening tests. Subjects are asked to evaluate signals coded by hardware in real-time to facilitate exposing the coder to a wide variety of instrumental, vocal, and synthetic audio signals.

Although coder performance is more rigorously evaluated using complex music signals, many important features are revealed by the sinewave error spectrum. Figures 5 and 6 are a comparison of both coder's 1 kHz error spectra with a 100 dB S.P.L., 1 kHz masking curve. The moderate delay AC-2 coder results are shown in Figure 5 and those of the low delay AC-2 coder in Figure 6. Both figures give an indication of the worst case performance of the coder because the 1 kHz auditory selectivity is the most severe. These comparisons assume a consumer playback sound level at 108 dB peak acoustic level, being limited by the maximum loudness capabilities of typical home loudspeakers and amplifiers. In both figures, the error spectra are shown for coder operation with, and without, the adaptively allocated portion of

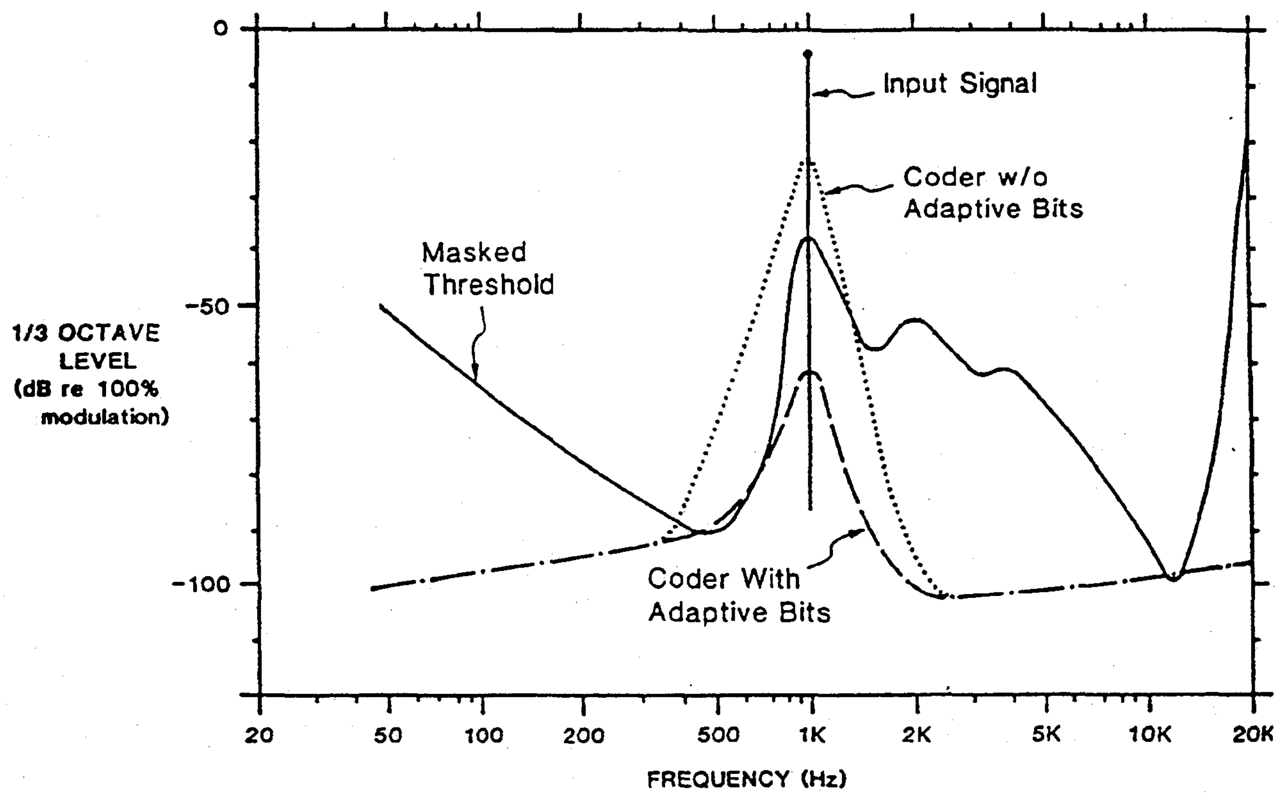

Fig. 5 Moderate Time Delay AC-2 Coder Performance with the Application of a 1 kHz Sinewave

the data. Both coding systems are interfaced to 16-bit ADC's and DAC's so the noise of the conversion process also is present.

Examination of Figure 5 shows that the error signal and converter noise under normal operation is just at or below audibility. The error spectra above 2.5 kHz for both situations are limited by the ADC/DAC noise floors and indicative of the 91 dB dynamic range of typical 16-bit conversion systems. The frequency region below 2.5 kHz is a result of coder operation and a significant deviation from an ADC/DAC noise floor results. In this region, the 1 kHz error spectrum under normal operation is substantially below the masked threshold curve, except for frequencies between 400 Hz - 700 Hz, where the error spectrum is comparable to the masked threshold. This indicates that a slight modulation noise may be audible, although in practice this has not been heard. The error spectrum shown without the adaptive portion of the data shows that modulation noise is now quite audible since the error spectrum is significantly above the masked threshold in the frequency range of 400 Hz - 1200 Hz. The generation of audible modulation noise indicates that the adaptive bit allocation process is necessary to preserve excellent sound quality. Notice that the error spectrum falls off less rapidly than that of the downward frequency portion of the 1 kHz masking curve. This is exactly as predicted by the earlier discussion of the requirements of filter bank selectivity.

Figure 6 shows the same comparison for the low time delay AC-2 coder. In this case, the audio performance is essentially noise-free in normal operation, but limited by modulation noise without the adaptively allocated bits. This time the coder dependent part of the spectrum extends to 5 kHz and the extension of the range where the coder affects the noise spectrum is due to the more gradual frequency selectivity of the low time delay filter bank. Similar to the case of the moderate delay AC-2 coder, the normal operation spectrum slightly exceeds the masked threshold curve in the region of 400-600 Hz, indicating the presence of a small amount of masking noise. As before, actual listening tests determine that no modulation noise is audible. The use of adaptive bits is shown to be important since the situation with no adaptively allocated bits indicates the presence of substantial modulation noise. In this case, the error spectrum exceeds the masked threshold by 25 dB at 500 Hz.

2 AC-2 CODING ALGORITHM

In Section 1, some of the groundwork for audio coder design was established. In this section, we build upon

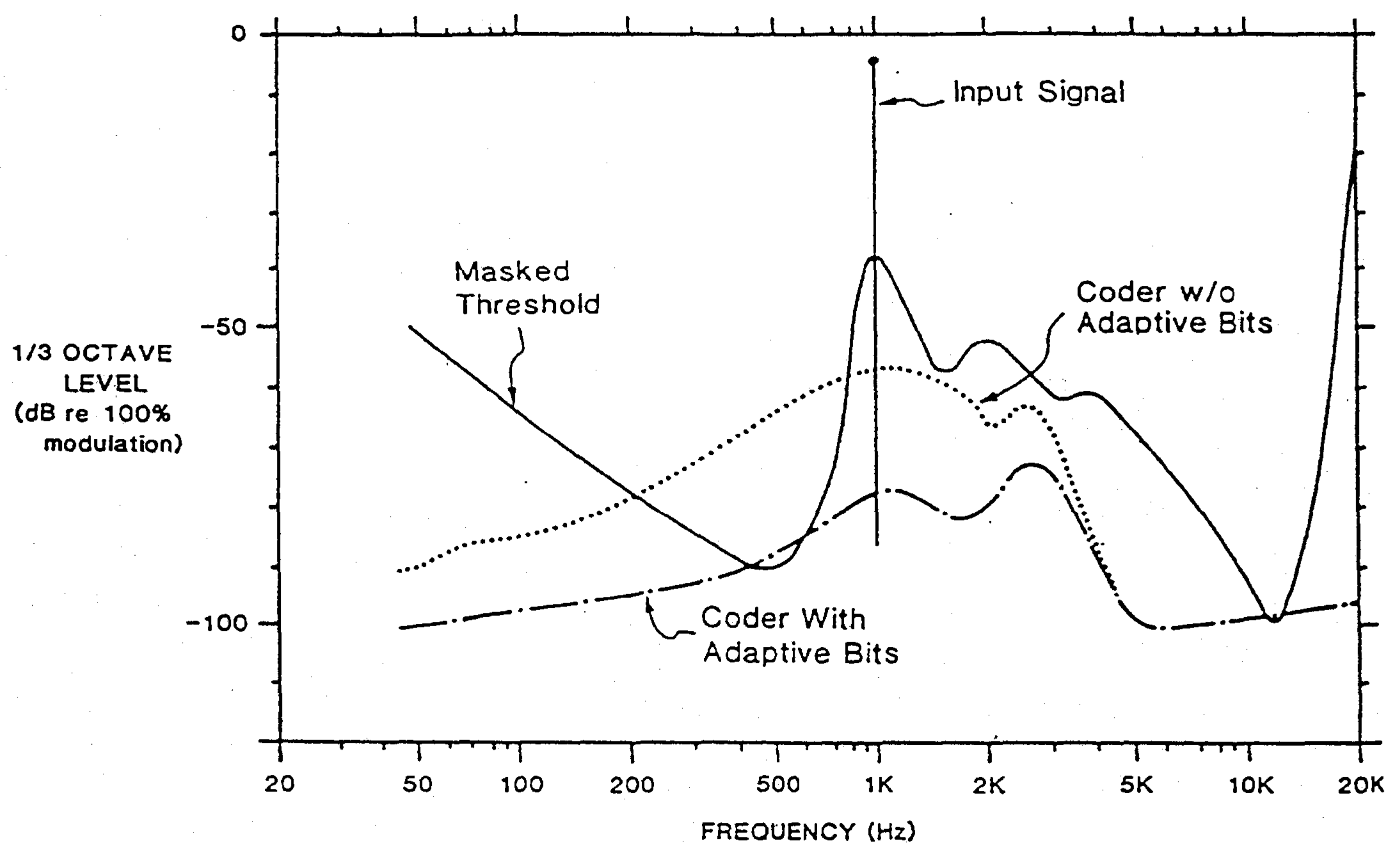

Fig. 6 Low Time Delay AC-2 Coder Performance with the Application of a 1 kHz Sinewave

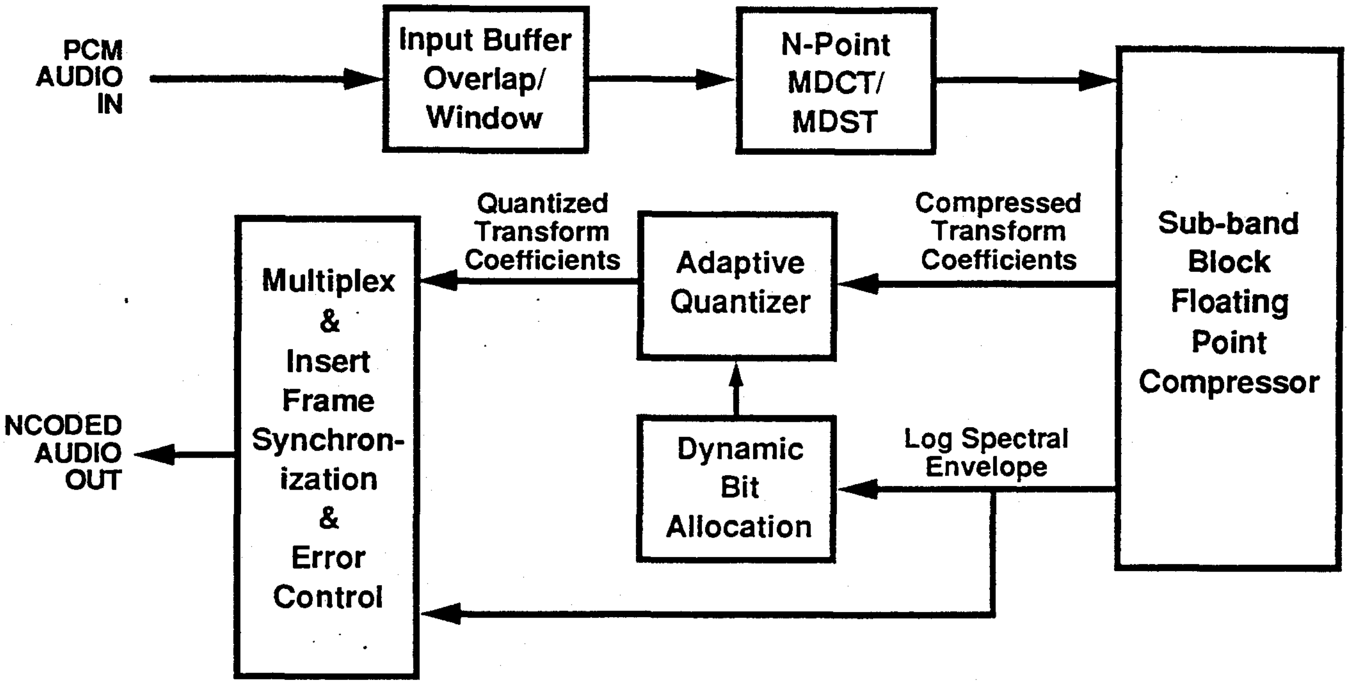

Fig. 7 AC-2 Digital Audio Encoder Family Block Diagram

this presentation by exploring the AC-2 coding algorithm in more detail. The description generally applies to all members of the AC-2 family; differences between the low and moderate delay versions are described where appropriate.

Figure 7 presents the block diagram of a generic AC-2 digital audio encoder. In the first stage of processing, PCM audio is buffered into frames of length N samples. Each new frame overlaps the previous one by 50%, i.e., the first N/2 samples in each frame are comprised of the last N/2 samples from the previous and present one. Consequently, each input sample is contained within exactly two consecutive frames. Next, the buffered samples are multiplied by a window function to reduce the effect of frame boundary discontinuities on the spectral estimate provided by the transform. The window also significantly improves the frequency analysis properties of the encoder.

The time-to-frequency domain transformation is based on evenly- stacked TDAC, consisting of alternating Modified Discrete Cosine (MDCT) and Modified Discrete Sine (MDST) transforms. A crucial advantage of this approach is that 50% frame overlap is achieved without increasing the required bit-rate. In a critically- sampled analysis technique such as TDAC, exactly N unique nonzero transform coefficients are generated on the average in an interval of time representing N input PCM samples. In TDAC, each MDCT or MDST transform of frame size N generates only N/2 unique nonzero transform coefficients, so critical sampling is achieved with 50% frame overlap. Any nonzero overlap used with conventional transforms (such as the DFT or standard DCT) precludes critical sampling, since each N-point transform generates N unique nonzero transform coefficients. Additionally, several memory and computation-efficient techniques are available for implementing the MDCT and MDST transforms.

TDAC is applied to model the auditory system by grouping adjacent transform coefficients into sub-bands for further decomposition and analysis. The number of coefficients per sub-band is computed a priori to approximate the nonuniform critical-bands. Transform coefficients within one sub-band are converted to a frequency block floating-point representation, with one or more mantissas per exponent, depending upon the sub-band center frequency. Each exponent represents the quantized peak log-amplitude for its associated sub-band. The exponents collectively provide an estimate of the log-spectral envelope for the current audio frame, computed on a critical-band frequency scale.

From a psychoacoustic perspective, the log-spectral envelope provides an ideal framework for estimating which sub-bands of a given audio frame are perceptually most relevant, and for ranking them in relative order of importance for dynamic bit allocation. Furthermore, the nonuniform frequency division scheme offers key advantages compared to one based on uniform-width filter banks. Accordingly, the AC-2 frequency division scheme reduces the need both for relying upon a complex masking model, and for using a second, higher-resolution filter bank in the encoder.

The dynamic bit allocation routine is completely feed-forward in nature and is constrained to produce a constant bit-rate as required for transmission applications. Bits are allocated in accordance with a set of deterministic rules derived from conservative use of single-tone masking curves. A portion of the routine employs a water-filling procedure in which sub-bands are ranked and allocated bits on a band-by-band basis.

The allocation routine provides step-size information for an adaptive quantizer. Each sub-band mantissa is quantized to a bit resolution defined by the sum of a fixed allocation and a dynamic allocation. The total fixed allocation for one frame outweighs the dynamic allocation in approximately a 4:1 ratio. For a given level of error protection overhead, this approach was found to provide more robust coding and error performance, since the number of most-significant mantissa bits is known a priori in the decoder.

In the final stage of the encoder, exponents are multiplexed and interleaved with mantissa bits for transmission to the decoder. Optional error correction codes may be added at this step. The amount of overhead information reserved for error control coding can be adjusted to give greater or lesser protection depending upon channel error performance for a given application.

Serial bitstream formats can be optimized for the application. In the DP501/DP502 digital audio encoder/decoder products employing AC-2, two independent channels are interleaved in a regular pattern of alternating 16-bit segments. This format allows for straightforward demultiplexing of the encoder bitstream into separate channels, and for recombining monophonic bitstreams from different encoder units. Provision is also made for the insertion of a 1200 bit/s auxiliary data stream, algorithm identification bits, ADC overload status, and other information.

In the AC-2 decoder, shown in Figure 8, the input bitstream is demultiplexed and errors, if any, are corrected. The received log spectral envelope is processed in a stage identical to the encoder bit allocation routine, which generates step-size information for the adaptive inverse quantizer. The fixed and dynamically-allocated portions of each mantissa are concatenated to regenerate compressed transform coefficients. A sub-band block floating-point expander then linearizes the compressed transform coefficients and passes them to an inverse MDCT/MDST transform stage. After the inverse transformation, a window identical to that used in the encoder is used to post-multiply the reconstructed time-domain samples for each frame. Adjacent windowed frames are overlapped by 50% and then added together to reconstruct the PCM output.

Total coding/decoding time delay is determined by the frame size N, the manner in which frames are processed, and the processor speed. In the low-delay coder, input

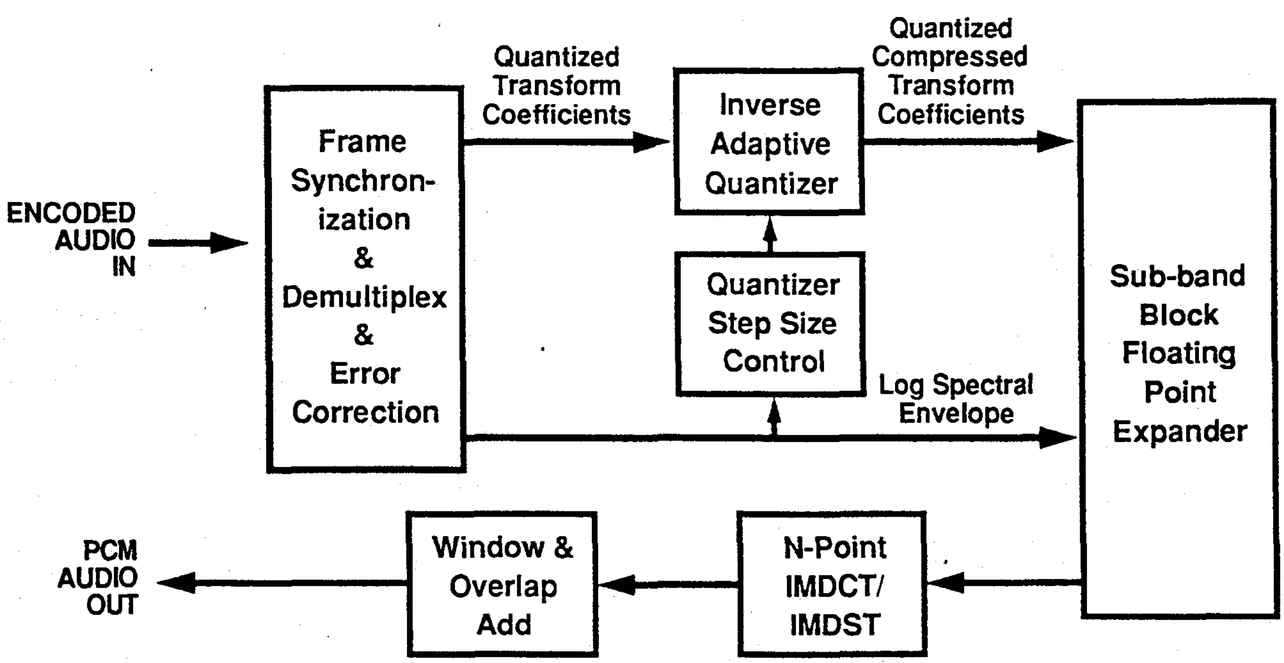

Fig. 8 AC-2 Digital Audio Decoder Family Block

frames are processed one-by-one, resulting in a theoretical minimum total coding and transmission delay of 2.5N samples when employing infinitely-fast encoder and decoder processors. In actual practice, the delay increases to about 3N samples for fully-utilized (finite-speed) encoder and decoder processors. With a frame size of N = 128 samples and a sample rate of 48 kHz, a delay of 8 msec. is obtained. In the moderate-delay coder, two successive frames from one channel are buffered and processed jointly. In this case, the total delay when using fully-utilized processors is about 4N samples, which results in less than 45 msec. of delay at a sampling rate of 48 kHz and with N = 512.

3 HARDWARE IMPLEMENTATION

All coders within the AC-2 family have been optimized for very low hardware implementation cost. By today's standards, cost is ultimately measured by the die size and package cost of a custom VLSI implementation. Accordingly, the cost equation must not only include such traditional complexity measures as multiply-add count and RAM/ROM memory usage, but regularity of computation and minimum word-length requirements as well. Considerable attention has been given to structuring the computations in AC-2 to minimize VLSI implementation cost and simultaneously achieve the audio performance objectives. At a sampling rate of 48 kHz, the total number of multiplies and adds per second in a stereo AC-2 encoder is about 2.7 million. The decoder complexity is slightly lower. This compares to calculations by Reader [1991], estimating a total of about 35 million multiplies and adds per second for a straightforward implementation of a current generation sub-band encoder, and 16 million multiplies and adds per second in the decoder.

The low computational complexity can be attributed to several factors. First, the computational structures employed are highly regular in nature. Second, an efficient technique has been found for implementing the evenly-stacked TDAC transform by combining a core FFT routine with pre-twiddle and post-twiddle operations. Third, the nonuniform frequency division stage and log spectral energy representation enables the use of a low-complexity dynamic bit allocation routine. Finally, the use of functions which are inefficiently implemented on programmable DSPs or in custom-ICs, such as logarithms, square roots, and divides, have been found unnecessary. The only functions required are multiply, add, integer left/right shift, normalize, and compare.

3.1 General-Purpose Programmable DSPs

Programmable DSPs provide a flexible and expedient path to real-time algorithm development, and as such provide an attractive means for a first implementation. An early embodiment of AC-2 based on the Fourier transform was implemented using six Texas Instruments TMS32010s by Fielder [1989]. This work subsequently led to an implementation employing TDAC and based on the Motorola DSP56001, as detailed by Davidson et al. [1990]. In the latter case, a single 27 MHz chip could either encode or decode two independent channels. Recent improvements in software run-time efficiency have reduced this speed requirement to 20 MHz.

3.1.1 24-Bit Fixed-Point

Since it's inception in 1987, the Motorola DSP56001 has proven to be a capable platform for implementation of a wide variety of audio processing algorithms. This general trend has been supported by several audio compression implementations, including AC-2. The DSP56001's 24-bit data path, flexible addressing modes, and dual-accumulator arithmetic logic unit (ALU) are keys to its successful application in audio.

In particular for AC-2, we found that the 24-bit word-length was sufficient for all arithmetic tasks. Furthermore, no elaborate scaling or rounding procedures were required. The dynamic range of the implementation, as measured from PCM input to output, is 108 dB. This figure greatly exceeds the theoretically-achievable dynamic range of 16-bit ADC and DAC converters, and is commensurate with next-generation 18 and 20-bit converter technologies.

One of the more time-intensive processing blocks of those shown in Figures 7 and 8 is the inverse transform, which requires about 18% of the total DSP processing time. Surprisingly, however, the most time-intensive tasks are bit multiplexing and demultiplexing. This indicates that a custom IC could save significant ALU resources compared to a DSP if dedicated logic performed the multiplexing and demultiplexing. This topic is discussed further in Section 3.2.

3.1.2 16-Bit Fixed-Point

A study was made to determine the feasibility of implementing an AC-2 decoder on a 16-bit DSP chip. The motivation for this work was to identify a lower-cost platform for the implementation of an AC-2 decoder, while maintaining the flexibility of a programmable DSP. Our results indicate that current generation 16-bit DSPs, such as the Texas Instruments TMS320C5x, Analog Devices ADSP-2105, and Motorola DSP56116, are sufficiently powerful to implement a single-chip stereo encoder or decoder.

An analysis of finite word-length effects was conducted in part by modifying the real-time AC-2 DSP56001 software to emulate a reduced word-length processor. The data word-length was selected on-the-fly with

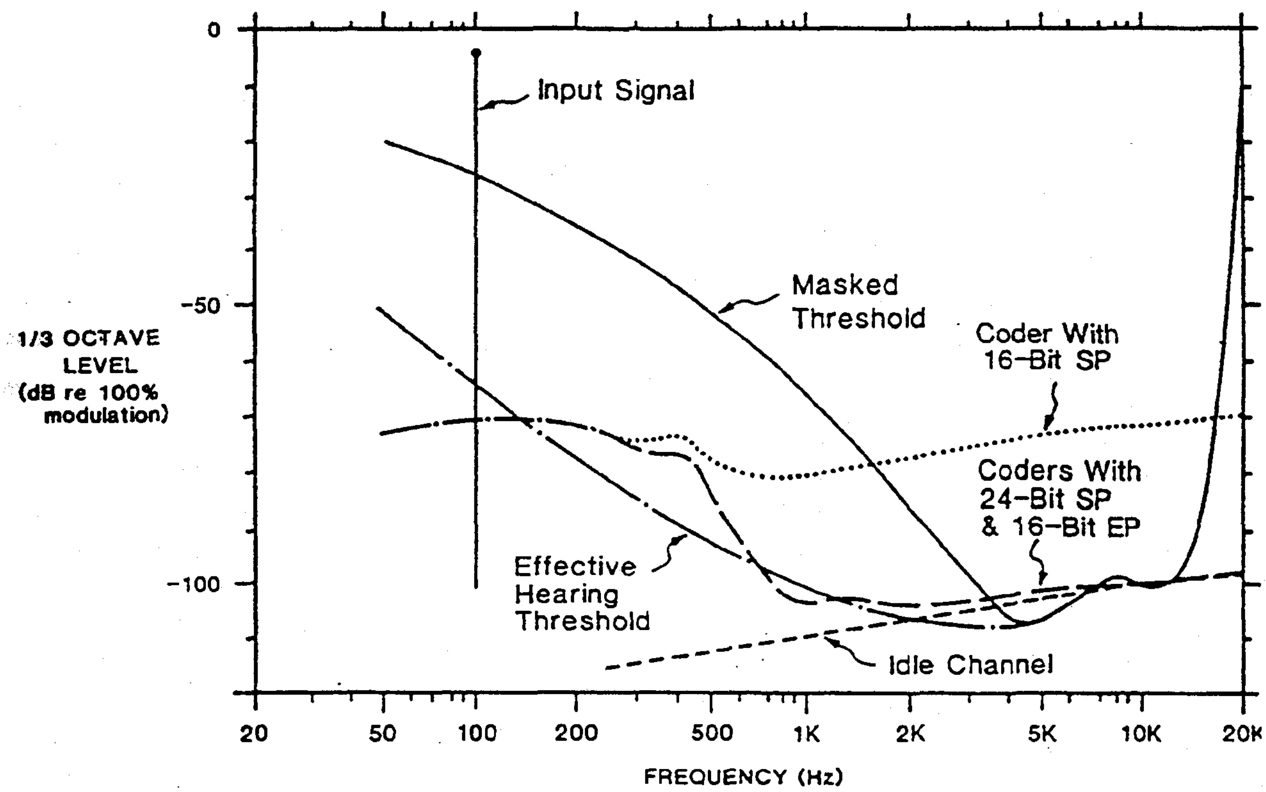

Fig. 9 Comparison of Filter Bank Arithmetic Noise of Single Precision (SP) 16-bit Arithmetic Versus Extended Precision (EP) 16-bit and Single Precision 24-bit Arithmetic

switches. Coefficient word-lengths could also be varied. This approach allowed us to independently adjust, and jointly minimize, the data and coefficient word-lengths in each processing stage of the coder. The real-time variable word-length simulation served as a valuable tool for rapid objective and subjective evaluation of finite precision arithmetic effects.

Figure 9 presents a plot of the spectral error between an original and a coded 100 dB S.P.L., 100 Hz sinewave as processed by both 24-bit and a 16-bit ALUs in the decoder. Results from the moderate-delay decoder are shown since arithmetic round-off noise in the inverse transform is highest for long frame lengths; round-off noise in the low-delay coder is more than 6 dB lower. The idle channel noise produced by 16-bit ADC and DAC converters is included to show when the coder is limited by the conversion process. The low frequency sinewave represents a demanding test signal since minimal masking of the 4 to 6 kHz region occurs, where the ear's hearing threshold is low.

At frequencies below 500 Hz, noise introduced by transform coefficient quantization dominates arithmetic round-off noise. This region is perceptually insignificant because both noise curves are below the masking curve. Above 2 kHz, round-off noise for the 16-bit ALU significantly exceeds the masking curve, indicating that 16-bit single-precision (SP) arithmetic is inadequate.

Most of the noise shown in Figure 9 is generated during the inverse FFT computation of the inverse MDCT/MDST transform computation. Therefore, conventional techniques for reducing round-off error in fixed-point FFTs apply, such as those described by Meyer [1989]. We found that the combination of dynamic scaling between IFFT stages, optimal rounding, and optimal placement of quantizers in the butterfly produced a significant, but still insufficient, reduction in round-off noise. Furthermore, such techniques may impose a three-fold increase in IFFT butterfly computation time within a general-purpose DSP.

Based on these results, a preferred approach is to employ an extended-precision (EP) scheme based on 16 x 32-bit multiplies, which for many 16-bit DSPs results in a fixed two-fold increase in butterfly computation time, and provides a digital noise floor which is more than 40 dB lower than that obtainable with 24-bit SP multiplies. All other processing stages of the decoder can be implemented with 16-bit SP arithmetic. The minimum required DSP clock speed using 16-bit EP is only about 18% higher than the equivalent rating for a 24-bit fixed-point or 32-bit floating-point device.

3.2 Full-Custom VLSI

In order for an audio processor to be utilized in high volume applications, the device cost must usually be low. Since programmable DSP chips frequently contain more hardware logic than required for a given application, we have considered the design of a special-purpose VLSI architecture for implementing an AC-2 decoder. Our findings indicate that a stereo decoder can be implemented on a die containing approximately 7,000 gates, plus ROM and RAM. The architecture is capable of implementing any of the coders in the AC-2 family with one IC.

The architecture consists of three sections: a bit demultiplexer, a quantizer step-size control, and an inverse transform and reconstruction processor. The chip inputs are a serial bitstream and data clock, and the output is one or more 20-bit PCM digital audio channels. The bit demultiplexer performs such functions as data de-scrambling and bit de-interleaving. The demultiplexer directs the unpacked exponent data to the quantizer step size control, and the unpacked fixed and adaptive mantissa bits to a dedicated state machine/barrel shifter. The quantizer step size control, composed of a simple programmable microcontroller, processes incoming exponents and directs the state machine and barrel shifter to concatenate fixed and adaptive transform coefficient mantissa bits. The reconstruction processor performs either an IMDCT or IMDST, producing one frame of PCM samples. These samples are then windowed and overlap/added with the previous windowed block of PCM data to reconstruct audio samples. Since the multiply-add rate of the audio synthesis stage is quite low, a bit-serial multiplier has been employed. The serial multiplier requires significantly less chip area than a single-cycle array multiplier of the same word-length.

4 CONCLUSIONS

Adaptive transform coding of audio signals with AC-2 technology offers a high-quality, low complexity approach for data rate reduction of professional grade audio. Two 20 kHz bandwidth examples of the AC-2 coding family have been discussed, providing 4:1 and 6:1 bit-rate compression at low and moderate time delays, respectively. The excellent sound quality and computational ease of implementation of the AC-2 technology make it a natural candidate for broadcast, computer multimedia, and digital storage applications. The 128 kbits/sec. data rate of the moderate delay coder make it very appropriate for Digital Audio Broadcast and High Definition Television applications. The low delay coder is optimized for music material contribution applications (i.e., studio to transmitter and contribution quality links) requiring excellent multi-generational sound quality and a time delay acceptable for off-air monitoring during voice announcing.

The performance of these systems has been quantified by examination of simple stimuli masking models which have been the driving force shaping the design of the employed filter bank structures. Sinewave masking models have been used because a comprehensive and complete model for complex signals is not widely agreed upon. As a result, extension of the simple models is necessary for the design of practical coding systems. It was shown that the AC-2 family used a conservative extension process which resulted a relatively small amount of adaptively allocated data. As a consequence, these coder techniques were robust with respect to difficult program material. Other benefits created by this approach were a relative insensitivity to the effects of data-stream errors and low computational complexity.

Issues of computational complexity and practical implementation were discussed in some detail. It was shown that the AC-2 coder family is straightforward to implement at 128 and 192 kbits/sec. In particular, implementation of a stereo encoder or decoder was readily accomplished in one 20 MHz Motorola DSP56001. It was also shown that a practical modification of the frequency division algorithm permitted the realization of full fidelity realizations on 16-bit fixed-point DSP chips. A custom approach was also presented. It was shown that the AC-2 algorithms lend themselves well to dedicated chip hardware because of their reliance on simple shift operations and a low-complexity bit allocation strategy. An entire stereo encoder or decoder can be implemented with a complexity of approximately 7000 gates, plus ROM and RAM.

In conclusion, the AC-2 coder family represents one of the most cost effective solutions to very high-quality music coding applications at a 4:1 to 6:1 compression ratio. Although only two coders with data rates of 128 and 192 kbits/sec. were discussed, this technology can be applied to other sample rates, lower data rates (i.e., 64 kbits/sec.), and other signal bandwidths as well.

5 REFERENCES

K. Brandenburg, H. Gerhaeuser, D. Seitzer, and T. Sporer, "Transform Coding of High-Quality Digital Audio at Low Bit Rates: Algorithms and Implementation," Proc. of IEEE Intl. Conf. on Comm., Atlanta, pp. 932-936, April 1990.

E. C. Carterette and M. P. Friedman, Handbook of Perception, pp. 305-319, Academic Press Inc., New York, 1978.

G. Davidson, L. Fielder, and M. Antill, "Low-Complexity Transform Coder for Satellite Link Applications," presented at the 89th Convention of the Audio Engineering Society, preprint 2966, Sept. 1990.

R. H. Ehmer, "Masking of Tones vs. Noise Bands," J. of the Audio Eng. Soc., vol. 31, pp. 1253-1256, Sept. 1959.

L. D. Fielder, "Evaluation of the Audible Distortion and Noise Produced by Digital Audio Converters," J. Audio Eng. Soc., vol. 35, pp. 517-535, July/August 1987.

L. D. Fielder, "Low-complexity Transform Coder for Music Applications," presented at the IEEE ASSP Workshop on Appl. of Sig. Proc. to Audio and Acous., New Paltz, New York, session 2- 1, Oct. 1989.

H. Fletcher, "Auditory Patterns," Reviews of Modern Physics, vol. 12, pp. 47-65, Jan. 1940.

N. H. C. Gilchrist, "Delay in Broadcasting Operations," presented at the 90th Convention of the Audio Engineering Society, preprint 3033, Feb. 1991.

J. D. Johnston, "Transform Coding of Audio Signals Using Perceptual Noise Criteria," IEEE J. on Sel. Areas in Comm., vol. JSAC-6, no. 2, pp. 314-323, Feb. 1988.

R. Meyer, "Error Analysis and Comparison of FFT Implementation Structures," Proc of IEEE Conf. on Acous, Speech and Sig. Proc., Glasgow, pp. 888-891, 1989.

J. P. Princen and A. B. Bradley, "Analysis/Synthesis Filter Bank Design Based on Time-domain Aliasing Cancellation," IEEE Trans. on Acous., Speech, and Sig. Proc., vol. ASSP-34, no. 5, pp. 1153-1161, Oct. 1986.

C. Reader, "Comments on Audio CD and Analysis of Audio Complexity," ISO/MPEG Document ISO/IEC/JTC1/SC2/WG11/MPEG91/082, May 20, 1991.

E. F. Schroeder, H. J. Platte, and D. Krahe, "MSC: Stereo Audio Coding with CD-Quality and 256 kbits/sec," IEEE Trans. on Cons. Elec., vol. CE-33, no. 4, pp. 512-519, Nov. 1987.

G. Stoll and Y. F. Dehery, "High-Quality Audio Bit-Rate Reduction System Family for Different Applications," Proc. of IEEE Intl. Conf. on Comm., Atlanta, pp. 937-941, April 1990.

E. Zwicker, "Subdivision of the Audible Frequency Range into Critical Bands," J. of the Acous. Soc. of Am., vol. 33, no. 2, p. 248, Feb. 1961.

E. Zwicker, G. Flottorp, and S. S. Stevens, "Critical Band Width in Loudness Summation," J. Acous. Soc. of Am. vol. 29, pp. 548- 557, May 1957.

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Design Considerations for Digital STL Applications

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DESIGN CONSIDERATIONS FOR DIGITAL STL APPLICATIONS

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Introduction

There has been a growing interest in the use of digital data transmission techniques to improve broadcast quality, both in terms of improving audio signal quality as well as improving signal robustness. Digital broadcasting systems such as STLs can benefit from the digital audio technology developed for recording and reproduction. This allows a dynamic range of at least 90 dB to be obtained and avoids certain distortion mechanisms. In addition, once analog audio signals have been coded into digital form, there is no further loss of quality when the digits are transmitted through a channel that meets a certain carrier-to-noise ratio (CNR) requirement. This requirement varies with the specific digital system employed, but never approaches the CNR requirement of 50+ dB for even mediocre analog transmissions. This very high CNR requirement makes analog channels highly susceptible to interference and subsequent audio degradation. Analog systems gradually become worse and worse in quality as the channel slowly deviates from the ideal noise-free channel. This is unlike properly designed digital systems that maintain uniformly good quality over a much wider range of conditions.

With a conventional FM STL, (a composite system is used as an example throughout this paper, but the principles apply to discrete systems as well) performance varies in proportion to the available carrier-to-noise ratio (CNR). Performance of the best systems can be quite good under ideal CNR conditions and results in an audio link with a dynamic range of about 80 dB. Unfortunately, real world conditions, with their lower CNR's, typically result in links with more modest dynamic range of about 70 dB; RF path fading or interference can cause the audio SNR to drop to only 45 dB (at which point the squelch circuits are probably activated).

On the other hand, properly designed digital links fare substantially better, having audio performance that is superior to an FM STL operating under the best of circumstances, and possess a much higher tolerance to RF path fading and interference. Digital links have a dynamic range of 90 dB or better and maintain excellent sound quality even, for example, under fades as deep as 30 dB. This allows the squelch level to be set lower for a digital system while allowing virtually ideal sound quality. Figure 1 is an illustration of the sound quality as the RF channel degrades in quality. The figure compares the performance of the aural FM STL to a digital STL system.

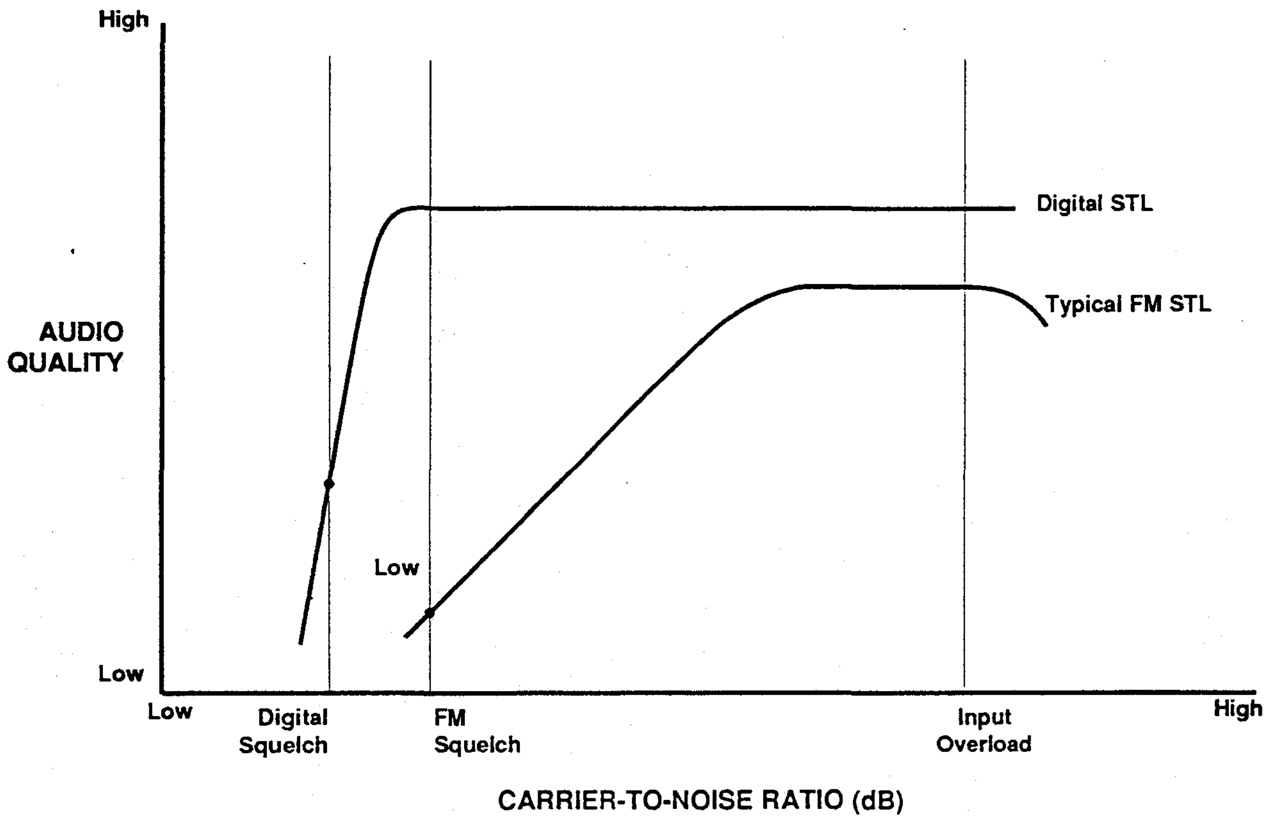

Figure 1: RF Carrier to Noise Ratio vs. Audio Quality Comparison for Typical FM and Digital Studio to Transmitter Links.

This figure shows the gradual improvement in FM STL audio quality with increasing CNR. This occurs because the audio SNR is directly proportional to the CNR until full quieting is achieved; at this point, the audio performance is at its best. As the RF signal strength continues to increase, eventually the input circuitry of the receiver enters overload and the audio quality starts to deteriorate. The best of present FM STLs are satisfactory audio links in the CNR region where the STL is in full quieting, producing an audio SNR as high as 80 dB. Unfortunately, even under these conditions, audio quality degrades for short periods of time due to RF fading effects. RF signal level fading occurs for a variety of reasons, and STLs occasionally experience conditions when the CNR degrades substantially. Under these conditions, even an FM STL which normally has a SNR of 80 dB may fall to below 60 dB or worse. Perhaps more insidious, especially in major metropolitan areas, interference from other STLs can degrade performance. The interference essentially degrades the CNR of the STL path, thus degrading audio performance.

Consider now the digital STL system. In this case, the system can be designed to have a quality comparable to the Compact Disc, with its 90 dB dynamic range, over most of the possible range of RF signal strengths. This extension of the region of optimum operation is attributable to the digital systems' more modest requirements for RF channel CNR's. In fact, a properly designed digital STL can accommodate a CNR range of 50 or 60 dB with no degradation in quality. When failure finally results, due to insufficient RF CNR, it occurs more rapidly than for the FM case, but can be designed to occur at much lower CNR's than the point where FM STLs are in squelch. Thus, a digital system is more robust in the face of fade or interference.

Basic STL Building Blocks

The design and construction of a digital STL system is inherently different from the classical FM STL that has served the radio industry for so many years. In fact, the practical implementation of such a system relies on the existence of integrated circuits that were unavailable or prohibitively expensive even a few years ago.

The block diagram of a conventional aural STL transmitter is shown in Figure 2. An FM STL transmitter consists of an audio section followed by an FM modulator (typically operating at some IF frequency), and an RF section which translates the IF signal up in frequency and increases its power. In the receiver, there are complementary circuits: the RF front end amplifies the incoming signal and converts its frequency to the IF frequency range, the demodulator recovers the audio signal, and the audio section provides proper signal conditioning for connection to subsequent equipment in the broadcast chain. Great care must be taken in the design of such a system to ensure that the entire signal path is free of noise and distortion.

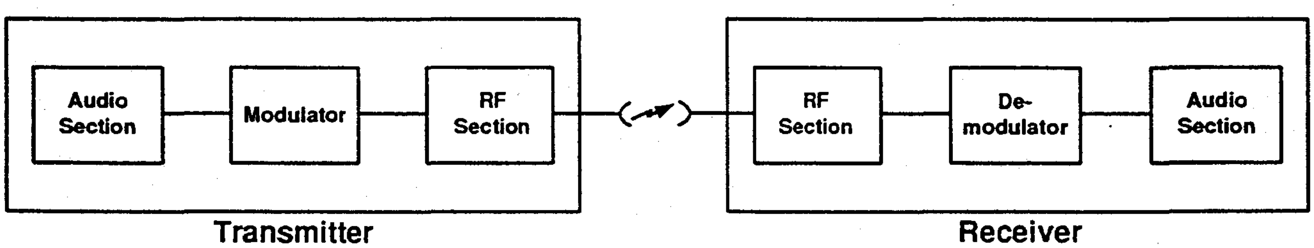

Figure 2: The Major Components of an FM STL.

The block diagram for a digital STL is more complex, as shown in figure 3. A digital STL transmitter is comprised of an analog to digital conversion stage, an audio coding system which is used to reduce the digital data rate (this topic will be discussed at length later in the paper), followed by a digital modulator that converts the digital data stream into a narrow-band IF signal, and then an RF section that translates the IF signal up in frequency and provides power amplification. In the receiver, there are complementary circuits. The RF front end amplifies the weak incoming signal and drops its frequency to IF, the demodulator converts it to a digital data stream, the audio decoder converts the data stream back into standard digital audio, and finally the digital to analog conversion section converts the data stream into analog audio.

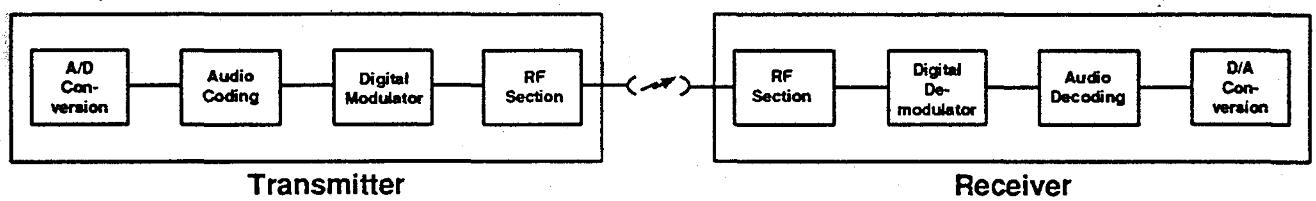

Figure 3: The Major Components of a Digital STL.

The penalty for the improved performance of a digital STL is its increased complexity. Fortunately, technologies developed for other applications can be applied. For instance, digital audio conversion technology has been developed and perfected for the Compact Disc and digital audio recording devices. Digital modem and RF technology has been developed for other communications applications. The one area that is relatively new and that has been necessary to develop for this application is the audio coder technology. Audio coder systems are necessary for the development of practical digital STLs, particularly in the 950 MHz range, because they allow the digital audio bit rate to be reduced to rates which result in a spectrally efficient digital STL. Dolby Labs has done considerable work in this area over the last 7 years and has applied this technology to designing a practical digital STL. The properties, structure, and limitations of low bit-rate audio coder technology will be discussed shortly.

Digital Modulation

Various types of modulation schemes exist to transmit digital audio. They differ in their degree of spectral efficiency: given a certain digital audio data rate, what is the resultant occupied bandwidth? Another important variable in the design of digital modulation schemes is the required CNRs that results in error free digital transmission. Ideal systems have the best combination of spectral efficiency and tolerance to poor CNRs.

The spectral efficiency of various digital modulation schemes is described in bits/sec per Hertz of bandwidth that the RF signal occupies. More efficient systems require less RF bandwidth to send a given data rate than others. Digital modulation systems also require a minimum CNR to transmit data below a given error rate. As the channel CNR decreases, the error rate goes up very rapidly; a decrease in CNR of 1 dB results in an error rate increase of 10:1. Figure 4 shows the performance of a number of digital modulation schemes, (having no implementation losses) and the theoretically derived Shannon limit, which represents the best that communication systems can do. This figure comes from an excellent book on digital communications by Feher:

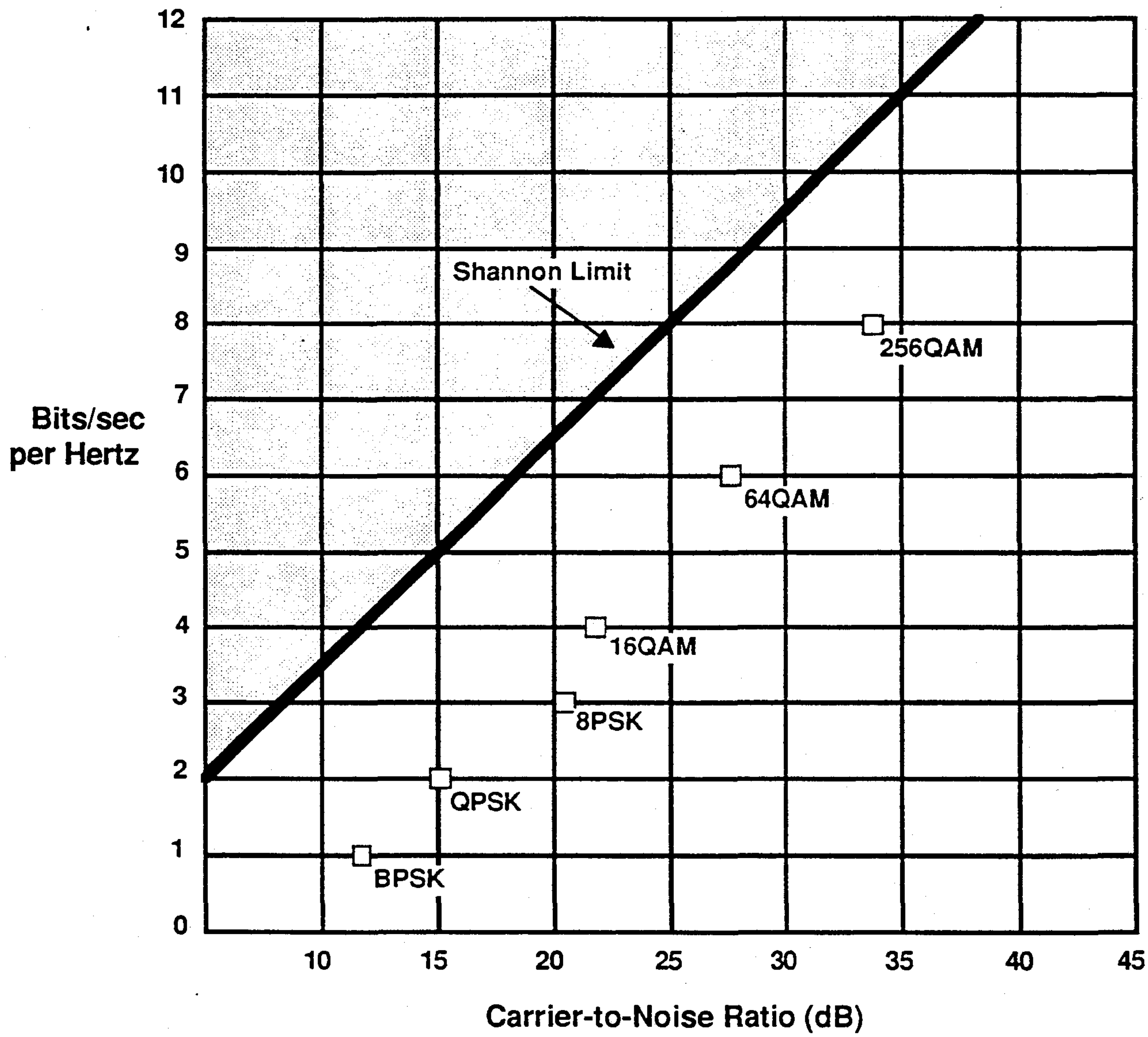

Figure 4: Spectral Efficiency vs. Carrier-to-Noise Ratio for Various Digital Modulation Schemes.

Figure 4 shows the properties of ideal implementations of binary phase shift keying (BPSK), quadrature phase shift keying (QPSK), 8 phase shift keying (8PSK), 16 state quadrature amplitude modulation (16QAM), 64 state quadrature amplitude modulation

(64QAM), and 256 state quadrature amplitude modulation (256QAM) digital modulation systems. These are listed in order of increasing complexity. The CNR requirements are those that result in an error rate of 10E-8 (one error in 100 million bits of transmitted data). The performance of the various modulation systems are compared to the Shannon limit and found to be approximately 8 to 12 dB worse than predicted by information theory. Although an 8 to 12 dB loss seems excessive, the modulation techniques shown nevertheless can be employed in practical digital systems.

The first 3 modulation schemes are the simplest of the group and represent binary information as either 2, 4 or 8 phase states in an RF signal. The complexity of these three systems is relatively low, and they result in fairly robust digital modulation systems. The second group of 3 are more complex but more spectrally efficient systems that encode the audio data by demultiplexing the data stream into two parts and then encoding the resultant data streams into either 4, 8, or 16 amplitude levels, the square root of the total number of states represented. The two data streams containing the amplitude signals are then used to modulate the RF signal in quadrature. Although more spectrally efficient than the first group, they unfortunately require a much higher CNR.

In practice, actual systems are worse in spectral efficiency due to implementation losses which result from the use of realizable filters and data recovery circuits. A practical QPSK modulator, one of the simpler and more commonly used schemes for digital modulation, has a practical spectral efficiency of 1.5 bits/Hz rather than 2 bits/Hz, and requires a CNR of 17 dB rather than 15 dB to achieve a 10E-8 error rate. Thus, a practical QPSK system may have a resultant performance that is 14 dB below the Shannon limit. Hybrid systems, employing some techniques not specifically designed for digital operation, have performance even farther from the Shannon limit, being as much as 20 to 30 dB worse.

The design of a good digital STL system requires the proper selection of a digital modulation scheme combined with a proper low bit-rate audio coding scheme. In the 950 MHz band, the FCC specifies that new STL systems must occupy a bandwidth of less than 300 kHz. The design thus requires the combination of a low bit-rate audio coder combined with an appropriate modulation technique. Since it is possible to trade off audio coder efficiency against digital modulation efficiency, the best combination must be found. Table 1 is a summary of a number of combinations of audio coding (or the lack of it) with various modulation schemes. The three examples given are compared to a conventional FM STL for reference. The reader should use the table as a summary of the examples discussed subsequently.

Type	Modulation	Bandwidth (kHz)	Comments
Conventional STL	FM	300-500	Not robust; variable audio quality
Digital (PCM)	QPSK	930	Spectrally inefficient
Digital (PCM)	64QAM	313	Requires high CNR; complex & costly implementation
Digital (Low Bit rate)	QPSK	170	Coder must sound good

STL Modulation and Coding Options.
Table 1

Consider first the difficulty in the design of a digital link if no low bit-rate audio coding is used. This seems an attractive starting point, since most people consider that the word "digital" has come to mean 16-bit PCM, sampled at 44.1 ksamples/sec: the CD standard. It is a benchmark by which other systems are judged, since everyone has heard the phrase "CD quality." Unfortunately, 16-bit PCM results in a very high bit-rate of about 1.411 M bits/sec for stereo (16 bits/sample x 44.1 k samples/sec x 2 channels). For a digital STL application, this data stream must be modulated onto an RF carrier for transmission.

What happens if stereo, 16-bit PCM is used as the digital signal source for a digital STL? With a practical QPSK digital modulation system, the 1.411 Mbits/sec PCM results in an occupied bandwidth of 930 kHz, well in excess of the required 300 kHz. Consider the use of the more spectrally efficient 64QAM. In this case, the theoretical spectral efficiency is 6 bits/Hz. But if the same assumption of practical implementation loss is used as for the case of the QPSK example, the spectral efficiency drops to 4.5 bits/Hz. This 64QAM system would then allow the PCM audio signal to fit within a bandwidth of 313 kHz, which is very close to the bandwidth target. The disadvantage of the more spectrally efficient technique is that it requires approximately 12 dB better CNR. This creates a problem for the STL designer because it adversely affects the robustness of the digital STL to RF path interference and noise. Digital modulation systems of this complexity are also expensive to build.

The previous example shows that while one could pick a more sophisticated modulation scheme such as 64QAM, the high CNR that is required could result in excessive errors during a fade or in the presence of interference. Error correction could alleviate the need for CNR, but at the expense of increasing the bit rate and system complexity. This occurs because error correction adds a measure of redundancy to the data. In addition, it is difficult to improve the path margin more than 4 or 5 dB because the error rate climbs so fast with decreasing CNR. Since the CNR requirement is 12 dB more severe for the 64QAM system, this makes turning a 64QAM system into one as robust (against channel noise and interference) as a QPSK system an impossible task.

The previous examples clearly show that the development of a digital STL for the 950 MHz band requires the use of low bit-rate audio coders to create a practical STL system. Consider the use of QPSK along with low-bit rate coding. If the bandwidth is constrained to be 300 kHz, then the 1.5 bits/Hz efficiency would require that the audio data rate be less than 450 kbits/s. Therefore if an audio coding system could be developed that has excellent audible quality and a data rate of less than 450 kbits/s, then a practical, excellent sound quality STL with substantial resistance to RF path fading could be built. Since Dolby Laboratories has developed a family of coding systems with excellent sound quality and data rates of as little as 128 kbits/sec per channel, the practical implementation of a superior digital STL is now possible. If one assumes the use of two channels of audio coded with two 128 kbits/sec audio compression systems, the resultant bandwidth is only 170 kHz, which is well within the requirement for occupied bandwidth.

In summary, digital STL systems promise to have substantially improved fade margin and freedom from interference and general overall superior audio performance when compared to conventional FM STLs. The most effective digital STL systems will employ powerful audio coding technology combined with robust digital modulation schemes to produce a superior radio. The importance of low bit-rate coder technology for this type of audio application cannot be over estimated. The selection of a suitable audio coding system will ensure excellent sound quality.

Audio Coding: The Key to STL Applications

Much of the data coded by PCM systems is not needed, often carrying no useful information about the signal. In a quiet passage, for example, less than one percent of the available levels represented by the PCM words may end up being used; only the first few bits are required to code the amplitudes of the low-level audio. Another problem with PCM is that quantizing errors (the errors produced by the representation of the signal by the PCM words) are broadband in frequency, and therefore are quite audible. Therefore, practical low bit-rate audio coders should employ a representation of the music signal that is more efficient than PCM, while avoiding its pitfalls.

Any processing to reduce bit-rate should not audibly compromise the source material; the dynamic range and bandwidth of the original audio must be maintained. Otherwise, we simply trade improved bit-rate for degraded signal quality. It is relatively easy to build a coder that works well for some types of program material. However, STL applications require coders that work very well for all types of program material. This is a significant point, one that can distinguish one variety of coder from another, rendering the inferior one unsuited for STL use.

Effective low bit rate coders depend on the understanding of auditory masking and the hearing limitations of the human auditory system. A simple model of this system is that the ear can be represented by at least a 30 band spectrum analysis process followed by a complex detection process in the brain. This spectrum analysis process divides the audio frequency band into bands of varying bandwidth called "critical bands." These critical bands are non-uniform, and increase in bandwidth with increasing frequency. They are as little as 50 Hz wide at low frequencies, and, at higher frequencies, correspond to approximately 1/5 of an octave.

What is significant about a critical band is that if a signal is present, it strongly inhibits the detection—or masks—the presence of lower amplitude signals within half a critical bandwidth. This occurs because the detection system of the ear in that band is overwhelmed by the presence of the larger signal. However, as one looks in other frequency bands, particularly as you get farther in frequency from the large signal, the ear's ability to detect spurious signals becomes less and less affected by the large signal. This reduction in masking of the lower level signal is particularly rapid for signals lower in frequency than the louder signal.

Thus, effective low bit rate need to coders confine the errors produced by the bit reduction process to each critical band of the signal causing the errors. Since this requires a minimum of 30 bands, systems with a fewer number must rely on other techniques to reduce the errors outside the critical band containing the signal of interest, or have a higher data rate than is absolutely necessary. In addition, this error confining process must have very sharp frequency selectivity for the lower band edges since auditory masking falls off so rapidly. This is a difficult feat, since auditory masking may fall off as much as 90 dB in less than an octave.

Figure 5 represents the situation when a 500 Hz tone at 80 dB sound pressure level is encoded by an optimal bit rate reduction coder.

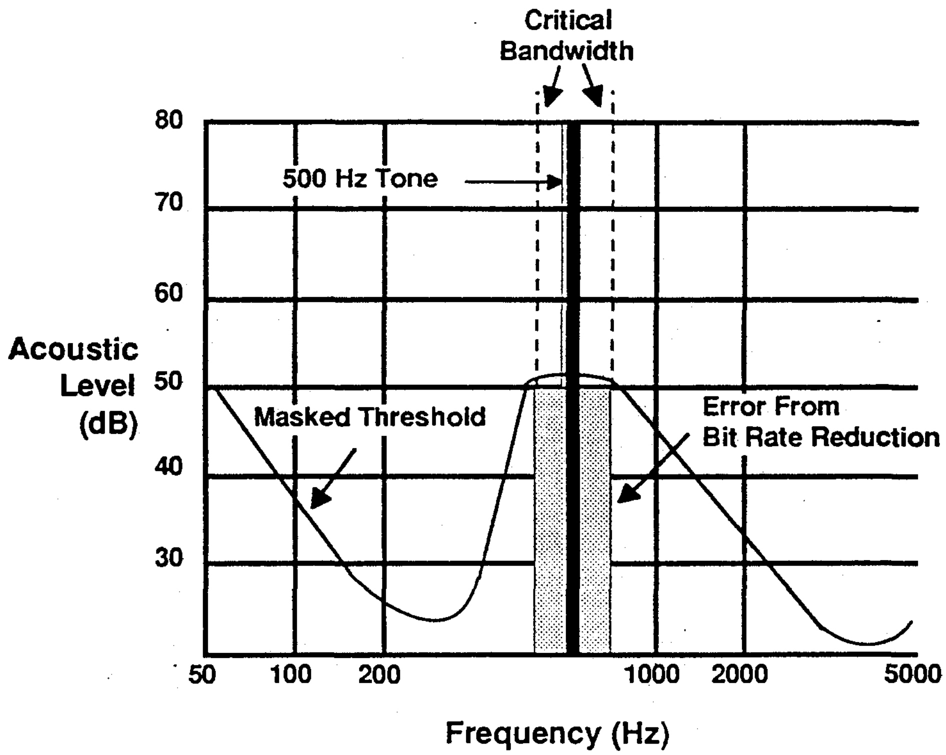

Figure 5: Comparison of Critical Band Masking and Optimal Coder Performance for a 500 Hz Sinewave Signal.

Examining this figure shows that the solid curve represents the threshold of masking, as modified by the presence of a 500 Hz tone. Note that the threshold is raised in the immediate vicinity of the tone, but is unaffected at frequencies far removed from the tone. The vertical dotted lines indicate the critical band associated with 500 Hz and the masking threshold curve is flat and maximally raised in that region. The asymmetry and the tendency for a rapid fall off of masking for lower frequencies is visible in the masking curve. The lower slope drops 25 dB in less than 1/2 octave, while the upper slope experiences only a 5 dB drop in the same interval. This figure shows that if quantizing errors resulting from the audio coding process can be confined to the shaded area as shown, they can be quite large but remain inaudible.

Frequency Division Techniques

The implementation of the most effective audio coding systems require the confinement of errors within one critical band. Sub-band coding techniques are useful in reducing the audibility of the errors produced by the bit rate reduction processes because they confine the frequency range of the errors to within a single sub-band. Coders using sub-band techniques that simulate the non-uniform bandwidth critical bands are very effective in producing excellent audio quality at low bit rates. Coders of this type allocate data capacity on a band-by-band basis, depending on the spectral characteristics of the input signal and its predicted masking characteristics. The disadvantage of straightforward implementations of this type of coder is that the generation of the necessary non-uniform frequency bands is computationally very intensive. The implementation of just the filter banks may consume a full DSP chip.

Two techniques have been used in an attempt to solve this complexity problem. One solution is to use a small number of uniformly spaced sub-bands for some frequency confinement of errors and then to apply adaptive prediction techniques to decrease quantizing noise, compensating for the limited confinement of quantization errors. Another solution is to use a highly frequency-selective frequency division process that is computationally efficient, such as the Fast Fourier Transform (FFT). The creation of the bands of appropriate bandwidth is achieved by grouping together the highly frequency selective transform bands into critical bands.

Coders of the first type described use adaptive predictive pulse code modulation (ADPCM), a well-developed technique used in speech coding. ADPCM reduces errors by modeling the frequency characteristics of the input signal. This is done by using an adaptive filter, fed by a feedback process that changes the filter's characteristics to match the input signal. The advantage of such a system is that it is relatively simple and does not require the transmission of any additional control information, due to the feedback control. However, there are 2 disadvantages of this system: First, signals that are not accurately modeled by the predetermined range of the characteristics of the adaptive filter will not be quantized with low errors. Secondly, transient signals which abruptly change the requirements for the adaptive filter can be compressed or cause the process to produce excessive errors. Because of these two problems, audio coders of this type are not ideally suited to the more critical demands of STL applications, which require excellent transient response and good sound quality for any conceivable musical signal.

Dolby Laboratories has concentrated on the other solution to the computational complexity problem for critical band frequency division. We use the FFT, a frequency division technique that is comparatively low in computational complexity. Dolby's approach to audio coding, designated Dolby AC-2, is our version of the general class of bit rate reduction systems known as adaptive transform coders. The term "transform" is based on the use of frequency transforms, which is discussed next; the "adaptive" nature of the process will be explained thereafter.

Frequency Transforms

Fourier analysis shows that any repetitive signal can be expressed as a series of sine and cosine waves—in other words, its harmonic components. For digital systems which are sampled in time, the Discrete Fourier Transform (DFT) is often used. This function converts a group or block of time samples into the same number of frequency components, or transform coefficients.

In Dolby AC-2, the Time Domain Aliasing Cancellation transform (TDAC) is used, a modification of the Discrete Fourier Transform (DFT). The TDAC transform is a special type of transform that gives the coding designer the advantages of the DFT without any data rate increase when transform blocks are overlapped and added—a technique that ensures the best frequency selectivity. For more information on TDAC and AC-2 in general, see a paper by Davidson, Fielder, and Antill discussing AC-2.

A convenient method of generating the DFT, or TDAC that we are actually using, is the Fast Fourier Transform (FFT). FFT techniques are able to divide the audio spectrum into frequency bands substantially smaller than the narrowest critical band while using a small part of the processing capability of a DSP chip.

Although the FFT is a very computationally efficient method of frequency division, it is a block process rather than a continuous sample-by-sample one. The consequences of this block character is that, unless special steps are taken, a multi-band filter bank implemented by a simple FFT will have poor frequency selectivity due to a discontinuities at the boundaries of adjacent FFT blocks. This situation can be remedied by "windowing" the input of the encoder's block transform and windowing the output of the decoder block's inverse transform.

Windowing is a process which smoothly reduces the gain of the system from unity at the middle of the block, to zero at the beginning and end of each block, thus allowing for smooth transitions between blocks. If no other steps were taken, the audio would be modulated at the block rate, with the audio signal always being at zero amplitude level at the block boundaries. This problem is solved by overlapping blocks in such a way that when you get to a block boundary where the audio has been faded to zero, you also have the contribution of an adjacent block at unity gain. The overlapping blocks that are windowed, twice, once during encode and once during decode, have been arranged by clever window design and overlap conditions to always have an effective gain of one through the audio coder. In this way, the detrimental effect of the block process is removed.

In the Dolby AC-2 coder, a group of 512 samples are taken as a block and transformed using this windowed modified transform. When the transform is taken, the 512-sample fragment of audio is now digitally represented by 256 frequency components—effectively, a 256-channel filter bank. These "transform coefficients" represent spectral parts of audio that are equally spaced at about 94 Hz intervals (48,000/512 = 94) up to the Nyquist frequency of 24 kHz.

These coefficients, which have 24-bit word lengths, (a result of the 24 bit accuracy of the DSP that we've used), are then grouped to form critical bands. Within each critical band, the coefficients are then quantized. In so doing, they are converted to a block floating point notation. In this notation, a control, or exponent word is determined, that represents the maximum of all the transform coefficients to within a power of 2. This is used to scale all the transform coefficients upward so that a simple truncation of the transform coefficients to reduce their word lengths creates the audio data, or mantissa (which is the other part of block floating point notation).

The exponents provide an estimate of the spectral distribution of the audio sample: larger exponents mean a greater spectral energy in the band, and vice versa. These exponents are transmitted as side information to the decoder. The mantissas carry the compressed amplitude information. The bit resolution to which they are quantized is determined partly by an adaptive process and partly by a fixed allocation process where the word lengths are predetermined as a function of frequency.

Adaptive Coding

In the Dolby AC-2 coding system, most of the word length allocation is fixed rather than adaptive to assist in the creation of a non-signal dependent coder that yields excellent sound quality regardless of the nature of the audio signal. The adaptive part, however, is still quite important and provides the first adjective in the term "adaptive transform coding". When greater accuracy in a particular band is required to preserve sound quality, the scaled transform coefficients or mantissas are allowed to have a longer word length than they would normally have, by allocating more bits to them. On the other hand, fewer bits can be allocated to frequency bands whose components are masked by signals in other bands, when for example, the signal level in an adjacent band is especially loud. Thus, each sub-band mantissa is quantized to a bit resolution that is the sum of a coarse, fixed-bit component, and a fine, dynamically-allocated component.

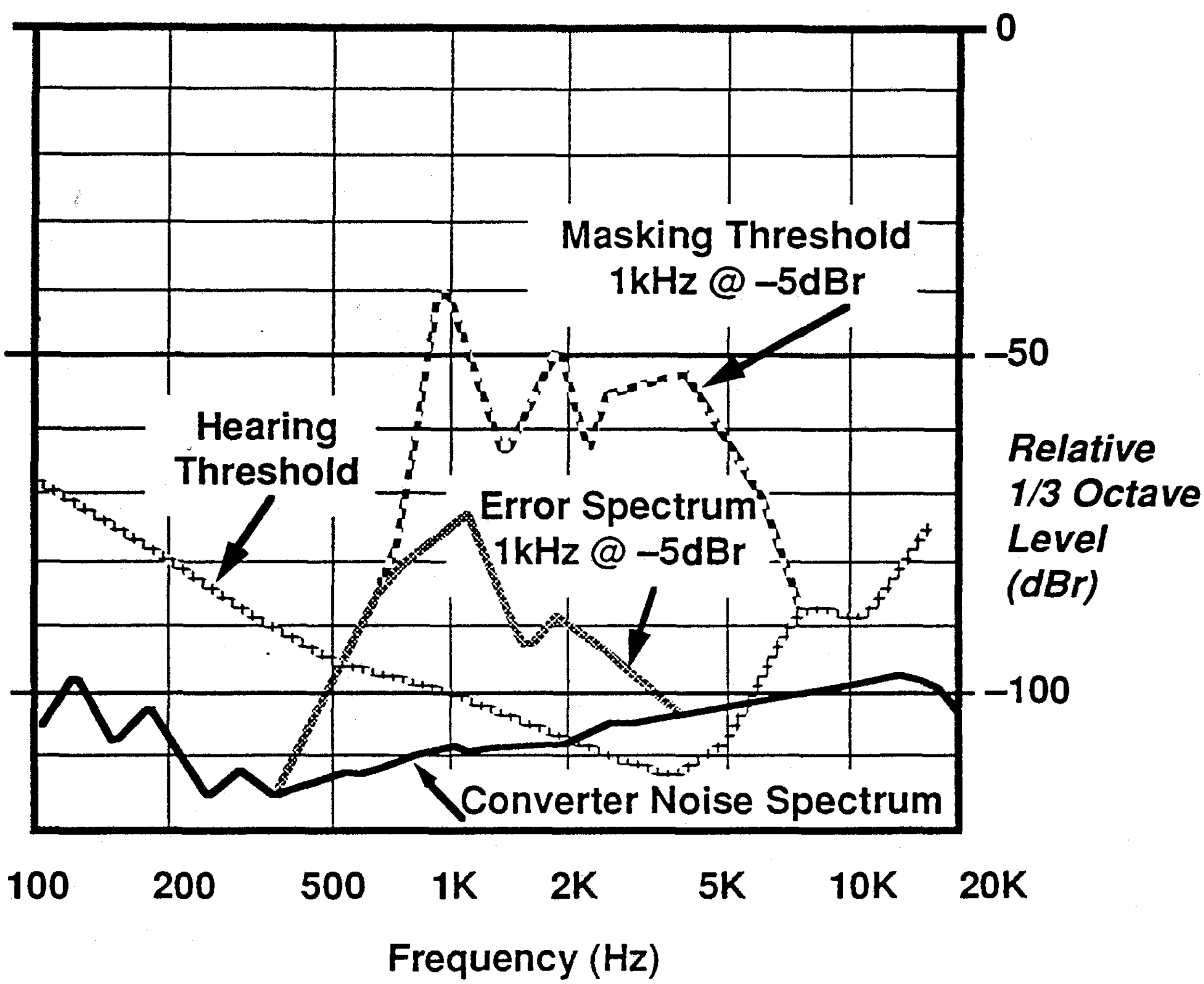

Figure 6: ADC + DDAC + AC-2 Coder Noise Spectrum for 1kHz Tone Compared to Masking Threshold.

Figure 6 shows the actual performance of one implementation of Dolby AC-2 (in our Model 500 Series stand-alone audio coding units). In this example, a 1 kHz tone is the input signal. Superimposed on the hearing threshold curve is a threshold curve, modified due to the presence of the tone. (As an aside, note that the threshold is raised for the second and third harmonics of the fundamental.) Between these two curves is the error spectrum of the AC-2 coder, which lies comfortably below the modified masking threshold.

Having accurately quantified the signal in terms of level and psycho-acoustically selected frequency bands, we can now multiplex the control words and audio mantissa words together and assemble them into a serial data stream. In addition, a synchronization word and parity words to perform error correction on the control words are inserted. By protecting the control words, which are the coarse level controls, the audible distortions of AC-2, due, for example, to high channel errors, are quite benign. The mantissa bits are left unprotected. Mantissa errors are proportional to the signal spectrum and, in general, are completely masked by the signal.

Summary

It has been shown that the successful implementation of a spectrum-efficient digital STL system requires the careful selection of a digital modulation scheme and the use of techniques especially designed for digital applications. Digital modulation techniques were shown to result in STLs with exceptional constancy of quality, when compared with FM systems, as the RF path degrades due to fading or interference. This greater robustness was accompanied by an unfortunate tendency for a digital system to require greater bandwidth than an FM STL. Fortunately, the recent development of powerful data rate reduction systems for music applications allow the digital date rate to be reduced to the point where the effective bandwidth of a digital STL is less than the equivalent FM implementation would require. Dolby's AC-2 audio coding family was shown to have the necessary audio quality while permitting effective bandwidth reduction to create a successful digital STL system. This system possesses better sound quality than the best FM STL, delivers this performance over a much wider range of operating conditions, and occupies a narrower RF bandwidth than FM systems. In addition, the use of Dolby's AC-2 family, the product of 7 years of audio coder research and 25 years of experience in psychoacoustics is the best choice for the highest quality audio applications.

The Dolby DP5500 DSTL™

At NAB '91 in Las Vegas, Dolby Laboratories announced progress in development of an all-digital STL system. It is based on Dolby AC-2 audio coding, combined with appropriate modem and RF technology, and provides a level of performance exceeding that of conventional FM (composite or discrete) systems, as well as digital schemes employing lesser coding systems. Digital modulation techniques combined with a high performance RF section provides a robust signal, more immune to fade and interference than FM systems, yet able to co-exist with them. For more information, please contact Dolby Laboratories.

References

K. Feher, Advanced Digital Communications, Prentice Hall Inc., New Jersey, 1987, p. 324.

G. Davidson, L. Fielder, and M. Antill, "Low-Complexity Transform Coder for Satellite Link Applications," Pre-print No. 2966, Audio Eng. Soc., presented at the 89th Convention, Sept. 1990.

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A Digital Stereo Generator for FM Broadcast

Michael Antill and Eric Benjamin

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A DIGITAL STEREO GENERATOR FOR FM BROADCAST

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ABSTRACT

The design of a composite FM stereo generator based on digital signal processing (DSP) techniques is described. A single programmable DSP IC performs the functions of pilot and subcarrier generation, matrixing, modulation, and test tone generation. A high performance, low complexity implementation for use in a digital studio-to-transmitter link (STL) is presented and compared to traditional approaches.

1. INTRODUCTION

With the economical integration of digital signal processing (DSP) technology, many signal processing tasks previously accomplished in the analog domain can now be implemented with reduced complexity and higher performance. One area that has benefited significantly from this new technology is that of broadcast audio, where the selective application of DSP can make simultaneous improvements in audio quality, stability of operation, and RF spectral efficiency. The Dolby Digital Studio-to-Transmitter Link (DSTL[1]) exemplifies the use of these techniques in the FM broadcast environment [1], and has provided the incentive to design a DSP-based composite stereo generator with its attendant advantages.

As shown in figure 1, the DSTL receiver is a modular design consisting of individual receiver/synthesizer, demodulator, audio decoder, D/A converter, alarm, power supply, and optional stereo generator modules. The latter utilizes a single programmable digital signal processor to generate a composite stereo signal from the PCM audio output of the audio decoder module. By keeping the audio signal in the digital domain until final conversion to composite analog, several important benefits can be simultaneously realized. Foremost among these are superior audio performance, stability of operating parameters, and the higher reliability and reduced costs associated with a low complexity design.

[1]DSTL is a registered trademark of Dolby Laboratories Inc.

2. FUNCTIONAL DESCRIPTION AND ALTERNATE APPROACHES

The basic function of any FM stereo generator is to provide a composite output signal consisting of the baseband (sum of left and right input channels), a 19 kHz pilot signal, and a 38 kHz suppressed-carrier subchannel, amplitude modulated with the difference of the input channels (L-R) [2]. Figure 2 depicts the spectrum of the composite stereo signal based on a uniform spectral distribution in the 15 kHz baseband.

Traditionally, two basic approaches have been taken in the implementation of this function [3,4,5]. In matrix-type generators, baseband and difference signals are generated by precision analog networks. The difference (L-R) signal and 38 kHz subcarrier are then fed to a balanced modulator (typically an analog multiplier) to create the double-sideband suppressed-carrier subchannel centered at 38 kHz. Baseband, subchannel, and pilot signals are then summed to produce the composite output. Performance of the generator is limited by a number of sources, including nonlinearities and noise within the modulator, and the difficulties of maintaining precise component alignment due to temperature effects and aging. Parameters subject to variation and/or degradation include audio distortion and dynamic range, channel separation, and suppression of spurious modulation products.

In switching-type generators, an electronic switch is used to alternately select the left and right input channels at a 38 kHz rate, followed by low-pass filtering to remove harmonics of the subcarrier. Although these designs do not require the precise generation of sum and difference signals, they are extremely sensitive to variations in switching duty cycle which can be temperature and/or component dependent. In addition, the finite switching time and channel impedances of real switches (and their variations) can further degrade performance in the same manner as matrix generators. Other circuit topologies have also been implemented which combine aspects of both matrix and switching generators to improve upon the performance of either technique; to some extent, however, they remain subject to the same deficiencies.

Most of the limitations encountered with these types of stereo generators can be traced to the fact that they are fundamentally analog signal processors, and as such are subject to the effects of component accuracy and aging, temperature and power supply variations, and calibration/alignment errors. In order to minimize these effects, DSP techniques have been employed in the design of a new stereo generator. As shown in figure 3, a single 24-bit Motorola DSP56001 makes up the signal processing core of the module, followed by a high performance digital-to-analog converter, output filter, and buffer amplifier. System alignment is facilitated by a user interface that directly controls the operation of the processor, and a simple reset controller monitors processor operating status. The composite signal is generated entirely in the digital domain by the DSP56001 before final conversion to analog output.

3. DESCRIPTION OF OPERATION

The digital stereo generator receives 16-bit PCM data for left and right channels at a sample rate of 44.1 kHz from the audio decoder module of the DSTL receiver. This bit-serial data stream is then routed to the serial input port of the DSP56001 for processing; figure 4 provides a flow diagram of DSP operations in the digital stereo generator. Under normal operation, the processor calculates sum and difference signals for the left and right channel inputs, with provision to scale the L-R signal from 95 to 105% of the baseband (L+R) level under user control. This allows the relative level of the modulated difference signal to be adjusted with a resolution greater than .005 dB in order to accurately compensate for any deviation in the FM exciter response.

Since modulation of the 38 kHz subcarrier results in a 53 kHz composite signal bandwidth, it is first necessary to upsample the matrixed signals by a factor of four. The oversampling process is the most processor-intensive part of the generator algorithm, and results in the removal of signal images below 88.2 kHz via a two-stage linear phase FIR interpolation filter. The computational burden is somewhat alleviated by exploiting the fact that the input signals have been sharply band-limited to 15 kHz by the audio coding process. As a result, sharp transition bands are not required in the oversampling filters, and the resulting specifications can be met by cascaded half-band filters [6]. These structures are characterized by alternating zero-valued coefficients, which reduce the total multiply-accumulate operations by nearly one-half. Filter response is characterized by passband ripple less than .001 dB and image attenuation greater than 90 dB.

A second-order recursive oscillator provides the 19 kHz pilot reference, which is then squared, offset by a half, and scaled by two to provide the 38 kHz subcarrier. In order to establish the correct phase relationship between pilot and subcarrier, pilot phase is shifted by linearly combining the present and previous output samples of the oscillator; pilot level is user scalable from 7 to 12.5%. The difference signal is multiplied by the subcarrier, added to the baseband and scaled pilot signals, and the combination is passed to the D/A converter via the output serial port of the processor. All of the tasks of pilot and subcarrier generation, modulation, composite summation and output take place at the upsampled rate of 176.4 kHz.

For system set-up, alignment, and test purposes, an audio test oscillator is implemented in the same manner as the pilot oscillator. Under user control, reference level tones at 400 Hz and 1 kHz can be substituted for both input channels, the right channel input can be disabled or inverted, and the pilot signal can be independently turned off. These controls facilitate adjustment of operating levels, optimization of channel separation, and the measurement of various performance parameters. For flexibility of operation, the generator also allows remote or local selection of stereo (composite) or mono (baseband only) output.

Performance of the DSP algorithm is nearly ideal, and the composite PCM signal is limited almost entirely by the accuracy of the input signals. Figure 5 displays the spectrum of the 16-bit composite output, using the internal 1 kHz test oscillator at a -2 dBFS level into the left channel only. In this case, input resolution is not limited to 16 bits, and the error spectra are due primarily to residual images of the oversampled input signal. The matrixing and modulation operations are linear to within the 24-bit arithmetic precision of the processor, and oscillator stability is crystal-based. Consequently, it is imperative that the D/A converter and output filter approach this level of precision as closely as possible in order to minimize any loss of performance in the conversion to composite analog.

Conversion to a quantized analog signal is performed by a 16-bit R-2R D/A converter designed for oversampling use. This part is characterized for high frequency operation and exhibits fast settling at the composite sample rate of 176.4 kHz. Although performance is excellent in oversampled applications, distortion increases rapidly above 20 kHz due to intermodulation (IM) in the internal current to voltage converter. It is therefore necessary to bypass this circuit with an external high-speed amplifier having a faster slew rate with reduced IM. By integrating the signal slightly, this stage also reduces the slew rate requirements for subsequent circuitry.

Following digital-to-analog conversion, the composite signal is filtered in order to remove images occurring above 123.4 kHz (176.4 kHz - 53 kHz). This filter must meet precise magnitude and phase specifications in order to maintain high separation between the multiplexed channels [3]. In particular, the overall frequency response must be flat to within +/-.017 dB of magnitude and +/-0.11 degree of linear phase in order to achieve a channel separation of 60 dB throughout the audio band.

The signal is initially filtered by a ninth order low pass elliptic filter with .01 dB passband ripple and an 85 kHz band edge. Attenuation of images is greater than 84 dB for frequencies above 120 kHz. This filter is followed by a second order high frequency boost to precisely compensate for the sin X/X loss due to the finite sample length of the D/A converter (1.43 dB at 53 kHz). The phase linearity of these networks is excellent within the composite signal passband (0-53 kHz), and the overall phase response is held to +/-0.1 degree with the aid of a second order phase compensator. This all-pass network supplies an additional 360 degrees of total phase shift, with a slope that compensates for the phase error accumulated in the elliptic filter and sin X/X compensator.

After filtering, the composite output is buffered by a two-stage amplifier with a frequency response of +/-.002 dB from 0 to 53 kHz and twin tone IM distortion below -100 dB. The design is capable of driving loads as low as 50 ohms, and DC offset at the output is held to within 2 mV. The buffered composite signal and its return are coupled to the output connector via a two pole form C relay that engages when the DSTL receiver is in the operate mode.

4. PERFORMANCE

The performance of the DSP algorithm is illustrated in figures 5, 7, and 9, which display the spectrum of the composite PCM output for various combinations of left and right channel inputs using the 1 kHz digital test oscillator. Maximum input level in the digital domain is -2 dBFS to avoid the possibility of clipping. All of the spectra are limited to 88.2 kHz, the Nyquist rate of the oversampled output.

Figure 5 represents the case of a single channel input signal (left only), which produces equal energy in both the baseband and modulated subcarrier signals. The primary artifacts are images of the baseband component at 87.2 kHz and of the modulated subcarrier at 49.2 kHz and 51.2 kHz. Image attenuation is a function of the oversampling filter specifications and exceeds 100 dB.

Figure 7 displays the condition of identical left and right channel inputs (mono operation), producing a full level baseband signal. The only significant artifact is the baseband image at 87.2 kHz at a level of -105 dBFS.

Figure 9 shows the situation where left and right channel inputs are of equal level and out of phase to produce a full level modulated subcarrier. Sideband images are present at 49.2 kHz and 51.2 kHz and are attenuated by more than 100 dB.

By comparison, figures 6, 8, and 10 represent generator performance at the composite analog output for the same input conditions as figures 5, 7, and 9, respectively. These spectra have been rescaled such that full scale represents 100% modulation. Performance is limited by the D/A converter and output filter circuits, where degradation is primarily caused by intermodulation and harmonic distortion. The composite stereo signal is particularly sensitive to nonlinear distortion due to its multiplexed nature, and distortion components can result in aliased signals and nonlinear crosstalk in the demodulated output.

Care must be taken to differentiate the performance of the generator from that of the measurement system - in this case, an FFT spectrum analyzer with a 16-bit A/D converter. For this reason, the spectra presented here are concatenations of separate baseband and subchannel measurements in which high quality analog filters were used to suppress high level signal components before analysis. The spectral noise floor displayed in the figures is also limited by the analyzer, except in the region of 80 kHz where additional noise is added by the generator output filter.

Figure 6 shows the analog spectrum for the case of a single channel input. Distortion artifacts include third order sidebands of both pilot and subcarrier modulation, and second and fourth harmonics of the pilot. Signal levels are below -90 dB for each of these components.

Figure 8 represents the mono (L+R) operating condition. The second order harmonic of the test tone represents the only significant distortion product at a level of -92 dBFS.

Figure 10 displays difference signal (L-R) modulation. Distortion products include third order sidebands of the subcarrier modulation, and the fourth harmonic of the pilot at 76 kHz. Each of these artifacts is more than 90 dB below full level.

The performance evaluation of any stereo generator is complicated by the fact that many specifications can only be measured accurately after decoding, and will therefore incorporate the performance of the decoder. Noise and distortion can be evaluated directly from the output spectrum, although the quality of the analyzer will be a limiting factor. By comparison, separation is a sensitive parameter that depends not only on the magnitude and phase response of the generator, but on the corresponding response of the FM exciter and its connecting cable. In actual practice, separation is often measured and optimized by means of the stereo monitor used in a particular installation. Loss of separation can also occur during transmission (via multipath distortion), and during reception and demodulation. Since degradation can occur at each step in the chain, high performance in the generator is particularly important.

5. CONCLUSIONS

DSP technology has made possible the design of a high performance, low complexity FM stereo generator in which the composite signal is synthesized entirely in the digital domain. The digital implementation of critical signal processing tasks minimizes the limitations of analog circuitry. Performance of the DSP algorithm is nearly ideal, and signal quality is dependent only on the digital to analog conversion process. Typical distortion products using a 16-bit D/A converter are more than 90 dB below maximum modulation.

6. ACKNOWLEDGEMENT

The authors wish to acknowledge the contributions of Mark Atherton and Grant Davidson in the definition of the digital stereo generator, and in the development of algorithms, software, and hardware for this product.

7. REFERENCES

1. R. Bell, "RF Design Considerations in the Development of a High-Spectral Efficient, Multi-Channel, All-Digital STL," Proc., 46th Annual NAB Broadcast Eng. Conf., Las Vegas, 1992.

2. A. Csicsatka and R. Lintz, "The New Stereo FM Broadcasting System - How to Understand the FCC Specifications and Generate the Composite Signal," J. Audio Eng. Soc., vol. 10, pp. 2-7, Jan. 1962.

3. T. Brook, "Stereo Coder, Part 1," Wireless World, vol. 83, pp. 79-82, Apr. 1977.

4. D. Hershberger and G. Mendenhall, "New Techniques for Generation of Composite Stereo Signals," Proc., 31st Annual NAB Broadcast Eng. Conf., Washington, 1977.

5. A.J. Oliveira, "A Design Method for High-Audio-Quality FM Multiplex Encoders," J. Audio Eng. Soc., vol. 40, pp. 130-141, Mar. 1992.

6. F. Mintzer, "On Half-Band, Third-Band, and Nth-Band FIR Filters and Their Design," IEEE Trans. on Acous., Speech, and Sig. Proc., vol. ASSP-30, pp. 734-738, Oct. 1982.

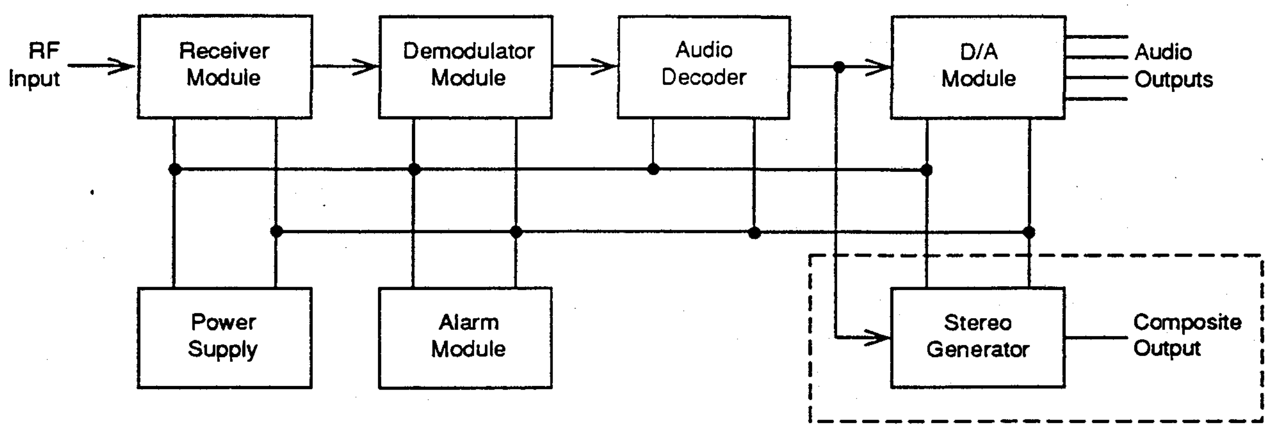

Figure 1. DSTL Receiver Block Diagram

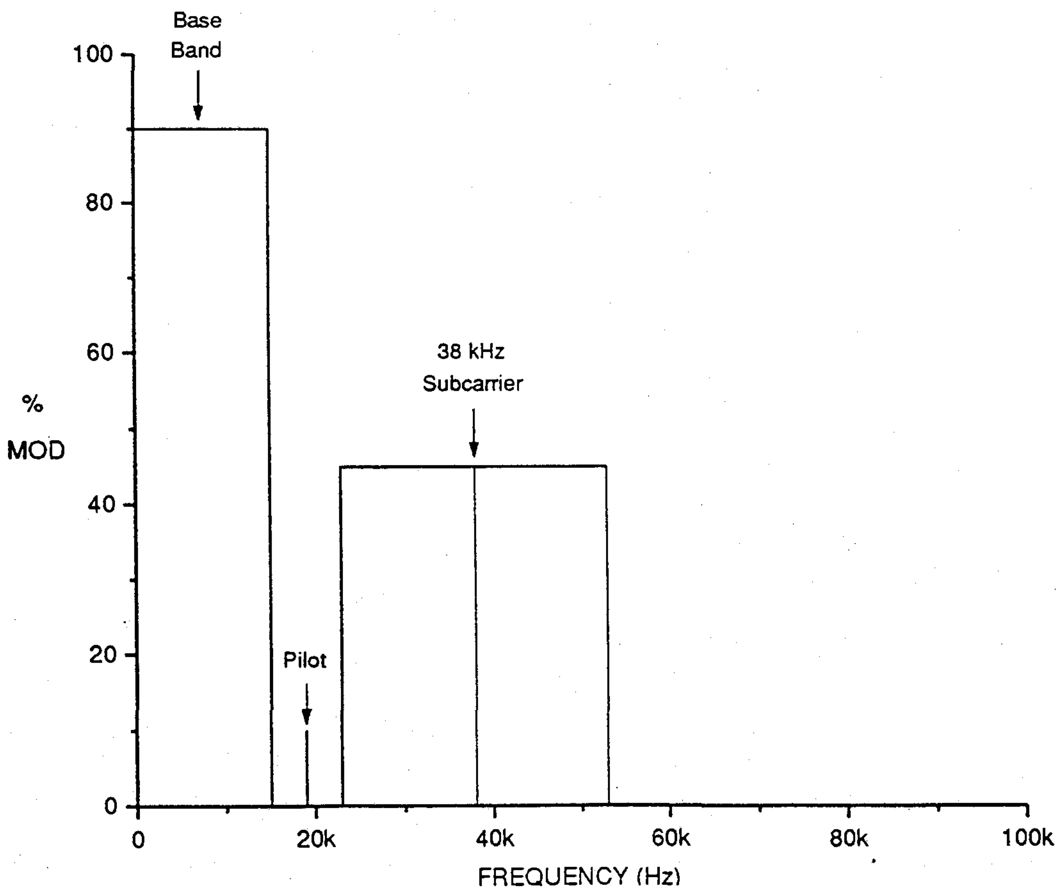

Figure 2. Composite Stereo Signal Spectrum

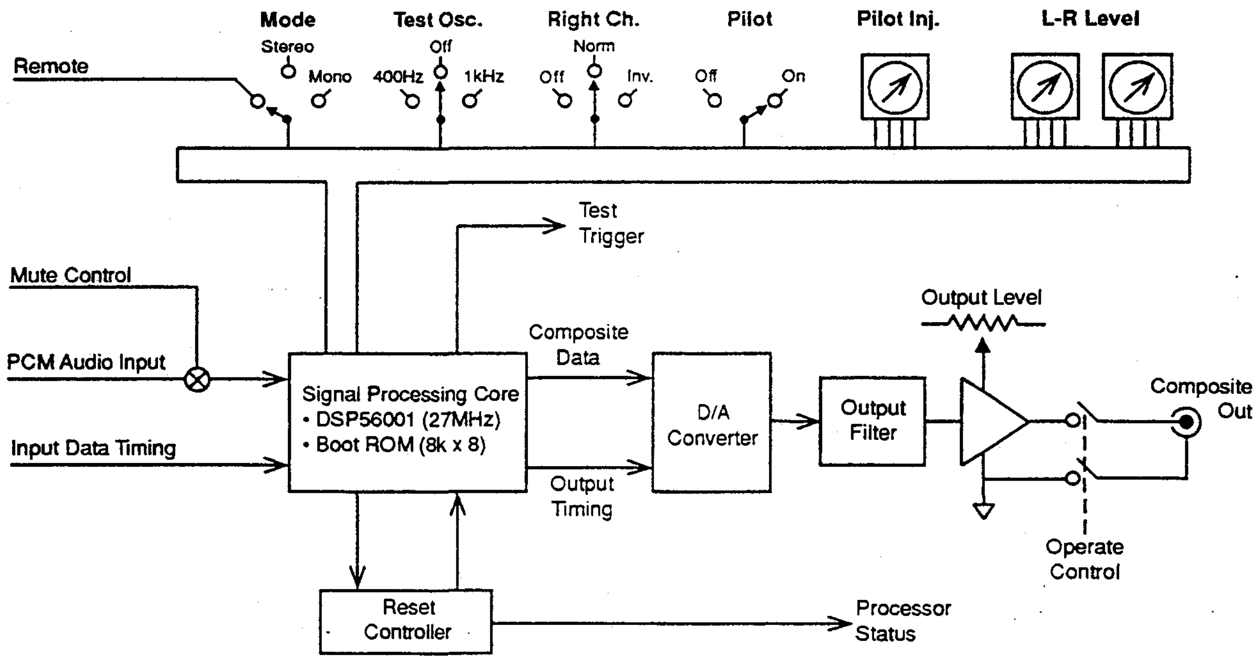

Figure 3. Stereo Generator Functional Block Diagram

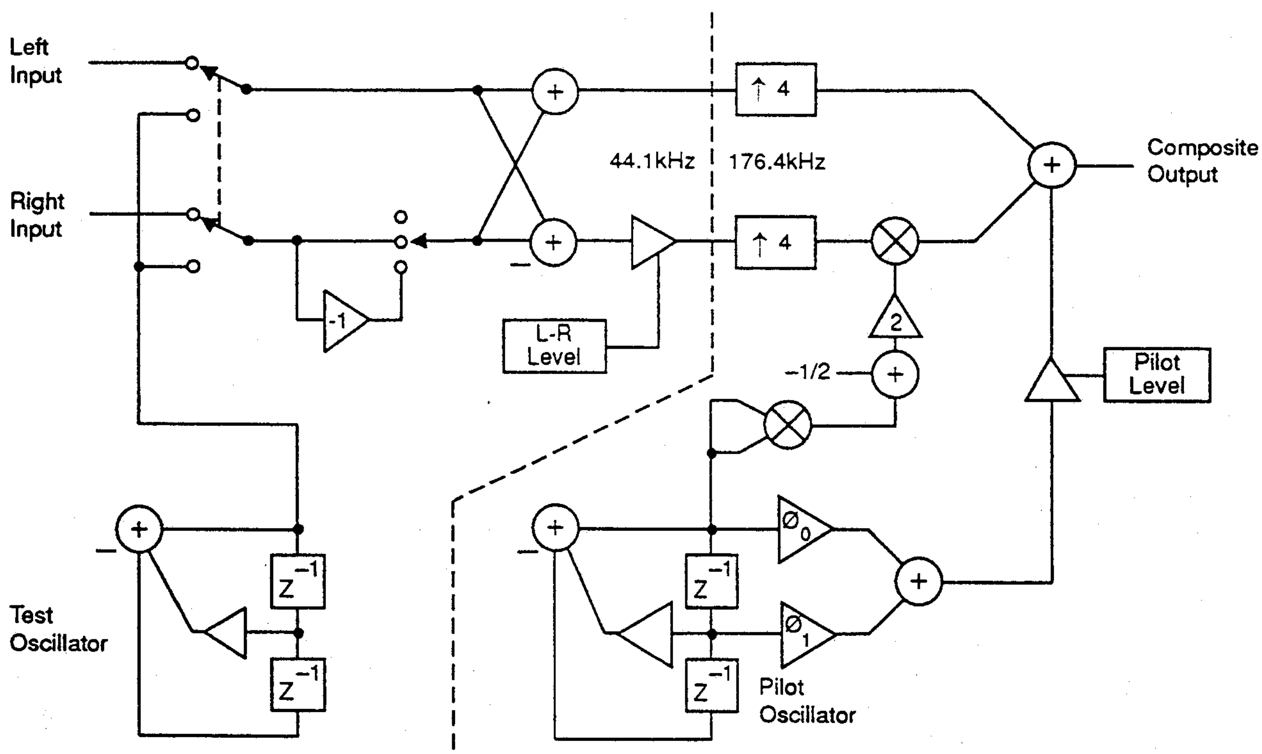

Figure 4. DSP Algorithm for Digital Stereo Generator

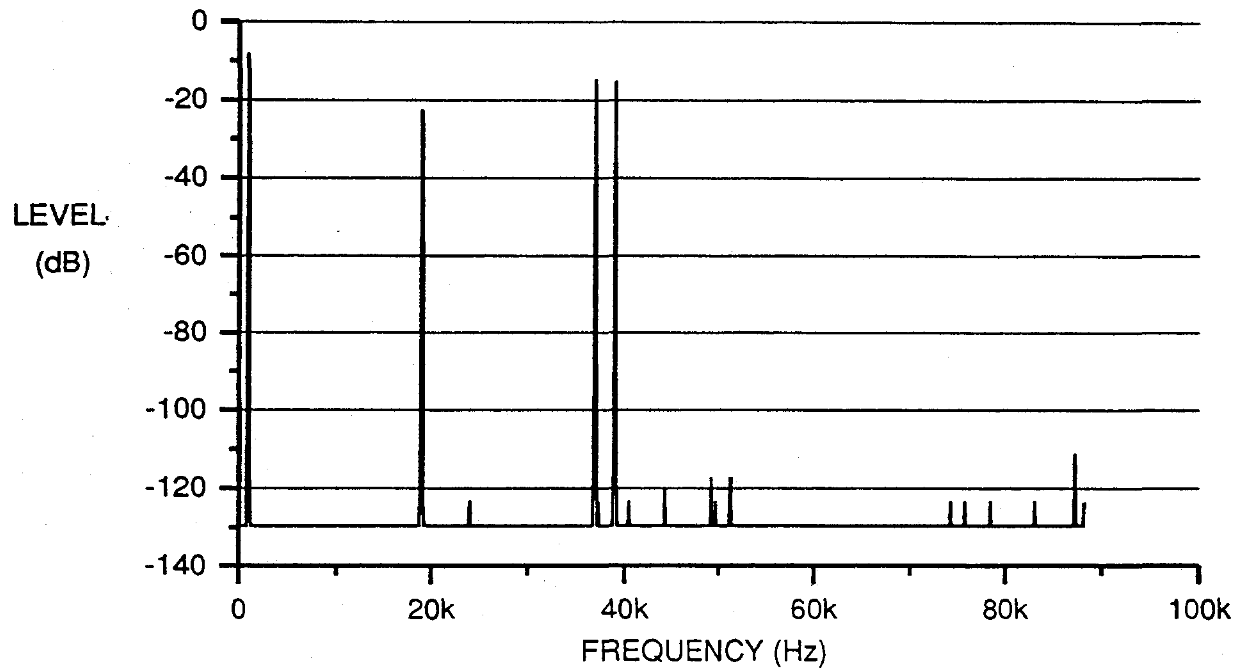

Figure 5. Spectrum of Composite Digital Output for Left Channel Modulation

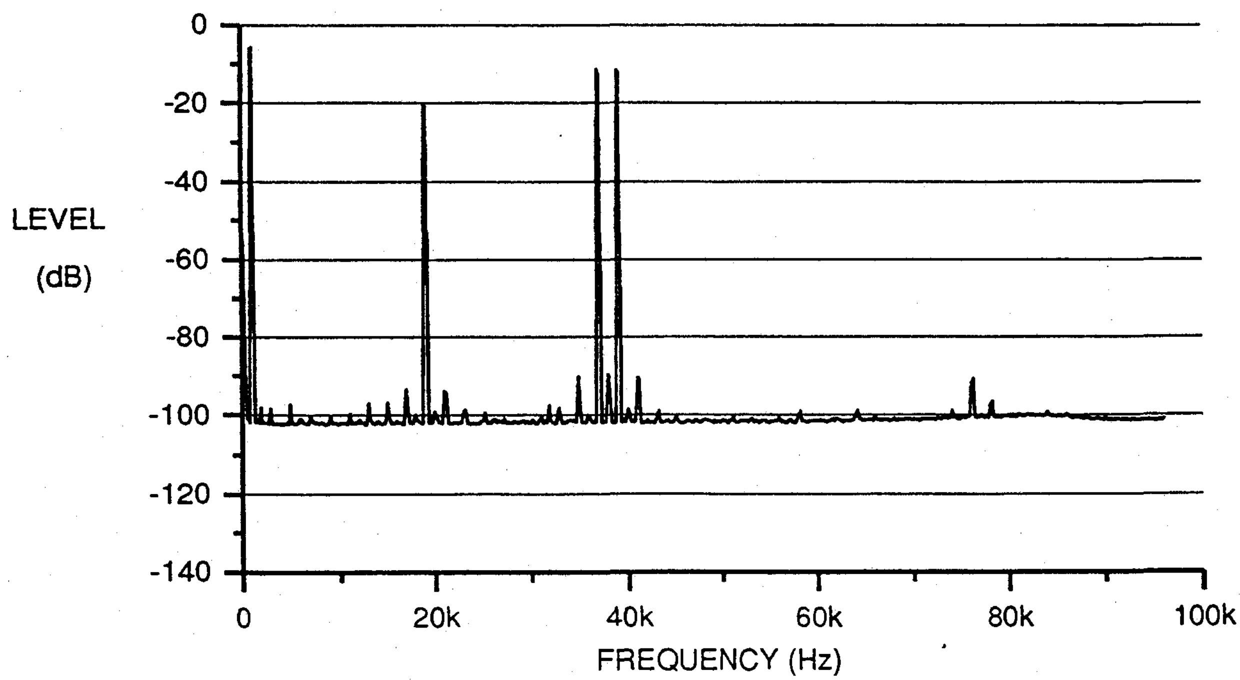

Figure 6. Spectrum of Composite Analog Output for Left Channel Modulation

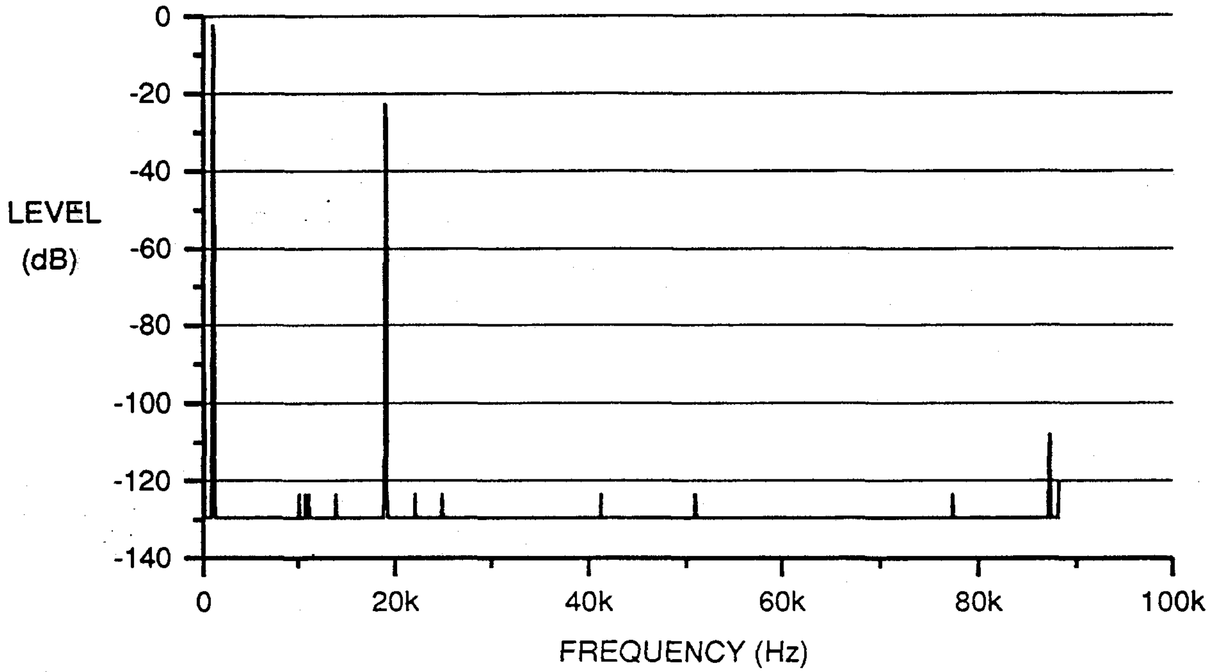

Figure 7. Spectrum of Composite Digital Output for
Left + Right Channel Modulation

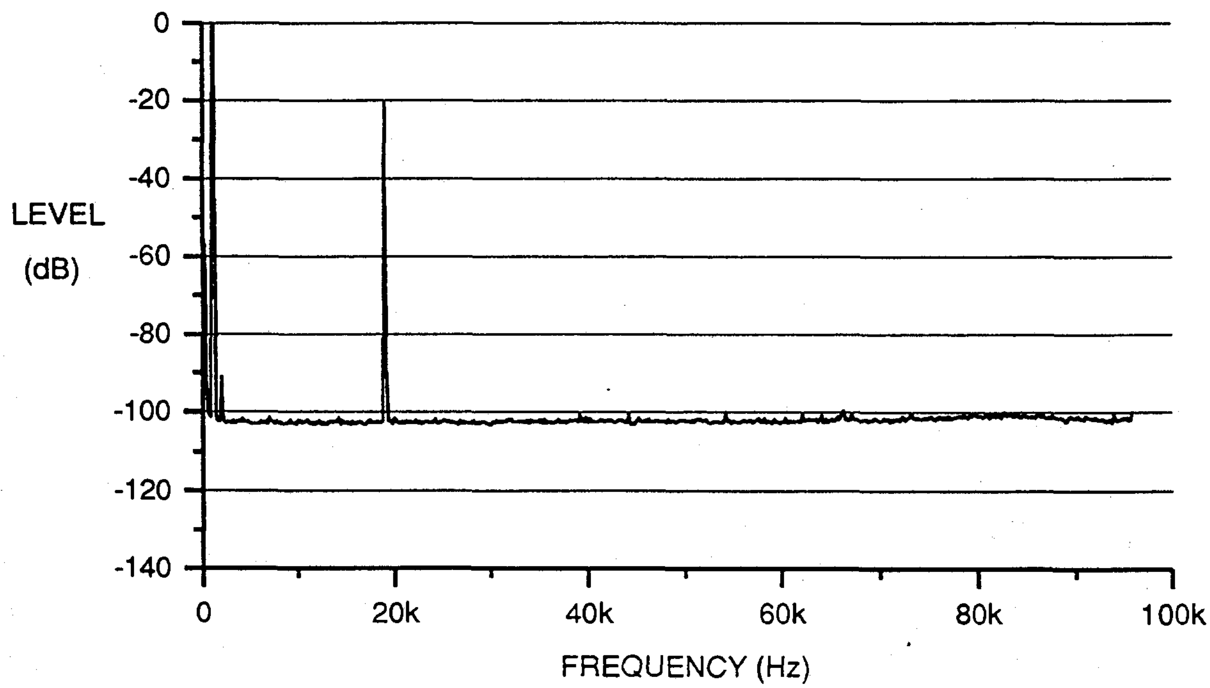

Figure 8. Spectrum of Composite Analog Output for
Left + Right Channel Modulation

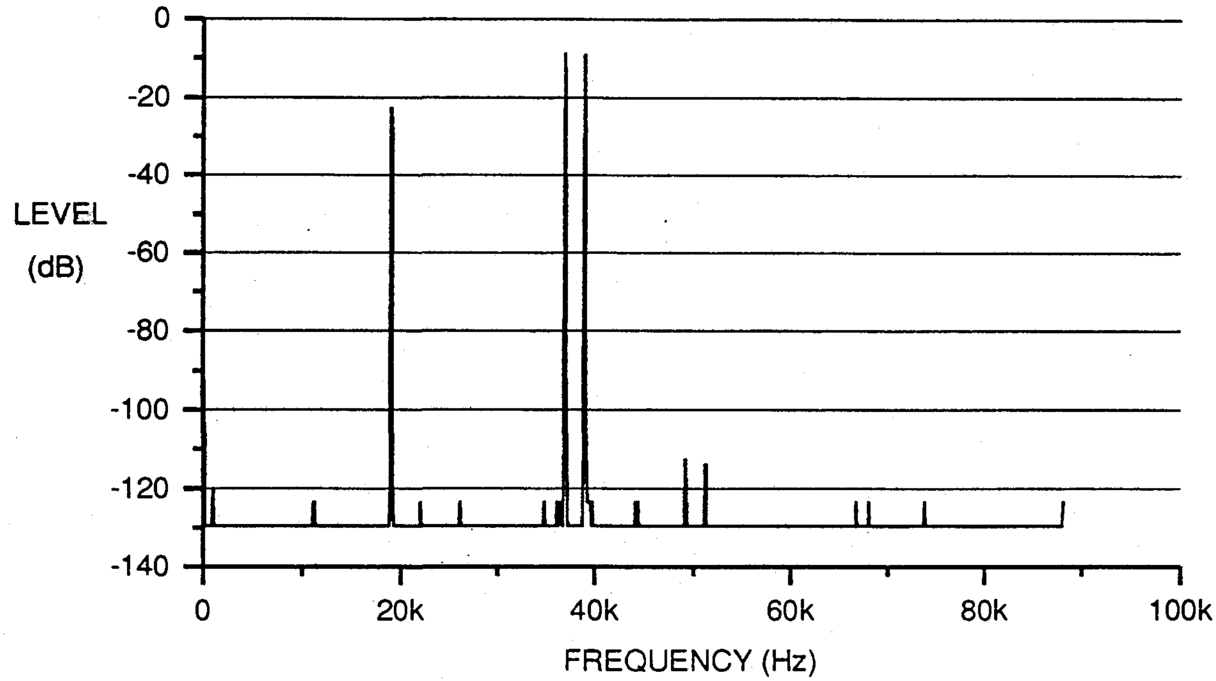

Figure 9. Spectrum of Composite Digital Output for Left – Right Channel Modulation

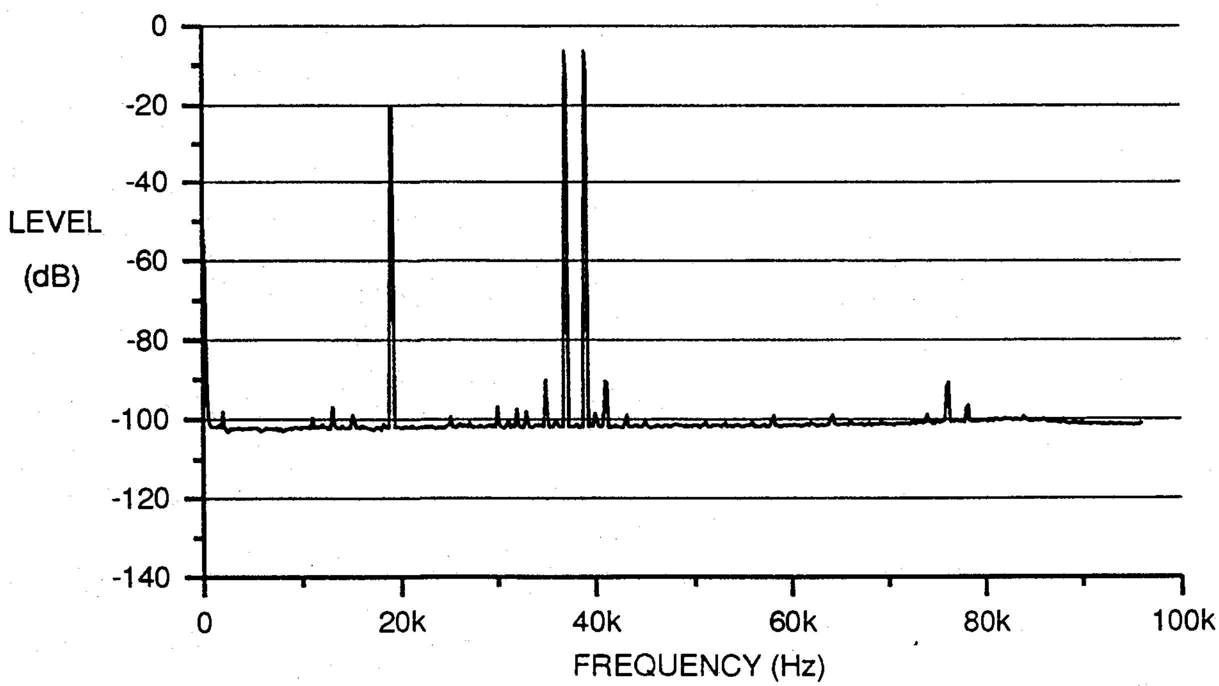

Figure 10. Spectrum of Composite Analog Output for Left – Right Channel Modulation

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LIMITED TWO YEAR

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