



DSP 6000 Digital Transmission System



The Ultimate Digital Transmission System

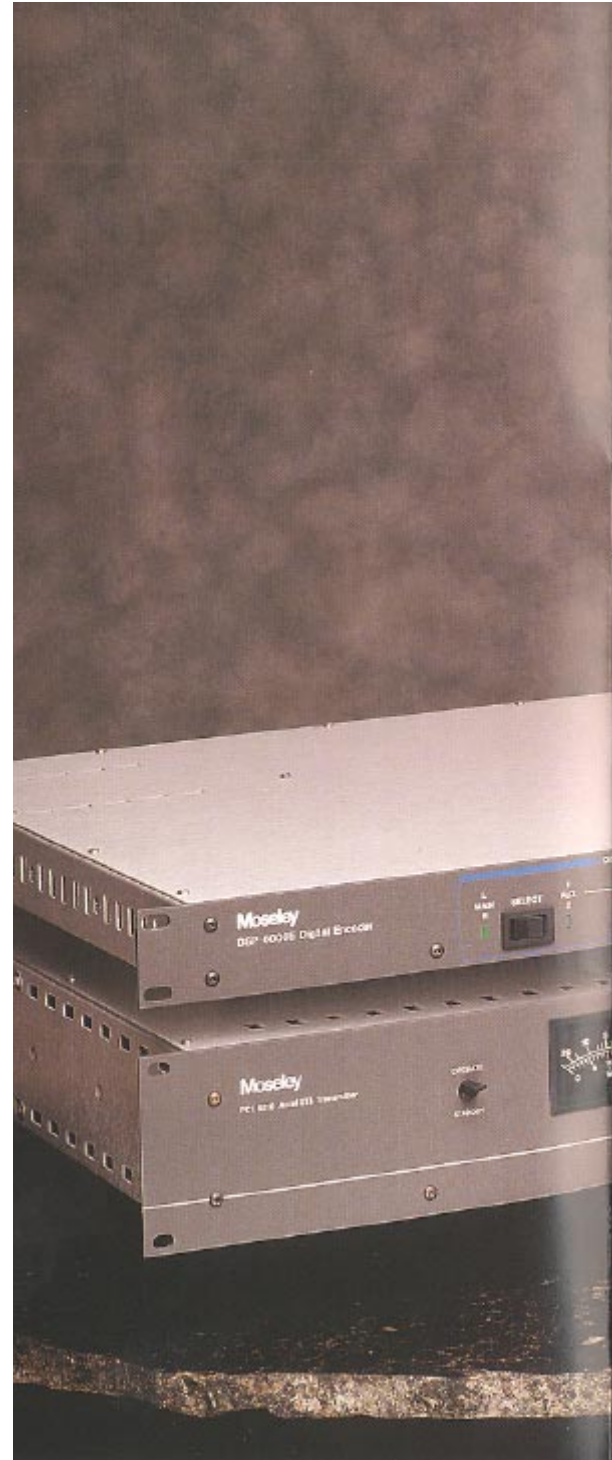
▲
RARELY DOES TECHNOLOGY
HAVE THE POWER TO
DELIVER SUCH DIVIDENDS
▼

Digital audio tapes, compact discs, digital cartridges and other digital storage media have made significant inroads toward the culmination of the “all digital” radio station. Traditionally, excessive bandwidth requirements have made the transmission

of digitized high-quality programming impossible with narrowband studio-to-transmitter links.

After years of work in the art of applying advanced technologies to the cause of a clean and transparent sound, Moseley introduced the world’s first spectrum-efficient, CD-quality digital STL—the DSP 6000. Designed to achieve the widest possible array of desirable characteristics, this open-architecture, modular system features interna-

- CD Quality
- Spectrum Efficient
- Higher Gain
- Constant SNR
- No Crosstalk
- No Background Chatter
- No Phase Distortion
- Multiple Hops
- Direct Digital I/O



tionally accepted digital connectivity, multichannel capability and harmonious coexistence with analog radios.

Following the phenomenal success of the DSP 6000 with over 2000 units installed in the first three years, Moseley is proud to unveil the new DSP 6000—giving new meaning to technical innovation and the relentless pursuit of product perfection. ◀



Above: The DSP 6000 system consists of the DSP 6000E Source and Channel Encoder, the DSP 6000D Source and Channel Decoder and any Moseley digital-ready STL transmitter and receiver. The DSP 6000E and DSP 6000D can also be used with other FM radios and/or digital circuits.

Right: State-of-the-art audiophile quality comes to audio transmission.



The Digital Advantage

The DSP 6000 offers broadcasters the opportunity to convey CD-quality audio between studio and transmission facilities. Stations going “all digital” will present a sonic advantage

equivalent to the LP vs. CD. In addition, the DSP 6000 uses technology to resolve STL problems that negatively affect sound and at the same time has a positive effect on the bottom line.

Instant Payback

It is no longer necessary to upgrade to expensive antennas for longer signal path lengths.

Compared to analog systems, the DSP 6000 requires 25 dB less signal to deliver 90 dB dynamic range. This reduces antenna and transmission line costs.

Constant SNR

Fades have no effect on signal-to-noise ratio (SNR). The DSP 6000

delivers its full dynamic range all the way down to the digital threshold. So, even quiet passages remain crystal clear and virtually noise-free.

Spectral Efficiency

The DSP 6000 can operate in existing channel allocations from 100 kHz to 500 kHz, coexisting with already established radios.

Selectable Spectral Efficiency

The DSP 6000 offers a choice of spectral efficiency of 1, 2 or 3 bps/Hz. Higher efficiencies allow a tradeoff between system gain and occupied spectrum.

Degradation-Free Multiple Hops

Multiple signal hops have no effect on a digital signal. With the DSP 6000, SNR, frequency response, distortion, stereo separation and crosstalk remain consistent throughout the entire signal path—no matter the number of hops!

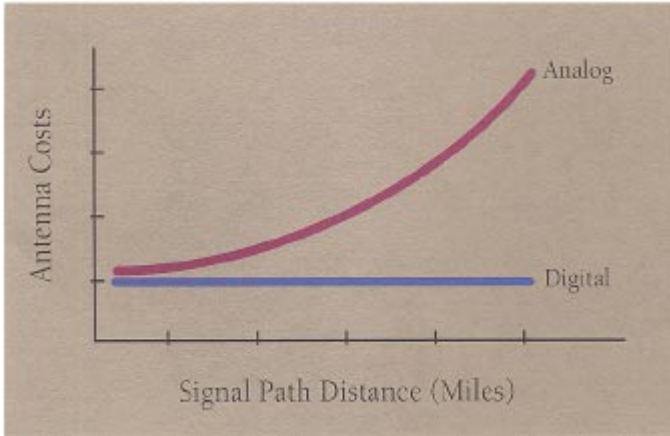
Choice of Source Coder

The DSP 6000 allows users to specify a choice of ADPCM or an ISO/MPEG source coder. Continuous development work will allow choice of additional source coders. Source coder flexibility allows stations to tailor their sound according to interface or format considerations.

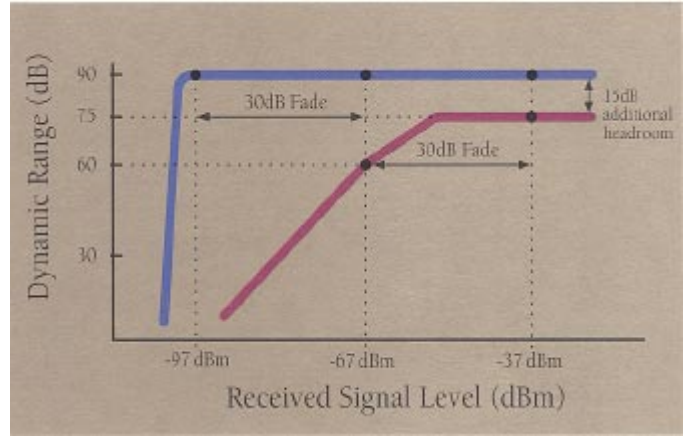
Direct Digital Interconnection

The DSP 6000 has a built-in AES/EBU digital audio interface to accept and/or deliver digital audio from/to other digital audio devices. Such connection eliminates multiple A/D and D/A conversions. ◀

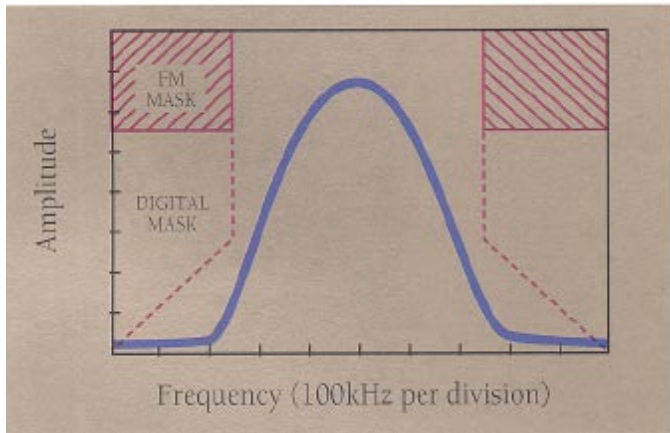
- *CD Quality Digital Audio*
- *90 dB dynamic range*
- *Constant dynamic range for all frequencies
30 Hz – 15 kHz*
- *80 dB crosstalk due to digital multiplexer*
- *THD .01% at 1 kHz*
- *Selectable efficiency of 1, 2 or 3 bps/Hz*
- *Choice of ISO/MPEG or sub-band ADPCM source coder*



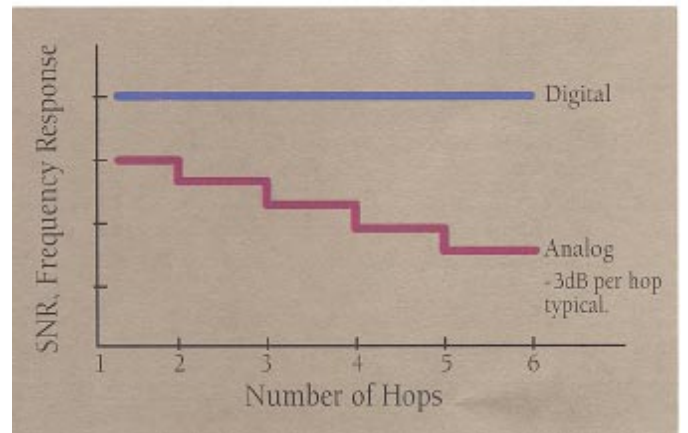
Instant Payback



Constant Signal-to-Noise Ratio (SNR)



Spectral Efficiency



Degradation-Free Multiple Hops

The Challenge of Digital Transmission

Linear 16-bit PCM for broadcasting (15 kHz audio) requires 512 kbps/channel at 32 kHz sampling rate. A typical STL configuration might require two 15 kHz channels for stereo and one 7.5 kHz SCA channel yielding an aggregate data rate of 1280 kbps. It is this high data rate that has made previous attempts at digital STL usage in the 500 kHz (and 300 kHz) bandwidths impractical. The required spectral efficiency suggests that complex coding techniques are necessary to squeeze the data into standard broadcast channels.

- 16-bit linear PCM
- Fits in existing channels
- Optimal partitioning
- State-of-the-art source/channel coding

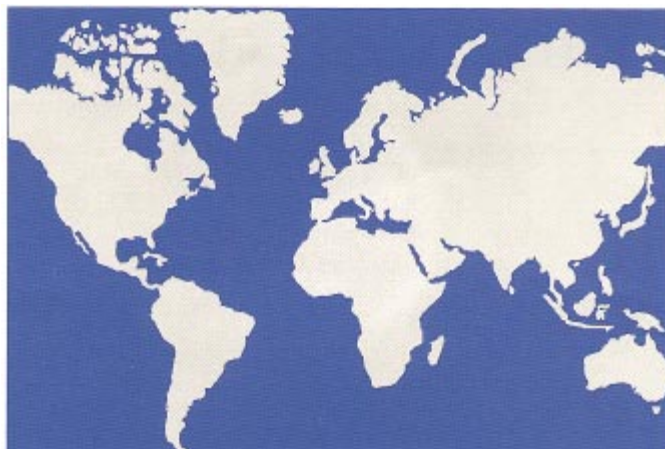
Such coding may be partitioned between source and channel encoding techniques to reduce the burden of complexity on either.

To choose the right source coder, you must accommodate several application-specific requirements: quality, bit rate, delay, bit error protection, post-processing capabilities, and hardware implementation complexity. The DSP 6000 offers a choice of two source coders: the

ISO/MPEG Layer II and a sub-band ADPCM source coder. The sub-band ADPCM encoder uses linear prediction and backward adaptive quantization to encode the difference between successive samples of music. Complexity is moderate as opposed to removing spectral components of music according to a subjective psychoacoustic algorithm; thus, its delay is only 3.8 ms. With lower delay, a professional announcer will have less of a problem listening to his voice off air during a live broadcast. In addition, ADPCM has excellent bit error immunity and gentle error handling, offering great benefits in broadcast radio transmission.

The ISO/MPEG algorithm has been accepted by the radio industry as the standard for digital bit rate reduction. The principle of the ISO/MPEG standard is based upon the psychoacoustic properties of the human ear (i.e., the way the human ear processes information). One of the important characteristics of human hearing is a phenomenon called spectral masking. In a typical audio signal, there will be several frequency components of such

Moseley is the pioneer and world's leading manufacturer of aural microwave links and supervisory control systems. Moseley products are in service in over 100 countries worldwide.



magnitude that they will mask adjacent signals below, thus forming a masking threshold. Since the human ear cannot distinguish any signals below this threshold level, the transmission of bits that con-

tain this information is unnecessary. ISO/MPEG Layer II has become an industry standard for use in digital transmission systems. However, it is a more complex algorithm resulting in a longer coding delay. Care was taken in the design of the DSP 6000 to allow direct digital interface with companion equipment in the broadcast chain. Open architecture, using internationally ac-

cepted I/O standards, allows audio that has already been bit reduced to completely bypass the internal source coding process of the DSP 6000. This feature gives the broadcaster the power to avoid an unnec-

essary cascade of algorithms, resulting in a transmission free from processing artifacts.

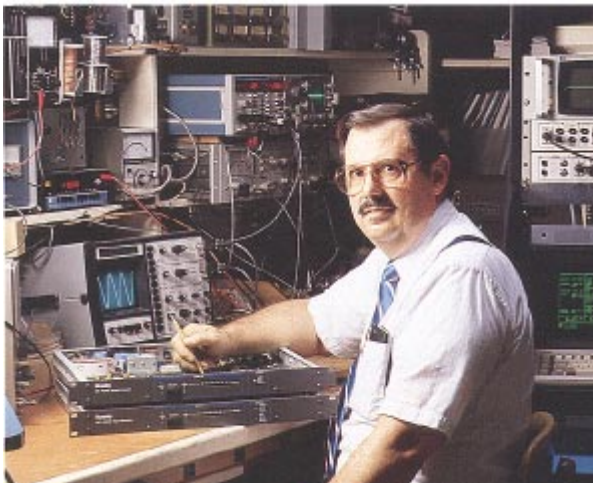
Channel Coding

Efficient spectrum utilization is a primary consideration for the choice of modulation format. The out-of-band spectral energy must be well suppressed to avoid interference of the digital carrier with existing analog services. Linear modulation formats such as QPSK or QAM will regenerate sideband energy when confronted with non-linear Class-C amplification found in conventional FM STLs, thereby limiting their usefulness. In contrast, the DSP 6000 uses a form of Continuous-Phase Frequency Shift Keying (CPFSK), known as partial response. CPFSK is a broad class of digital frequency modulation particularly well suited to FM transmission with non-linear amplification.

Along with high and user-selectable spectral efficiency, CPFSK signaling has good error performance and speed tolerance. Another benefit of partial response CPFSK signaling is that error detection can be rapidly obtained without adding redundancies into the data stream. ◀

- *Choice of ISO/MPEG or ADPCM source coder*
- *CD quality*
- *Low coding delay*
- *Processing friendly*
- *Preserves peaks*
- *Multiple encode/decode capability*

- *Spectrum efficient*
- *Constant envelope*
- *Error detection*
- *Robust*
- *Selectable efficiency*



Moseley's knowledgeable staff of sales, administrative, and technical personnel are an important part of our commitment to problem solving.

System Components

THE DSP 6000 Digital Transmission System consists of the DSP 6000E Encoder, the DSP 6000D Decoder and any Moseley digital-ready STL transmitter and receiver. The DSP 6000E and DSP 6000D can also be used with other FM radios and/or digital circuits. Moseley transmitters and receivers are available in a number of bands in the 140 MHz to 1900 MHz frequency range.

DSP 6000 Encoder

The DSP 6000 accepts up to four audio program channels through active balanced inputs via XLR connectors. Each input, programmable for 7.5 or 15 kHz bandwidths, is routed to a source encoder. The encoder first converts it to 16-bit linear PCM data with a 64-times oversampling A/D converter capable of 95 dB dynamic range. Alternatively, the source encoder also accepts either AES/EBU formatted data or 16-bit linear PCM direct from a digital source. The digitized audio is then passed through a user-defined choice of ISO/MPEG source encoder or the sub-band ADPCM encoder. The source encoder output feeds the channel encoder. The channel encoder will also accept data from an external source from 64 kbps to 512 kbps. This facilitates transmission of other services such as basic rate ISDN or Musicam encoded material.

Next, the encoded material is routed through the channel

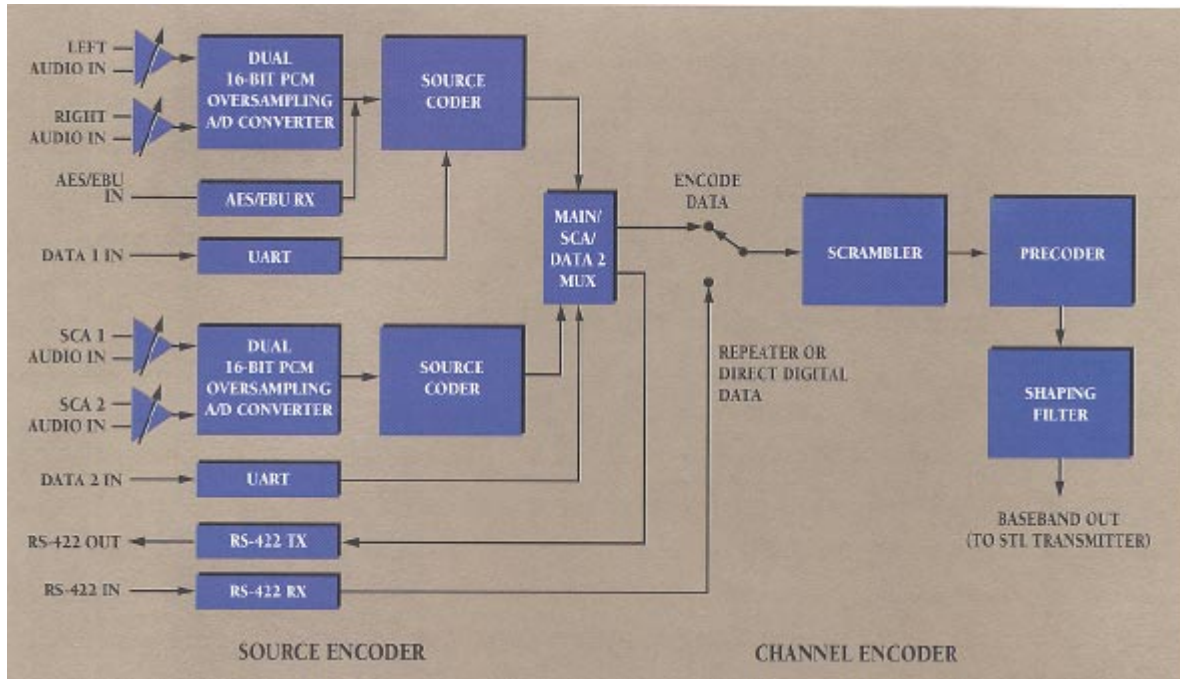
encoder's scrambler to prevent discrete spectral lines from interfering with adjacent channels. The data is precoded to prevent error propagation in the receiver due to correlative properties of each received bit.

Finally, a raised cosine low-pass filter Nyquist-shapes the data to minimize inter-symbol interference. The shaped data at the channel encoder's output drives the STL transmitter.

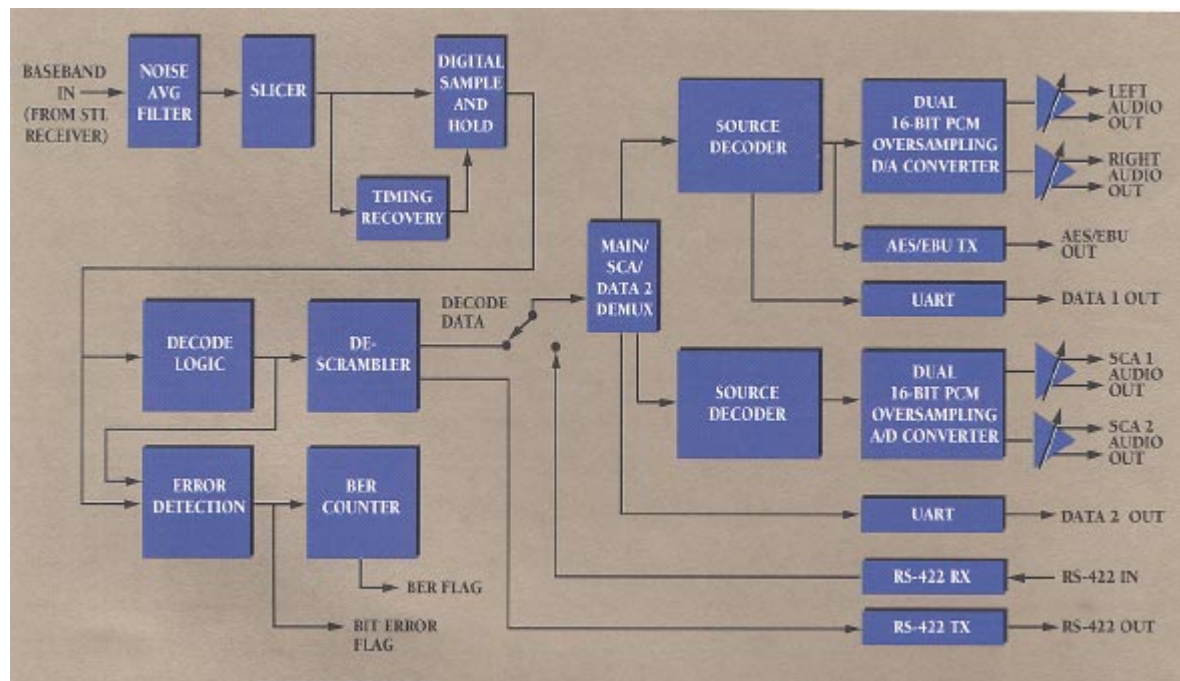
DSP 6000 Decoder

The recovered baseband signal from the STL receiver is first noise averaged by a raised cosine shaped low-pass filter in the channel decoder. From the recovered tri-level baseband signal, both bit clock and data recovery are accomplished. The bit clock synchronizes the rest of the system to the recovered data. Bit-by-bit detection is performed on the filtered baseband signal using a digital sample-and-hold. Following data sampling, decode logic reconstructs and descrambles the data for output to the digital source decoder. The channel decoder also provides bit-error and BER threshold detection for signal quality indication.

The recovered data is fed to the source decoder which performs the inverse function of the source encoder. Sixteen-bit linear PCM is output as either formatted AES/EBU or unformatted serial data. Analog output is also available by use of a 4-times over-sampling D/A converter. ◀



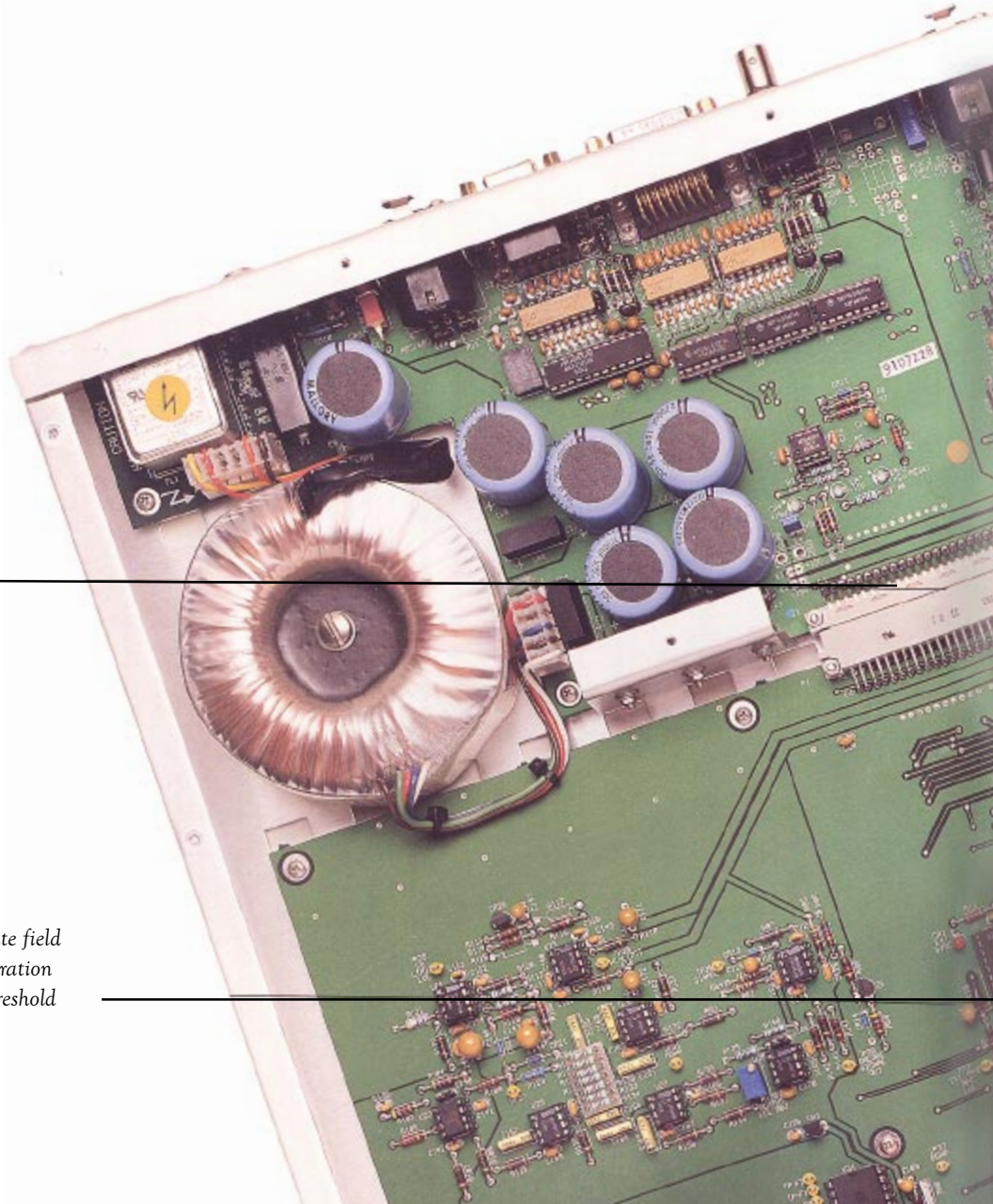
The DSP 6000 source and channel encoder fully configured with four audio and two data channels. AES/EBU stereo 16-bit PCM data input for direct digital interface.



The DSP 6000 channel and source decoders with four audio and two data channels. Comprehensive bit-error rate monitoring. AES/EBU output for direct digital interface.



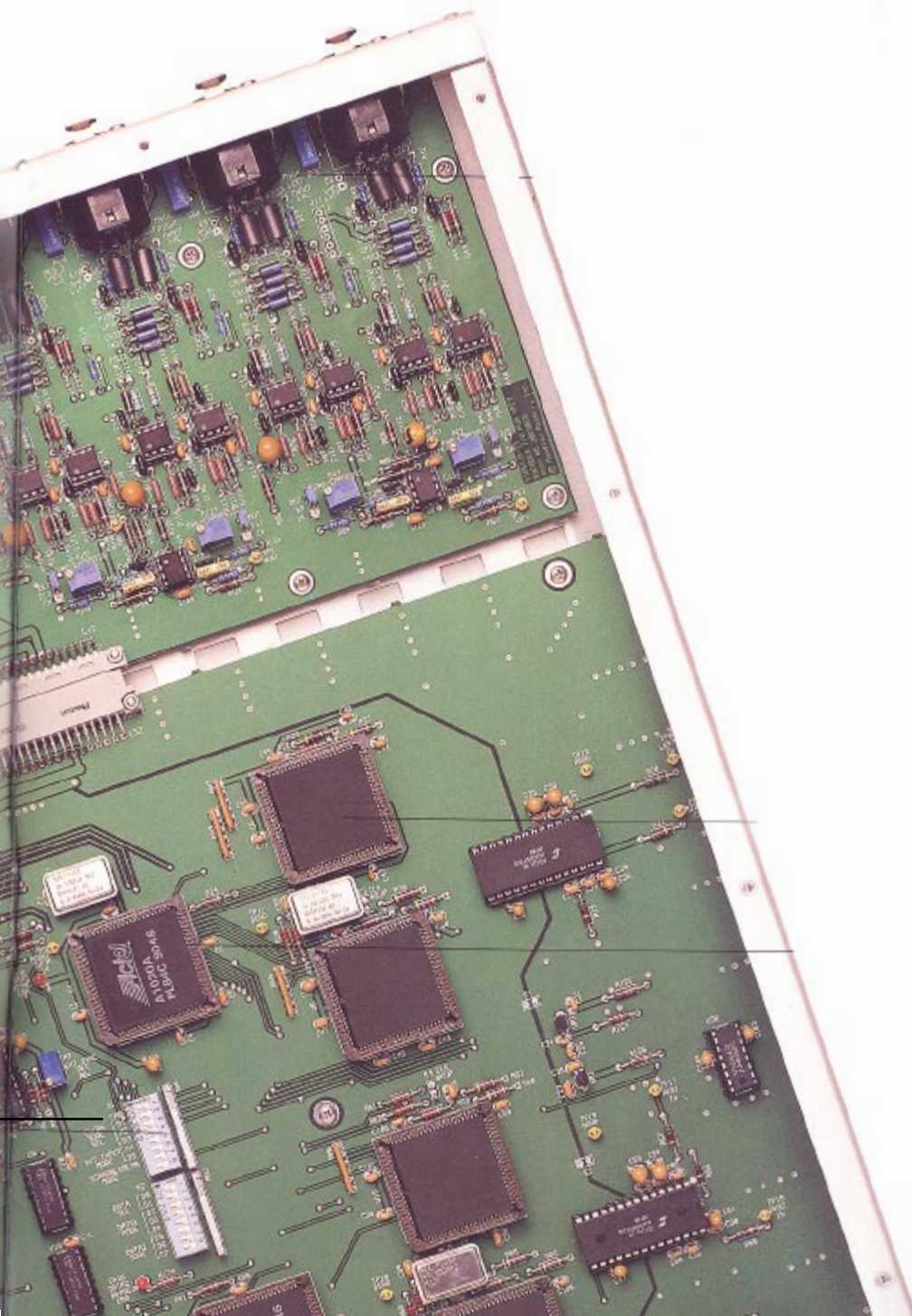
*Built-in RS422 driver
for digital telco circuits*



*Integrated construction
eliminates harnessing to
enhance reliability*

*Dip switches facilitate field
expansion, reconfiguration
and bit error rate threshold
selection*

Extensive Status Monitoring



to four audio pairs)

— Plug-in choice of source coder sub-band ADPCM or ISO/MPEG

— Use of field programmable gate array reduces circuit density, enhances reliability and facilitates field upgrades

The Future Sounds Good

With so many audio choices available to the consumer, radio broadcasting's future is increasingly dependent on reaching target audiences with CD-quality audio. As a vital link in the broadcast chain, the DSP 6000 digital STL system allows the progressive radio broadcaster the ability to maintain the digital audio sound he's worked so hard to develop at the studio and to convey it digitally to the transmission facility.

Open Architecture

The use of international standard digital I/Os allows the DSP 6000 to utilize other digital hardware and source material. This commitment to open architecture gives you the power to make appropriate choices, and to make these solutions work with both your existing and future investments.

Fractional T1

A built-in V.35/RS-422 interface opens the way for utilizing the DSP 6000 with digital circuits leased from the telephone company. The limited number of RF channels in the STL band, coupled with the need for a digital link to the transmitter, has forced some broadcasters to use digital circuits as an alternative to a traditional RF STL. Use of digital circuits (Full T1 or

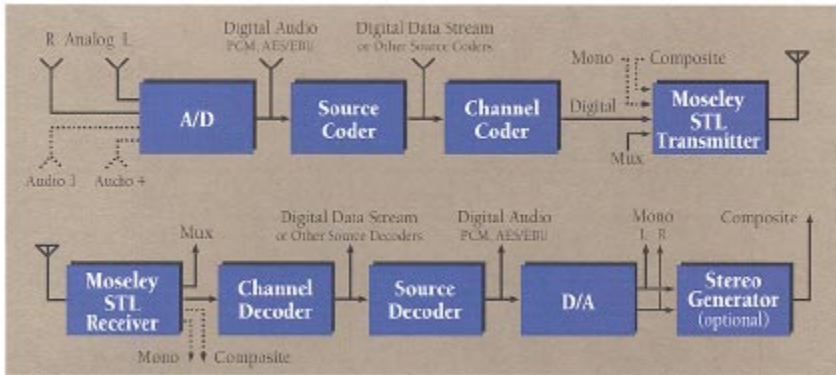
Fractional T1) entails a monthly payment dependent on the number of time slots used for the transmission. The DSP 6000 uses only a quarter of the number of time slots used by conventional digital systems. A lower data rate offers substantial savings over leased line costs.

Digital Repeaters

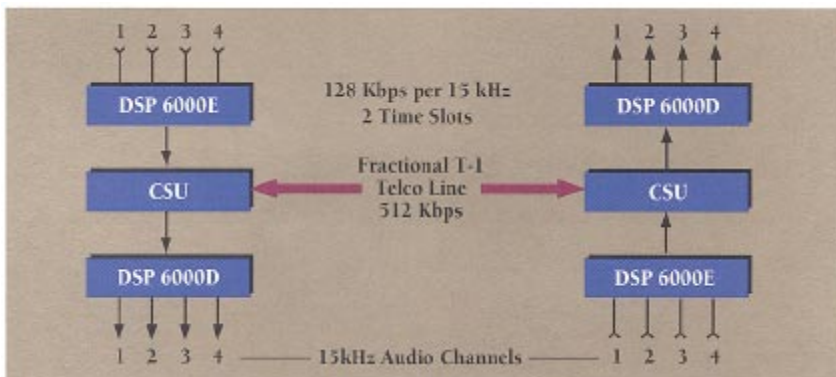
At a midpoint in a repeater configuration, the DSP 6000 permits digital regeneration and/or audio to be dropped off and inserted. Digital regeneration allows multiple hop STLs to be free of the audio degradation normally associated with analog signal regeneration. Creating a cost-effective digital network with multiple program and transmission points is now possible.

The Digital Transmission Chain

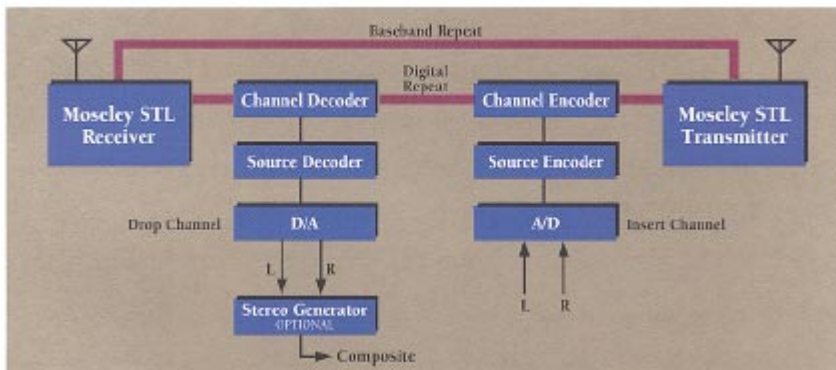
With the ability to digitally link all key elements of the broadcast chain, the DSP 6000 is the driving force toward the culmination of an all-digital radio station. The latest in digital audio processing can be used with the DSP 6000, either at the transmitter or the studio. By eliminating traditional audio processing problems like overshoot and lack of peak level preservation, the digital radio station has the advantage—the *ultimate advantage*.



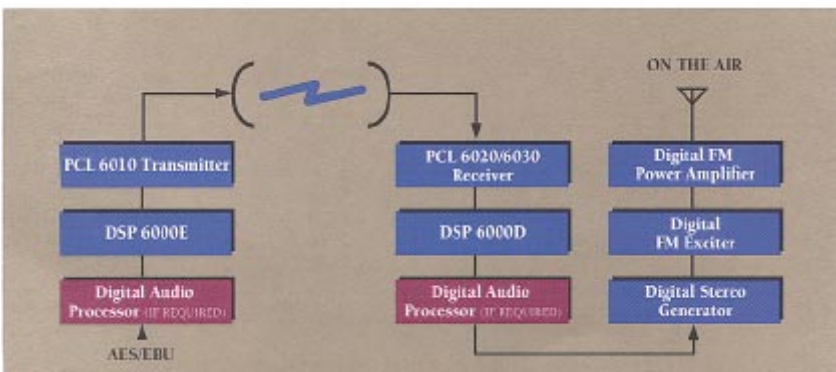
Open Architecture



Fractional T1



Digital Repeaters



The Digital Transmission Chain

Redefining Transmitters & Receivers

The frequency-agile PCL 6020 and PCL 6030 STLs are ideal RF companions to use with the DSP 6000 Encoder and Decoder. Built to withstand the harshest RF environments and available in frequencies from 145 MHz to 1.9 GHz, they have become the choice of broadcasters worldwide.

As the leaders in STL technology for 30 years, Moseley has engineered the PCL

6000 Series to offer unique and innovative features. Both the PCL 6010 transmitter and the PCL 6020 and 6030 receivers use a synthesized reference oscillator, thereby eliminating fixed-frequency crystals.

The physical module count on the PCL 6000 Series is kept to a minimum for highest MTBF without comprising ease of operation or maintenance.

All oscillators, up-converters, discriminators and audio sections are individually isolated by jumpers for

alignment or repair.

PCL 6010 Transmitter

The PCL 6010 uses direct modulation techniques. A synthesized reference oscillator is used for FM generation. Conversion of the FM-modulated oscillator frequency to the final output frequency is done via an up-converter/mixer technique instead of the usual frequency multiplication. For long STL paths, an external 40 Watt amplifier is available.

PCL 6020 Receiver

The dual-conversion PCL 6020 Receiver uses an FM quadrature detector to provide your system with maximum fidelity. Excellent selectivity characteristics ignore adjacent channels in your area. The receiver IF bandwidth can be set for channel spacing of 100 kHz to 500 kHz, depending on RF congestion in your area and channel availability.



PCL 6030 Receiver

The triple-conversion PCL 6030 Receiver uses a digital pulse-counting discriminator for extremely low distortion and low noise characteristics. Superior selectivity characteristics ignore adjacent channels 25 dB stronger than your received signal. An optional PCL 6060 front-end module with high selectivity and adjustable gain preamp is recommended for front-end protection at cellular telephone sites. ◀

- User programmable
- Frequency agile
- Excellent selectivity
- Switchable monaural/composit/digital operation
- Comprehensive metering
- Built-in receiver transfer circuitry
- 145 MHz to 1.9 GHz
- 100-500 kHz channel spacing

Specifications

System

Frequency Response	
Main Wide	20 Hz to 15 kHz \pm 0.2 dB.
Main Narrow	20 Hz to 7.5 kHz \pm 0.2 dB.
Aux Wide	20 Hz to 15 kHz \pm 0.2 dB.
Aux Narrow	20 Hz to 7.5 kHz \pm 0.2 dB.
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Distortion	< .01% at 1 kHz.
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Dynamic Range	90 dB static encoder, 90 dB static decoder
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Channel Crosstalk	-80 dB
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Level Stability	< 0.2 dB
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Data Coding Method	Choice of ISO/MPEG or sub-brand ADPCM.
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Sample Rate	32 kHz, 44.1 kHz, 48 kHz
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Time Delay	< 3.8 ms ADPCM, 49 ms ISO/MPEG.
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Bit Error Immunity	> 10^{-4} for no subjective loss in audio quality.
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Sensitivity	-93 dBm at 10^{-4} BER (PCL 6000 at 256 kbps), h = 0.5; -97 dBm, h = 0.7 at 1 bps/Hz.
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System Gain	132 dB with PCL 6010 Transmitter at 256 kbps, 890-960 MHz. 133 dB at 140-470 MHz at 1 bps/Hz.*
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Spectral Efficiency	1 bps/Hz std., (higher efficiency, contact factory)
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Data Channels	
Data 1	RS-232, selectable baud rates from 4800, 2400, 1200, 300 baud.
Data 2 (optional)	RS-232, selectable baud rates from 9600, 4800, 2400, 1200, 300 baud.
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Program Channels	1-4 channels. Channels programmable as 15/7.5 kHz for Main and 15/7.5 kHz for Aux. Two 15 kHz channels standard.
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Transmission Rates	Selectable from 64, 128, 205, 256, 273, 341, 410, and 512 kbps depending on channel configuration.
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Power Requirements	115/230 Vac \pm 10%, 50/60 Hz.

* System gain 7 dB less for 2 bps/Hz and 14 dB for 3 bps/Hz.

Transmitter and Receiver

STL	Choice of either a digital-ready Moseley PCL 6000 or PCL 606 STL transmitter and receiver.
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Frequency Range	140-176, 200-240, 300-330, 440-470, 890-960 MHz, 1.5-1.7 GHz.
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Power Source	120/240 Vac, \pm 10%, 50/60 Hz 100 watts. 12/24 Vdc (contact factory).
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RF Output	
890-900 MHz	10 watts maximum, 5 watts minimum.
140-470 MHz	15 watts maximum, 7 watts minimum.
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RF Output Connector	Type N female, 50 ohm.
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Frequency Stability	Better than 0.00025%, 0° C to +50° C.
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Spurious and Harmonic Emission	More than 60 dB below carrier level.
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Modulation Capability	Analog, digital, or combined.
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RF Input Connector	Type N female, 50 ohm.
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Sensitivity	See system specs.
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Channel Spacing	User specified 100-500 kHz. Depends on data rate.
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Adjacent Channel Rejection (500 kHz)	Digital to digital 35 dBc. Digital to analog 20 dBc. Analog to digital 30 dBc.

FM Radio Requirements

Program Bandwidth 3 dB	< 10 Hz to > R/4 Hz where R=data transmission rate.
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Amplitude Variation	< 1 dB Transmitter and Receiver.
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Group Delay Variation	< 1 μ s Transmitter, < 2 μ s Receiver.
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THD and IMD	< 3% dB Transmitter and Receiver.
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Signal-to-Noise Ratio	> 30 dB Transmitter, > 50 dB Receiver.
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Program Input/ Output Impedance	> 1 kohm Transmitter, < 100 ohm Receiver.
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Deviation	64 kHz to 512 kHz depending on system configuration.
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Frequency Stability	Depends on application.
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Spurious and Harmonic Emission	Depends on application.

Specifications subject to change without notice.

DSP 6000E Encoder

Audio Input	Electronically balanced input, XLR type, female. CMRR > 60 dB.
Audio Level	Each channel adjustable from -10 dBu to +18 dBu, multi-turn, rear panel accessible.
Digital Audio Input	AES/EBU standard, RS422A transformer balanced XLR type, 110 ohms input impedance. Sample rate 32 kbps.
Source Coder Digital Output	RS-422
Data Inputs 1 & 2	RS-232, 9-pin D-type, female.
Interface Input	RS-422, 15-pin D-type, female. Input for regenerated data and clock from decoder in R PTR MODE. For unformatted linear 16-bit PCM data and frame sync in PCM MODE.
Status Output	15-pin D-type female provides active-low indication of system fault, clock fault, modem fault, processor fault, mux fault, AES/EBU fault, and DATA channel fault.
Indicators Bargraph Display	Dual ten-LED display indicates peak program level relative to 100% full scale (0dB) A/D input overload level for audio program channels. Display switchable from MAIN program to AUXILLIARY program channels.
Main	Indicates bargraph is displaying main Left/Right program levels.
Aux	Indicates bargraph is displaying auxilliary Aux 1/Aux 2 program levels.
AES/EBU	Indicates status of digital audio input.
Fault	System fault alarm.
Power Requirements	115/230 Vac @ 50/60 Hz, 30W.

Digital Circuits

Fractional T1/CEPT-1 Data Rate	64 kbps/7.5 kHz, 128 kbps/15 kHz. Choice of 1, 2, or 4 channels only.
Digital Interface	RS422/EIA530
Fractional T1 CSU	Any fractional T1 CSU that adheres to the T1 trunk format (with or without ESF) can be used to access the T1 network. Multiport CSUs can be used for routing different channels.

DSP 6000D Decoder

Audio Output	Electronically balanced input, XLR type, male.
Audio Level	Each channel adjustable from -10 dBu to +14 dBu, multi-turn, rear panel accessible.
Data Inputs 1 & 2	RS-232, 9-pin D-type, female.
Digital Audio/Interface Outputs	RS-422, 15-pin D-type, female. Output for regenerated data and clock (to encoder) in R PTR MODE. For unformatted linear 16-bit PCM data and frame sync in PCM MODE. Output for AES/EBU digital audio. RS-422 transformer balanced XLR type.
Status Output	15-pin D-type female provides active-low indication of system fault, loss-of-data-sync, loss-of-signal, clock fault, modem fault, processor fault, BER fault, Demux fault, mute status, mute relay contacts, and mute disable input.
Indicators Bargraph	Dual ten-LED display indicates peak program level relative to 100% full scale (0dB) D/A output level for audio program channels. Display switchable from MAIN program to AUXILLIARY program channels.
Fault	System fault alarm.
Main	Indicates bargraph is displaying main Left/Right program levels.
Aux	Indicates bargraph is displaying auxilliary Aux 1/Aux 2 program levels.
Sync	Bicolor indication for data synchronization.
Signal	Bicolor indication for incoming baseband signal.
Data Error	Indicates received data error.
BER	Indicates alarm condition when received bit error rate exceeds preset level (selectable from 10^1 to 10^6).
Power Requirements	115/230 Vac @ 50/60 Hz, 30W.

Digital Stereo Generator Option

Channel Separation	> 70 dB (30 Hz-15 kHz).
SNR	-90 dB typical.
Distortion	0.03%.
Output Level	3.5V peak to peak unbalanced.

These specifications are subject to change without notice.

