

CDQ2000 Reference Manual

Multi-rate Digital Audio CODEC

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Crystal Clear Digital Audio™

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Introduction

The CDQ2000 enlivens your digital audio transmissions by giving you CD-quality sound. This universal audio coder and decoder (codec) converts analog audio signals into their digital equivalent, compresses the digital stream for transmission, expands the stream upon receipt, and converts the digital signal back into an analog signal. Believe it - this seemingly simple process of the CDQ2000 gives you the clearest and highest sound quality possible in today's digital transmission world.

An analog signal that may be mono or stereo is input to the CDQ2000. You can set the codec to provide the best compromise between quality and analog bandwidth. The CDQ2000 encoder then uses its 16-bit analog-to-digital converter (ADC) to sample the input audio signals at the professional rate of 48,000 samples per second.

The CDQ2000 compresses the audio signal by using the ISO MPEG layer 2 algorithm (MUSICAM). The resulting digital signal is output through V.35, X.21, or RS422 digital connectors. Users can select the output digital bit rate from a range of 56 kilobits per second (kbps) to 384 kbps, depending of the quality of the desired output.

The digital bit stream is transmitted via Switched 56, or ISDN digital telephone lines, or via satellite to the receiving end. There, the CDQ2000 decoder expands the compressed digital signal into linear form. It then converts the digital signal into analog with its 18-bit digital-to-analog converter (DAC). Both the ADC and the DAC circuits of the CDQ2000 use the most advanced oversampling techniques to provide the ultimate in digital audio.

The CDQ2000's unique feature, a built-in inverse multiplexer, can split a 112 kbps digital output into two separate 56 kbps data streams for transport over two independent Switched 56 lines. It can also split a 128 kbps digital output into two separate 64 kbps data streams for transport over two independent ISDN lines. This feature keeps transmission costs low since a 20 kHz stereo signal can be sent over two switched digital lines. These lines may have up to 0.5 second differential delay between them, but the CDQ2000 realigns the signals for low cost digital audio of the highest quality.

Since the CDQ2000 exploded onto the digital scene, it has been globally praised as an extraordinary codec of distinguishable excellence.

MUSICAM Overview

Introduction

Developments in the fields of consumer audio electronics and professional audio processing have been increasingly influenced by digital technology. Until five years ago, developments in the field of source coding were mainly restricted to the bit-reducing coding of speech signals for telecommunications applications.

Today, source coding techniques are playing an even greater role in the field of high quality digital audio. The reasons for this are the direct relationship between the low bit rates associated with compression and the costs associated with the transmission and storage of compressed audio.

The bit-rate for high-quality stereo audio signals (1,411 kbs for a CD) can now be reduced by the MUSICAM algorithm to about 200 kbs. This is the result of major progress in the development of source coding techniques that utilize knowledge of the human ear. This means that the average quantization of the audio signal at a sampling rate of 44.1 kHz would be approximately 2 bits per sample in the mono channel instead of the 16 bits per sample used in CD's. Despite this high reduction in the bit rate, no quality differences are discernible to a trained ear. A slight impairment only becomes audible at higher compression rates. Additionally, MUSICAM offers the flexibility of independently adjustable audio sampling rates (32 kHz, 44.1 kHz, 48 kHz...) and digital bit rates (56 kbs, 64 kbs, 112 kbs, 128 kbs, 192 kbs, 256 kbs, 384 kbs...) as well as embedded data within the audio bit stream. All of these features are incorporated in the recently approved ISO MPEG audio standard. No other audio compression algorithm has undergone the scrutiny and testing subjected to MUSICAM as a result of the ISO selection process. The ISO standards committee has selected a truly universal digital audio source coding system with the flexibility to meet different system demands. Current and future audio systems adhering to the ISO MPEG audio standard will be able to interoperate easily and reliably. This will allow manufacturers to build sophisticated audio equipment and consumers to purchase hardware without the fear of obsolescence.

MUSICAM Compression Concepts

The main principle of MUSICAM is the reduction of redundancy and irrelevance in the audio signal. Every audio signal contains irrelevant signal components that have nothing to do with the identification of the audio signal (i.e., determination of timbre and localization). These irrelevant signals are not significant to the human ear and are not required by the information processing centers in the brain. The reduction of irrelevance means that these signal components are not transmitted. This results in a lower bit rate without any perceived degradation of the audio signal. Furthermore, it is possible to allow a certain degree of quantizing noise that is inaudible to the human ear due to the masking effects of the audio itself.

Every audio signal produces a masking threshold in the ear depending on a time varying function of the signal. To understand this masking effect, the concept a masking tone must be defined. A masking tone is simply a high amplitude audio signal occurring over a relatively narrow frequency span and is often called a masker. Typically, in an audio signal there exists a number of these masking tones occurring at several different frequencies.

A masking tone renders smaller amplitude tones close to it inaudible due to its masking effect. The exact shape of the masking effect is called the masking threshold. The aggregate of all the maskers defines a global masking threshold and the parts of an audio signal below the global masking threshold are inaudible. They are said to be masked and therefore need not be transmitted. Other signal components above the masking threshold only require the level of quantization to keep quantization noise below the masking threshold, and thus the quantization induced noise remains inaudible. Quantization noise can be better adapted to the masking threshold of the human ear by splitting the frequency spectrum into sub-bands.

The quantization of the analog time samples required for each sub-band is dependent on the minimum masking value in each sub-band. This minimum masking level is a measure of the allowed quantization noise that is just below the level of perceptibility. Sub-bands whose desired signals are well below the masking threshold (and are thus irrelevant for the human ear) do not need to be transmitted.

In each 24 millisecond period, a calculation of the masking threshold is performed for each sub-band. This threshold is then used to compute the psychoacoustically best allocation of the available bits. This process is called dynamic bit allocation. Audio data is quantized using the dynamic bit allocation and thus the required bit rate for time-variant audio signal's changes continuously due to the changing masking threshold. If there is an insufficient number of bits to hide the quantizing induced noise completely, then the noise is placed in the least objectionable place in the audio sample. If there is an excess number of bits, then the extra bits are used to reduce the quantizing induced noise to as low as possible level. The allocation of the extra bits is crucial and allows multiple encode-decode cycles as well as post production of the audio.

The total transmitted bit stream contains quantized audio values as well as auxiliary information describing bit allocation and scale factors, all of which are required by the decoder to reproduce the audio information.

The scale factors are determined by searching for the maximum sampling value in each sub-band and quantizing the result using 6-bit sampling. The scale factors have a dynamic range of 120 dB that is sufficient for future encoding for quantized PCM signals using up to 20-bit sampling yet still retain their dynamic range. All necessary information is encoded into MUSICAM frames each of which represents about 24 milliseconds of real-time audio.

All the complex calculations of the MUSICAM algorithm are performed by the encoder. Decoders are designed to be universal. MUSICAM decoders can be constructed which correctly decode and play back audio information that has been encoded by a range of MUSICAM encoders.

This aspect of the MUSICAM algorithm is crucial because it enables refinements in the encoding process to further improve performance without impacting decoders that are already installed.

Performance Considerations

Introduction

Before discussing the various quality aspects of MUSICAM, it is necessary to define the terms used to represent the field of use of the audio. The 4 commonly discussed fields of use are:

- Contribution
- Distribution
- Emission
- Commentary

The term contribution grade is used to describe quality suitable for digital mastering. Its use would be in the transmission of a digital master from one archive to another. It is assumed that the original copy is in a 16 bit linear PCM format and it is to be compressed, transmitted, decompressed and stored in a 16 bit linear PCM format at the distant end. Because the audio is the source of future compression/decompression cycles, any contribution grade compression system must be able to withstand many encode-decode cycles and post production without any apparent degradation.

Distribution grade systems are used to transmit audio between two storage devices. However, the number of encode-decode cycles is limited to only a few. Distribution grade systems are used when the number of audio compression-decompression cycles is limited.

Emission grade systems are used when there is only one compression-expansion cycle anticipated. This is the case when audio is compressed and transmitted from one place to another, decompressed and stored on an analog tape and the only future manipulations done are in the analog domain.

Commentary grade systems are used for transmitting voice grade audio.

These definitions make no mention of the analog bandwidth or the exact definition. They are vague terms used to describe ability of the audio to withstand multiple encode-decode cycles. In all cases, the compressed audio is assumed to be indistinguishable from the original.

ISO Background

The only independent measurements of audio quality of MUSICAM types of compression systems have been done by the MPEG ISO committee. Four algorithms in July of 1990 were tested and the winner according to the rules of the tests was MUSICAM. This algorithm was adopted and it was agreed that, to the extent possible, the best features of the second place algorithm, ASPEC, would be incorporated into MUSICAM to produce the final ISO standard.

The ISO committee decided to have a layered standard with 3 layers. Layer 1 is a very simplified version of the original MUSICAM algorithm. Layer 2 is essentially the MUSICAM algorithm as tested, and Layer 3 is a modification of Layer 2 that includes various features of ASPEC. It was anticipated that the resulting audio quality would improve with higher layer number. After the layers were defined, they were implemented according to the standard and each layer was tested in the May 1991 tests.

The results of these tests were surprising because Layer 3 scored lower than Layer 2. It has recently been decided that additional work on Layer 3 was needed and that layer would be retested in December of 1991. Layers 1 and 2 have been frozen in their present state because they have met their design objectives. As a result of the ISO effort, the MUSICAM algorithm is now properly called the MPEG Layer 2 compression algorithm.

It is clear from the most recent ISO tests that no compression scheme performs acceptably at 64 kbs. Work at that bit-rate is the subject of further research and will be addressed in a future standard.

The intensity or joint stereo mode of compression supported by Layer 2 (called Layer 2A) was not tested during the May 1991 tests.

It is important to recognize that the ISO tests have provided a wealth of knowledge about the MPEG Layer 2 algorithm. Other algorithms such as SEDAT, AC-2 and APT-X did not even participate in the ISO tests and their strengths and weaknesses are unknown. It is certainly clear that MPEG Layer 2 has been demonstrated to be a superior algorithm. This claim can be supported by a large body of test data. Other algorithms have little or no independent test data to substantiate their quality claims.

Quality vs. Bit Rate

The MUSICAM design allows the digital bit-rate, analog bandwidth and quality to be generally related by the formula

$$\text{Digital Bit-Rate Quality} = \frac{\text{Digital Bit-Rate}}{\text{Analog Bandwidth}}$$

As indicated above, the quality increases as the bit-rate increases and the analog bandwidth is kept constant. Similarly, if the digital bit-rate is kept constant, and the analog bandwidth is decreased, then the quality improves.

The ISO test in Stockholm in May 1991 has demonstrated that at a digital bit rate of 256 kbs per stereo channel; MPEG Layer 2 is statistically identical to the original signal. This means that the panel of approximately 60 highly trained listeners could not distinguish the original uncompressed source material from the audio compressed by the MPEG Layer 2 algorithm. The conclusion of the ISO tests (at 256 kbs per stereo channel) was that MPEG Layer 2 is transparent.

MPEG Layer 2 scored 5 on the MOS (mean opinion score) scale where the lowest is 1 and the highest score is 5.

It is important to note that no other algorithm tested at ISO (including ASPEC) was considered transparent in the 256 kbs stereo tests. The ISO tests were conducted on stereo channels composed of two mono channels so that the combined bit rate was 256 kbs per stereo channel. The audio quality at 192 kbs was determined by ISO to be 4.5 on the MOS scale using stereo encoding and 2.0 for a mono channel at 64 kbs.

The MPEG Layer 2 algorithm provides the following qualities at various bit rates.

- contribution 384 kbs (stereo, Layer 2)
- distribution 256 kbs (stereo, Layer 2)
- emission 192 kbs (stereo, Layer 2A)
- commentary 64 kbs (mono, Layer 2)

The classification of 192 kbs for the emission grade is based on recent work at the IRT (Institute for RundfunkTechnique) and relies on the intensity (joint) stereo coding technique for additional compression.

Tolerance to Transmission Errors

The ISO MPEG Layer 2 data block consists of two parts. The first is the header and consists of framing, bit allocation, scale factors and other side information. The second part of the frame is the audio data. In the case of 256 kbs per stereo channel, the length of a 24 millisecond frame is 6144 bits, the header part of the frame is approximately 300 bits and the remainder of the frame is the audio data. The bit integrity of the entire header is vital since it defines the layout of the remainder of the frame. Any bit error in the header causes degradation because the following parts of the frame would be decoded incorrectly and thus 24 milliseconds of audio would be lost.

An error in the data part of the frame can range from imperceptible to just barely noticeable. This is because a single bit error only affects a single data sample and thus only a very small time. If the bit error occurs in the least significant bit of the data sample, the effect of the error is minimal. However, if the error occurs in the most significant bit (the sign bit) then the effect is more pronounced.

The header of an MPEG frame is protected by an error protection polynomial and provides the ability to detect errors that occur in the header. The data part of the frame is unprotected and any error occurring in the data part of the frame remains. The error strategy used for the ISO MPEG system is as follows. If an error is detected in the header, the last frame (24 milliseconds) of audio is repeated. If, in the succeeding frame, an error is detected in the header, the second and all succeeding frames with errors are muted. This error mitigation technique has been shown to be effective for bit rates of approximately 10^{-5} . This error rate represents error rates easily achievable by transmission systems. Using this strategy, there is a smooth degradation of the

audio quality as the error rate increases until the error rate becomes excessive at this point the audio output mutes.

Tolerance to Multiple Processing

To understand the effect of multiple encode and decode cycles it is important to review the predominant effect that allows MPEG audio to achieve its compression. This is the hiding of quantization noise under a loud signal. MPEG audio adjusts the degree of quantization induced noise in each sub-band and thus hides more noise (uses fewer bits) in the sub-bands that contain large amounts of audio energy.

The quantizing noise raises with each encode and decode cycle and after a sufficient number of cycles, the noise level becomes perceptible. The degradation process is gradual and depends upon level of the quantizing noise on the original. For example the following table list the approximate numbers of total encode and decode cycles before the noise becomes significant.

Bit Rate	Number of codings
384 kbs	15
256 kbs	5
192 kbs	2
128 kbs	1

It is important to understand that these are approximate and the exact number depends highly on the source material.

Post Production Processing Effects

Post production processing of compressed audio is a complicated effect to model. For example, an equalizer changes the level of a range of frequencies, while limiting and compression are non-linear processes. Very little test data is available to ascertain the effects of post processing. Private communications with the IRT suggest that MPEG layer 2 is robust against the effects of post processing and the degree of robustness depends on the compression rate. In particular, 384 kbs audio is unaffected by post processing while 128 kbs audio is somewhat sensitive to post processing. It is not easy to define tests to measure the effects of post processing but an international standards body (CCIR) is specifically designing test to determine the effects of both transcoding and post processing. These tests were conducted in November of 1991 and represented the first time such tests were performed by an independent organization.

MPEG audio represents the most tested, documented and reviewed audio compression algorithm in the world. It is significant to note that no other compression technique has survived this crucial review process as well as the MPEG algorithm and, many other algorithms have elected not to participate in this review process. It is precisely these untested algorithms that make the boldest

claims. MPEG audio provides the security of the international review process to insure the highest quality audio possible with today's technology.

The MUSICAM Advantage

The MUSICAM digital audio compression algorithm has been designed to take advantage of future advances in psycho acoustic research. To make this possible, the decoder is designed to be a slave to the encoder. This technique allows the entire system to be upgraded by simply changing the encoder software. Once this change is made, the entire network is upgraded and the encoder enhancements are reflected at the output of all decoders.

The MUSICAM algorithm is designed to operate at multiple bit-rates. This gives the user the ultimate flexibility to make the tradeoff between quality and cost. The use of higher bit-rates (384 kbs) allows nearly an arbitrary number of transcodings and extensive post processing while still maintaining transparency. The middle bit-rates (256-192 kbs) allow lesser amounts of manipulation while the lower bit's rates (128 kbs) are the most sensitive to these effects. As advances in the research progress, today's bit-rates required to achieve a desired quality will decrease and the ease of MUSICAM to accommodate these advances provides a significant advantage. This is being demonstrated by the research into intensity coding of stereo signals. This shows that the data rate of 192 kbs for stereo signals will most likely be the new standard rate for transparent audio and will supplant the 256 kbs rate accepted as the standard today.

MUSICAM is able to embed other information within the audio bit stream. Again, in the MUSICAM design, the data rate of this ancillary information is completely flexible and thus is entirely in the hands of the system designer. This data rate is completely determined by the encoder and thus may be changed at any time with no modifications to the decoders. The inclusion of data in the audio bit stream reduces the bits available for audio data and thus the system designer can make the delicate tradeoff between the ancillary data rate and audio quality.

The flexibility of MUSICAM to adapt to current and future needs is a powerful feature necessary to prevent the obsolescence of any system based on it. There is now no need to divine future system needs because the system can be easily changed to accommodate its ever changing requirements.

Installation

Figure 1 shows a block diagram of the CDQ2000 containing an encoder and a decoder. Figures 2 and 3 depict the separate encoder and decoder respectively. Figures' 4a and 4b show two popular operating modes of the CDQ2000. Figure 4a shows the connection of the combined encoder/decoder and Figure 4b illustrates the connection of the CDQ2000E encoder with a CDQ2000D decoder.

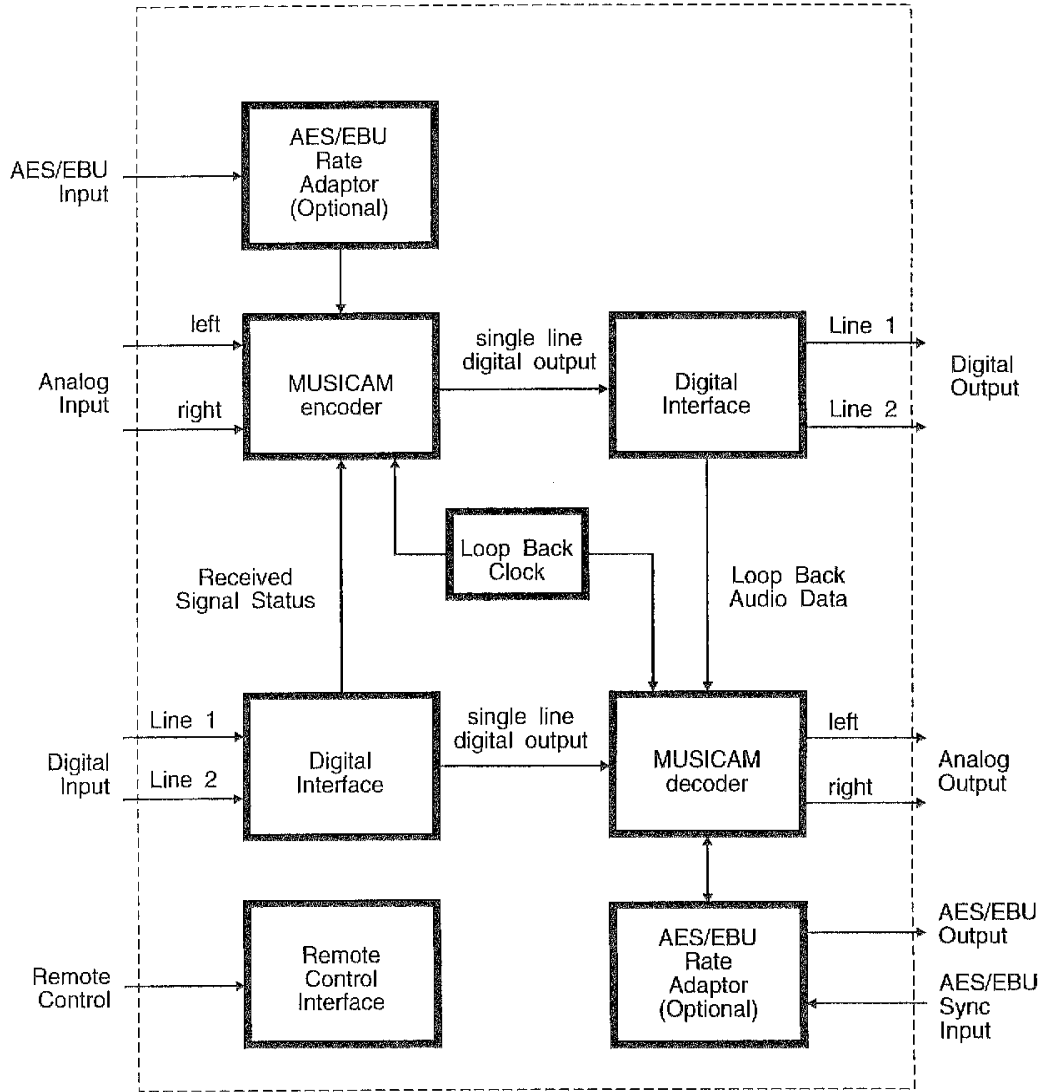


Figure 1
CDQ2000ED Block Diagram

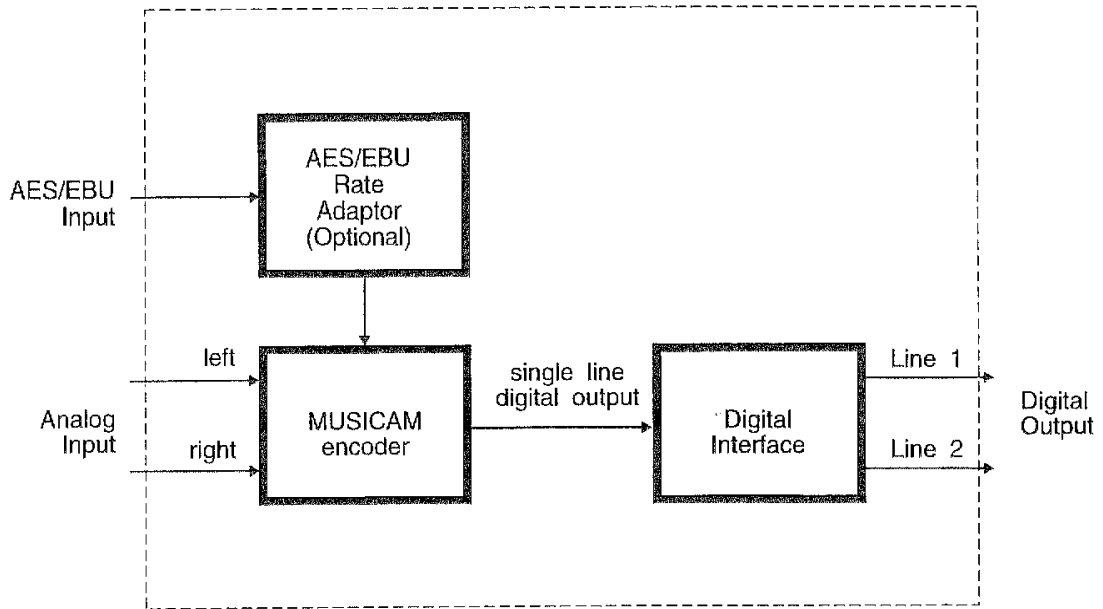


Figure 2
CDQ2000E Block Diagram

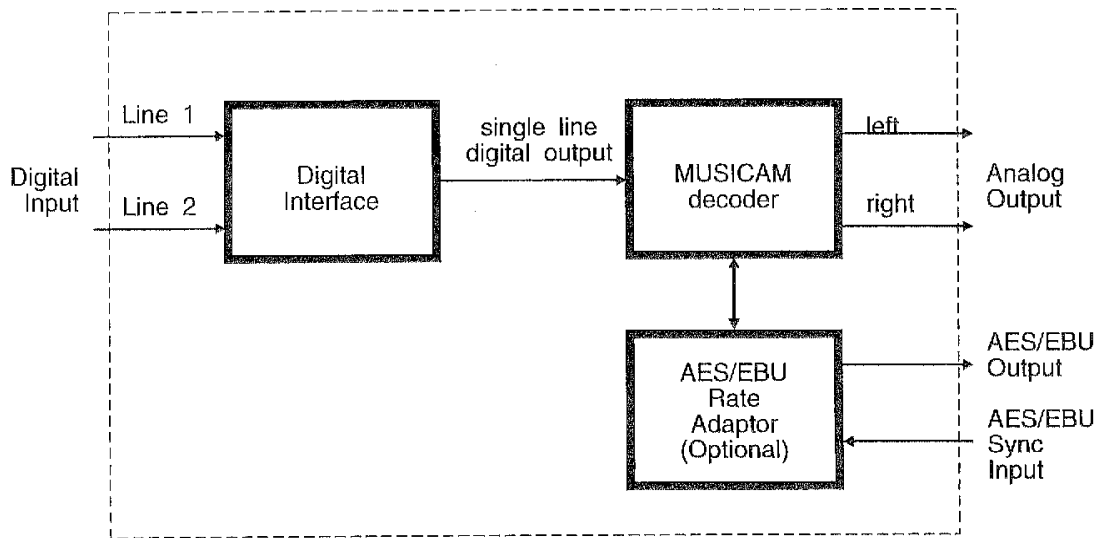


Figure 3
CDQ2000D Block Diagram

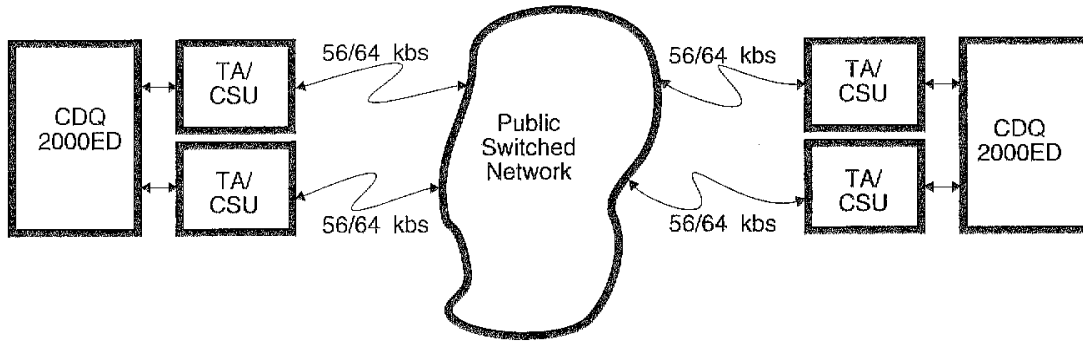


Figure 4a
 CDQ2000ED Typical interconnection via the public switched network

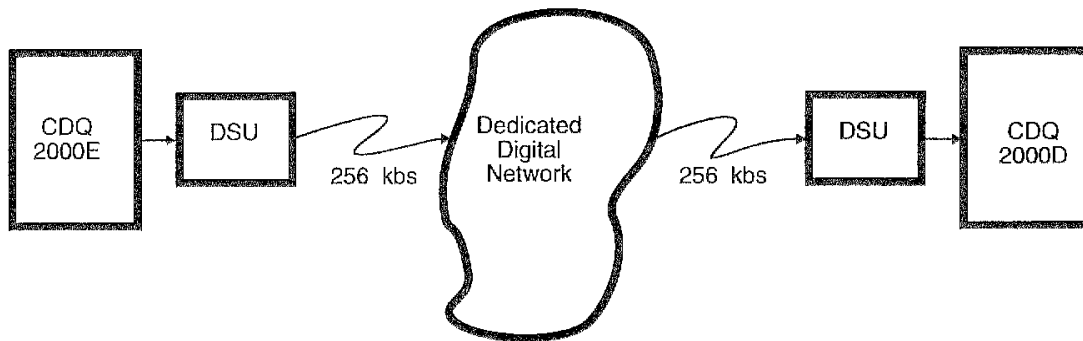


Figure 4b
 CDQ2000E/D Interconnection via a dedicated digital network

Quick Start for the CDQ2000ED

The simplest method of operation of the CDQ2000 is to set the front panel switches as shown below. No other equipment is needed except an audio system. In this configuration, the analog input signals are compressed and looped back the decoder section. Note that in the loop back mode, the decoder Line Select switches are not used. They are automatically set to match the encoder Line Select switches.

Encoder

Loopback	IN
Mode	Stereo
Type	Dual Mono
Line Select	
Line 1	▲
Line 2	▼
Rate Select	128k

Decoder

Line Select	
Line 1	▲
Line 2	▼

Encoder Dip Switches

1	Down
2	Down
3	Down
4	Down
5	Down
6	Down
7	Down
8	Down
9	Down
10	Down

Decoder Dip Switches

1	Down
2	Down
3	Down
4	Down
5	Down
6	Down
7	Down
8	Down
9	Down
10	Down

Now observe the front panel LED's. They should be as follows:

Encoder

Overload	
Left	Off
Right	Off
Mode	
Mono	Off
Stereo	On
Joint	Off
Alarms	
Line1	Off
Line2	Off
Sum	Off
PLL Locks	
Main	On
Line1	On
Line2	On

Decoder

Overload	
Left	Off
Right	Off
Mode	
Mono	Off
Stereo	On
Joint	Off
Alarms	
Line1	Off
Line2	Off
Sum	Off
PLL Locks	
Main	On
Line1	On
Line2	On

If all the switches are set and all the LED's illuminate properly, connect the audio and begin testing. Remember that the maximum level is +18 dBu and that the input impedance is set at the factory to high impedance bridging mode.

To try different digital bit rates, simply change the bit rate knob to the desired position and wait for the CDQ2000 to lock to the new rate. Similarly for mono, just change the mode switch on the encoder to mono.

Notice that at the bit rate of 128 kbs; the joint light rarely goes on. This is because there are sufficient bits to encode the audio as two separate mono channels. At 192 kbs, the joint LED flashes quite often and at 128 kbs and below, the joint light is on continuously. Setting the bit rate at 192 kbs and watching the Mode LED indicates the complexity of the audio. When the Stereo LED is illuminated, the audio is relatively simple. When the Joint LED is illuminated, the audio is more complex; the encoder must utilize a higher compression technique based on the fact that the input is a stereo signal.

Operational Modes

In all modes of operation, first select the type of digital interface. This is done by appropriate selection of the dip switches on the back. Soon after the digital line is connected, the corresponding PLL Locks LED should illuminate indicating the presence of digital clock. If the appropriate PLL Locks LED does not light, check the digital facility or the interconnecting cable. The main PLL LED should illuminate indicating that the AD (or DA) PLL has locked. After a few seconds, the summary alarm LED should be extinguished on both the encoder and the decoder indicating that all sub-systems are functioning.

Single Line Mode

This is the simplest mode of operation. Connect a digital line to the rear panel digital input connector labeled Line 1 or Line 2. Then set the front panel line 1 or line 2 Line Select switch according to the digital line. Digital audio will be transmitted over the selected line.

Single Line with Redundancy Mode

This mode is similar to the Single Line mode because digital audio is transmitted over only one line. In this mode, the CDQ2000 will select a line with a good PLL lock indication and valid MUSICAM framing. If either of these conditions are not present, the codec will alternatively search line 1 and line 2 for the correct signals. The first line to qualify, becomes the active line. This means that the this mode can work with one or two digital lines connected and the codec will select a working line.

Combined Line Mode

The Combined Line mode utilizes two 56 or 64 kbs lines which provides an aggregate bit rate of 112 or 128 kbs. If two CDQ2000ED's are used, then the system will operate with only one of the lines functional. In this case, the CDQ reverts to mono at a reduced bandwidth. In this mono mode, only the left channel is active. When both of the digital lines become active, then full bandwidth stereo operation is restored.

If a CDQ2000E is connected to a CDQ2000D, then both of the digital lines must be operational for the system to function. If either of the digital lines become inoperative, then the audio is muted.

To enter the Combined Line mode, select either 112k or 128k on the front panel bit rate selector rotary switch. The switch setting corresponds to using either two 56 or 64 kbs lines respectively. Next, set the two front panel Line Select switches to Off and observe the front panel Lines LED illuminated indicating the use of the Combined Line mode.

If a CDQ2000E/D combination is used, the encoder's front panel Mode LED's will either show Stereo or Joint independent of the number of digital lines that are actually working. The decoder will mute the audio until it decides that acceptable digital signals are present. This is indicated at the decoder by observing the Mode and the Alarms LED's. If the decoder is receiving valid digital data on both lines, then the Mode LED's will show either stereo or Joint and the Framed LED will be off.

If a two CDQ2000ED's are used, the encoder's front panel mode with either show Mono (indicating only one digital line working), Stereo or Joint (indicating both lines working) or no illuminated LED's (indicating invalid digital data on both lines).

Digital Interfaces

The CDQ2000 family provides a variety of digital interfaces. Including V.35, X.21 leased circuit and RS422. Each of these digital interfaces requires clock and data to be exchanged between the CDQ2000 and the terminal equipment. The CDQ2000 always expects the clock to be provided by the terminal equipment. The encoder section outputs data synchronized with the clock and the decoder expects the data to be synchronized with the clock. Figures 5 and 6 show the interconnection of the CDQ2000 to a generic piece of terminal equipment. The timing relationships are shown in Appendix B.

The data and clock lines are differential requiring a pair of wires for each signal. The control lines in the V.35 interface are single ended and require only one wire for each signal. The X.21 control lines are differential. The RS422 interface does not support any control lines. Any input control lines defined are ignored by the CDQ2000 and any output control lines defined are held at constant values. See Appendix A for the definition the pins used for each type of interface.

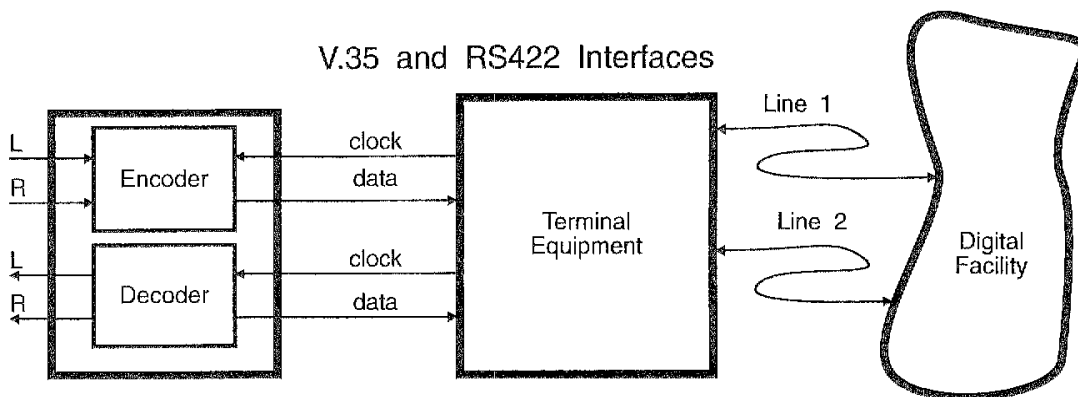


Figure 5
Basic interconnection to digital network

Each interface defines a voltage level for each of the signals. In the case of V.35 and X.21, a connector type is also defined. The connector defined in the V.35 specification is not used by the CDQ2000 because of its size. Instead, a smaller DB25 connector is used. In the case of the V.35 interface, the CDQ2000 conforms to the electrical specification but requires an adapter cable to convert the DB25 connector to the connector specified in the V.35 specification. The connector and the pin-out chosen for the V.35 interface in the CDQ2000 are a common deviant found in many systems. It is important to remember that V.35 has a separate clock for transmitted and received data. Appendix E describes the pin-out required for a DB25 to V.35 connector. The RS422 interface specification only defines the electrical voltages at the interface and leaves the

pin-out and meaning of the pins to the hardware designer. The RS449 interface specification utilizes the electrical specifications of RS422 but specifies a mechanical connector. RS449 also specifies numerous control signals besides clock and data. The CDQ2000 RS422 interface pin-out is specified in Appendix A. The RS422 interface also has a separate clock for the transmitted and received data. The CDQ2000 RS422 interface also echoes the transmitter clock. If the terminal equipment clocks the encoder data with the echoed clock, then the CDQ2000 may be located up to 4000 feet from the terminal equipment without having to worry about the encoder to clock skew.

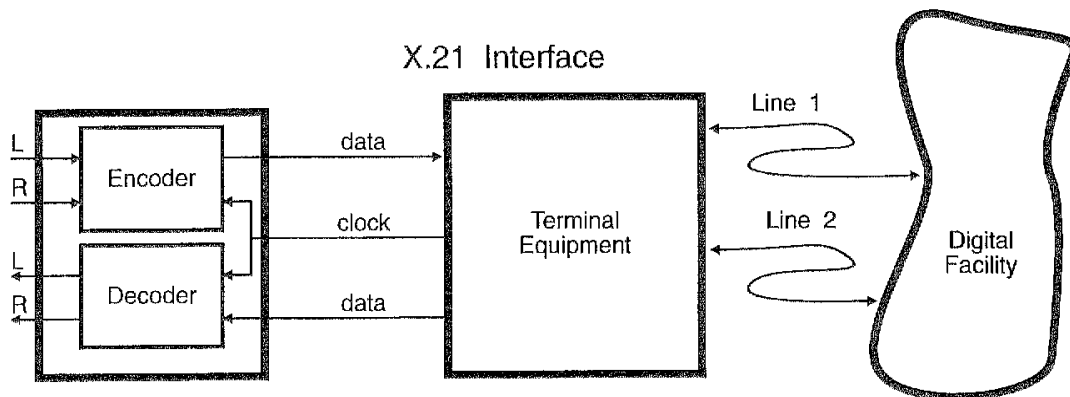


Figure 6
Basic interconnection to digital network

The X.21 interface specification is in general a very complex specification. The general specification allows a mechanism for communication between the customer equipment and the network. This communication path can be used for things such as dialing. A sub-set of the specification, called the leased circuit, restricts the interconnection to only clock and data and a very simple control signal. The mechanical connector required is the DB15 with the pin-out specified in Appendix A. The electrical specification is RS422. The X.21 interface has only one clock for both the transmit and received signals.

Since the X.21 utilizes the RS422 electrical interface, the CDQ2000 can use the same connector for both interfaces. In the case of the X.21 interface, the single clock is used internally for both the transmit and received timing. The selection of the type of digital interface is governed by rear panel dip switches. See Appendices C and D for the appropriate settings.

Operating the CDQ2000 Encoder and Decoder Separately

The CDQ2000ED contains both an encoder and a decoder. These two units interact in several ways. It is possible by the selection of Decoder Dip Switch 6 to force the decoder to operate as a separate unit. See Appendix D for the setting of this dip switch.

When the dip switch is set so that the encoder and decoder co-operate, the Line Select switches on the decoder are inoperable when the encoder Line select switches are both off and the data rate is 112 or 128 kbs. This is the combined line mode and is indicated by the illumination of the Lines Two LED.

Differences between the CDQ2000ED and the CDQ2000E/D

The CDQ2000ED contains an encoder and a decoder while the CDQ2000E contains the encoder portion only. The CDQ2000D contains only a decoder. The following list differences between a system that contains a CDQ2000ED at both ends of the transmission facility (see Figure 4a), a system that has a CDQ2000E at one end and a CDQ2000D at the other end (see Figure 4b).

1. CDQ2000ED system can simultaneously transmit audio in both directions.
2. The CDQ2000ED system reverts to a reduced bandwidth mono system if operating in the Combined Line mode and one of the lines fails. The CDQ2000E/D system mutes in the same situation.
3. The CDQ2000ED can operate in loopback for system check out. The CDQ2000E/D system requires an external customer supplied unit for loopback operation.
4. The CDQ2000ED system encodes and decodes at the same bit rate. A system with a separate CDQ2000E and CDQ2000D could operate at different bit rates for encoding and decoding.

Operation of the CDQ2000E and CDQ2000D

The CDQ2000E encoder and the CDQ2000D decoder units require the presence of an external clock. This clock must be provided by an external source such as a CSU/DSU.

See Figure 7 for the interconnection of a CDQ2000E and a CDQ2000D in a loop back configuration.

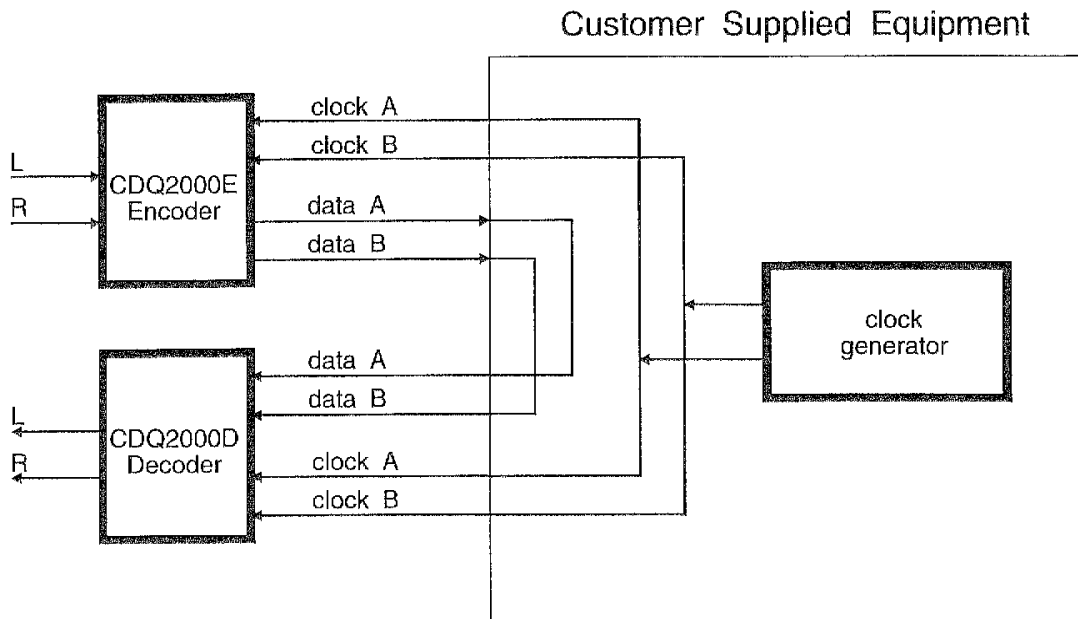


Figure 7
CDQ2000E/D Loop Back Interconnection

Ancillary Data Port

The CDQ2000 provides for transmission of asynchronous data via a RS-232 interface. This interface provides a transparent channel for the transmission of 8 data bits. The data format is 1 start bit, 8 data bits, 1 stop bit and no parity bits. This interface is capable of transmitting at the maximum data rate selected by the encoder and decoder data rate dip switches and thus no data pacing such as XON/XOFF or CTS/RTS is provided. Appendix C describes the encoder and decoder dip switches.

The encoder RS-232 data rate can be set from 300 to 19,200 bps. The use of the ancillary data channel decreases the number of bits available to the audio channel. The reduction of the audio bits only occurs if ancillary data is actually present. The data rate can be thought of as a maximum data rate and if there is no ancillary data present, then no data bits are transmitted. A typical example of this situation occurs when the CDQ2000 encoder is connected to a terminal; when the user types a character the character is sent to the decoder at the bit rate specified.

The setting of the decoder baud rate selection dip switches must be done considering the setting of the encoder. The decoder dip switches must be an equal or higher baud rate relative to the encoder. For example, it is possible to set the decoder ancillary baud rate to 9,600 baud. In this case, the encoder baud rate may be set to any value from 300 to 9,600 but not 19,200. If the decoder baud rate is set to a higher rate than the encoder, the data will burst out at the decoder's baud rate. The maximum sustained baud rate is controlled by the encoder.

The algorithm for the transmission of ancillary data is for the encoder to look during each 24 millisecond MUSICAM frame interval and see if any ancillary data is in its input buffer. If there are characters in the encoder's input buffer, then the maximum number of characters consistent with the selected baud rate are sent. During a 24 millisecond period, the table below shows the maximum number of characters sent for each baud rate.

Bit Rate	Number of Characters
300	1
1200	3
2400	6
3600	9
4800	12
7200	18
9600	24
19200	47

The CDQ2000 provides no error detection or correction for the ancillary data. The user assumes the responsibility for the error control strategy of this data. For example, at an error rate of 10^{-5}

(which is relatively high) and an ancillary data rate of 1200 baud, 1 out of every 83 characters will be received in error. Standard computer data communication protocol techniques can be used to maintain data integrity.

When designing an error protection strategy, it must be remembered that the CDQ2000 may occasionally repeat the last 24 milliseconds of audio under certain error conditions. The effect on the audio is nearly imperceptible. However, the ancillary data is not repeated.

Digital Audio Input/Output (Optional)

The CDQ2000 provides for direct digital audio input and output via AES/EBU interfaces. The professional digital audio data rate of 48k is used by the CDQ2000. If the consumer rate of 44.1k is desired, an external rate converter must be used.

The encoder receives direct digital input via the XLR connector on the rear panel. Analog or digital (but not both simultaneously) signals may be input to the CDQ2000 as selected by the front panel switch. If the digital input is selected, the CDQ2000 locks to the incoming AES/EBU input and displays the lock condition via a front panel LED. If digital audio input is selected, the AES PLL lock light must be illuminated before audio is accepted for encoding. In normal operation, the CDQ2000 locks its internal clocks to the clock of the telephone network. For loopback, it locks its clocks to an internal clock. In either case, the clock used by the CDQ2000 is not at precisely the same frequency as the AES/EBU input. To prevent slips from occurring due the presence of two master clocks, a rate synchronizer is built into the encoder section to perform the necessary rate conversion between the two clocks.

The decoder outputs direct digital signals via the rear panel XLR connector. Additionally, the decoder may be synchronized to an external clock by an additional XLR connector (SYNC) on the rear panel. If no input is present on the decoder AES/EBU SYNC input line, then the output AES/EBU digital audio is generated by the internal clock source that is either at the telephone or internal clock rate. If the SYNC input is present, then the digital audio output is generated at the frequency of the SYNC input. The presence of a valid sync source is indicated by the illumination of the front panel AES PLL LED. The sync frequency may be slightly different from that of the CDQ2000 clock source and again rate synchronism is performed to prevent any undesired slips in the digital audio output. The SYNC input is assumed to be an AES/EBU signal with or without data present. The CDQ2000 only uses the framing for the frequency and sync determination.

Encoder Front Panel Description

Overload

Left - The illumination of this LED indicates that the peak analog input of the left channel is within 3dB of the maximum input level.

Right - The illumination of this LED indicates that the peak analog input of the right channel is within 3dB of the maximum input level.

Loopback (CDQ2000ED only)

Depressing this switch causes the output of the encoder section to be directed to the decoder. This tests the entire system with the exception of the final digital line drivers and receivers.

The Line Select switches on the decoder are set to match those of the encoder when the unit is in loopback. Thus the setting of the decoder Line Select switch has no effect when the unit is in loopback.

Mode

- | | |
|--------|---|
| Mono | This LED indicates that the codec is operating in the mono mode. |
| Stereo | This LED indicates that the codec is operating in the stereo mode. |
| Joint | This LED indicates that the codec is operating in the stereo mode with advanced compression techniques to enable more compression than is normally available. |

Alarms

- | | |
|--------|--|
| Line 1 | This LED indicates a malfunction on the input side of digital line 1. |
| Line 2 | This LED indicates a malfunction on the input side of digital line 2. |
| Sum | This LED indicates a summary alarm. It illuminates if any relevant alarm is active. The alarm relay closes any time the Sum LED is illuminated. The summary alarm LED illuminates for a minimum of .024 seconds. |

PLL Locks

Main	This LED is illuminated when the main AD phase lock loop is functioning properly.
Line 1	This LED is illuminated when phase lock is achieved on digital input line 1.
Line 2	This LED is illuminated when phase lock is achieved on digital input line 2.

PLL

AES	This LED is illuminated when the AES/EBU receiver has locked to a digital input signal.
-----	---

Mode

Mono	Placing the switch in this position, directs the codec to operate in Mono. In this mode the audio source is the left channel and the right channel is ignored. (See Appendix G, Decoder switches 8 and 9.)
Stereo	Placing the switch in this position, directs the codec to operate in either the of the two stereo modes - Dual Mono or Joint Stereo. In the Dual Mono mode, both of the input analog channels are handled independently. In the Joint Stereo mode, further compression may be used based on the assumption that the input signal is stereo and there exists additional redundancy between the left and the right channels. See the section discussing dual mono and joint stereo compression modes.

Input

Digital	In this position, the input to the encoder is via the AES/EBU input. The AES PLL lock light must be illuminated for this input to function.
Analog	In this position, the input to the encoder is via the two XLR connectors on the rear panel of the encoder.

Type

- Dual Mono Encodes the left and right channels totally independently. This allows the transmission of audio on two totally independent left and right channels.
- Joint Stereo Utilizes the fact that the left and right channels contain a true stereo signal to achieve additional compression.

Line Select

Line 1 & Line2

These two switches are used to control the use of the digital lines. The switch settings should match the switch settings of the associated decoder.

Line1	Line2	
▲	▲	Single line with redundancy
▲	▼	Single line mode - Line 1
▼	▲	Single line mode - Line 2
▼	▼	Combined line

Single line with redundancy Mode

In this mode the CDQ2000 examines both lines looking for a clock signal at the rate selected. If a valid clock is found on either line 1 or line 2, then the line is considered active. The status of line 1 and line 2 can be seen by examining the front panel PLL Locks LED group. The line 1 and line 2 lock indicators show the status of the lines. The CDQ2000 will then select an active line with a valid clock and use that line until the clock becomes invalid. Line 1 has priority over line 2 if both lines are valid. At that point, it switches to the other line if its clock is valid.

Single Line Mode

In this mode, the CDQ2000 directs the digital output to the selected line without regard to the validity of the status of the line.

Combined Line Mode (valid for 112/128kbs only)

The CDQ2000 Codec is designed to take advantage of the economy provided by the terrestrial switched digital circuits by permitting the user to combine a pair of switched 56kbs or ISDN 64kbs circuits for high fidelity stereo transmission.

This mode instructs the CDQ2000 to use both of the lines if the data rate is 112 kbs or 128 kbs and valid clock is received on both lines. If this is the case, the front panel LED labeled Lines - two will illuminate confirming that the combined mode has been selected. The combined mode splits the audio over lines 1 and 2 and synchronizes both lines at the receiver by accounting for the delay between the two lines.

Bit Rate Selector

This rotary switch selects the bit rates used by the encoding process. If the Combined Line mode is used, it represents the sum of the bit rates of the two lines. The allowed bit rates are:

56 kbs
64 kbs
96 kbs
112 kbs
128 kbs
192 kbs
256 kbs
384 kbs

Decoder Front Panel Description

Overload

See corresponding encoder front panel description of these LED's.

Mode

See corresponding encoder front panel description of these LED's.

Alarms

- | | |
|--------|--|
| Framed | This LED is illuminated if proper framing of the received digital bit stream has not been achieved. |
| Ber | This LED is illuminated if a bit error has been detected by the receiver. See the section discussing error protection. |
| Line 1 | See corresponding encoder front panel description. |
| Line 2 | See corresponding encoder front panel description. |
| Sum | See corresponding encoder front panel description. |

PLL Locks

See corresponding encoder front panel description.

PLL

- | | |
|-----|--|
| AES | This LED is illuminated when the Decoder senses the sync AES/EBU input for the decoder. The decoder always outputs via the AES/EBU output, but it will synchronize with an external AES/EBU input. |
|-----|--|

Line Select

See corresponding encoder front panel description.

Power

This LED indicates the presence of power to the codec.

Encoder Rear Panel Description

Power

This connector is used to connect the CDQ2000 via an IEC power cable. The voltage required by the CDQ2000 can be from 85 to 265 volts AC at 50 to 60 Hz.

Audio Input

The Left and Right XLR female connectors accept analog audio input at a peak level of +18 dBu. An internal jumper can be set for 600 ohm or 20,000 ohm input impedance. See Appendix A for the pin-out of this connector.

AES/EBU Input (present only if AES/EBU option ordered)

This XLR female connector accepts 48k AES/EBU digital audio input. The CDQ2000 will illuminate the AES PLL Lock LED on the encoder section if this signal is present. The front panel input switch must be set for digital to utilize this input.

Alarm

This connector presents the alarm relay contacts. This relay closes whenever the front panel Sum Alarm LED is illuminated. See Appendix A for the pin-out of this connector.

RS232

This connector is used for inputting RS232 data for transmission to the decoder. See Appendix A for the pin-out of this connector.

X.21/RS422

The line 1 and line 2 connectors are used for connection to the digital terminal equipment. See Appendix A for the pin-out of this connector. Also, see Appendix C for the dip switch settings to enable this interface.

V.35

The line 1 and line 2 connectors are used for connection to the digital terminal equipment. See Appendix A for the pin-out of this connector. Also, see Appendix C for the dip switch settings to enable this interface.

Dip Switches

These switches control various options of the encoder. See Appendix C for the dip switch definitions.

Decoder Rear Panel Description

Power (Not present on the CDQ2000ED)

This connector is used to connect the CDQ2000 via an IEC power cable. The voltage required by the CDQ2000 can be from 85 to 265 volts AC at 50 to 60 Hz.

Audio Output

The Left and Right XLR male connectors output analog audio at a peak level of +18 dBu. The output impedance is 60 ohms. See Appendix A for the pin-out of this connector.

AES/EBU Output (present only if AES/EBU option ordered)

The AES/EBU digital audio signal is output on this XLR male connector at a sampling rate of 48k. The output is transformer isolated.

AES/EBU Sync (present only if AES/EBU option ordered)

This XLR female connector is used to input an AES/EBU signal. If this signal is present, the AES/EBU output is frame synchronized to it. This input is transformer isolated and expects an AES/EBU signal at 48k. This signal may or may not carry any valid data. It must contain valid AES/EBU framing. If a valid signal is present, the decoder AES PLL lock LED will illuminate indicating that the AES/EBU output is synchronized to this line.

Alarm

This connector presents the alarm relay contacts. This relay closes whenever the front panel Sum Alarm LED is illuminated. See Appendix A for the pin-out of this connector.

RS232

This connector is used for inputting RS232 data for transmission to the decoder. See Appendix A for the pin-out of this connector.

X.21/RS422 (Only present on the CDQ2000D)

See the encoder for a description of these connectors.

V.35

See the encoder for a description of these connectors.

Dip Switches

These switches control various options of the encoder. See Appendix D for the dip switch definitions.

Appendix A

CONNECTORS

V.35 Connector: (DB25 Connector)

- | | |
|----|--|
| 1 | Protective ground |
| 7 | Signal ground |
| 3 | Received data A-wire (input) |
| 16 | Received data B-wire (input) |
| 17 | Receiver signal element timing A-wire (input) |
| 12 | Receiver signal element timing B-wire (input) |
| 15 | Transmitter signal element timing A-wire (input) |
| 14 | Transmitter signal element timing B-wire (input) |
| 2 | Transmitted data A-wire (output) |
| 13 | Transmitted data B-wire (output) |
| 20 | Data terminal ready (output)* |
| 4 | Request to send (output) * |
| 8 | Received line signal detector (input) * |

Connector: DB25S (female)

* Unused by CDQ2000

X.21 Connector:

1	fg	frame ground
2	t(a)	Transmitted data A-wire (output)
3	c(a)	Control element A-wire (output) **
4	r(a)	Receiver data A-wire (input)
5	i(a)	Indicator element A-wire (input) *
6	s(a)	Timing element A-wire (input)
7	b(a)	Byte timing A-wire (input) *
8	sg	signal ground
9	t(b)	Transmitted data B-wire (output)
10	c(b)	Control element B-wire (output) **
11	r(b)	Receiver data B-wire (input)
12	i(b)	Indicator element B-wire (input) *
13	s(b)	Timing element B-wire (input) *
14	b(b)	Byte timing B-wire (input)
15		No connection

Connector: DB15S (female)

* unused by CDQ2000

** B > A always

RS422 Connector:

1	fg	frame ground
2	t(a)	Transmitted data A-wire (output)
3	c(a)	Transmit timing echo A-wire (output)
4	r(a)	Receiver data A-wire (input)
5	i(a)	Indicator element A-wire (input) *
6	s(a)	Receiver timing element A-wire (input)
7	b(a)	Transmit timing A-wire (input)
8	sg	signal ground
9	t(b)	Transmitted data B-wire (output)
10	c(b)	Transmit timing echo B-wire (output)
11	r(b)	Receiver data B-wire (input)
12	i(b)	Indicator element B-wire (input) *
13	s(b)	Receiver timing element B-wire (input)
14	b(b)	Transmit timing B-wire (input)
15		No connection

Connector: DB15S (female)

* unused by CDQ2000

RS422 is identical to X.21 except that there are separate clocks for the receiver and the transmitter. Also the RS422 echoes the transmitter clock.

Alarm Connector - Encoder and Decoder:

The Alarm connector consists of a DPDT relay. The relay contacts are defined below.

- | | |
|---|-------------------|
| 1 | Common 1 |
| 2 | Not used |
| 3 | Normally closed 2 |
| 4 | Common 2 |
| 5 | Normally open 1 |
| 6 | Not Used |
| 7 | Normally open 2 |
| 8 | Not used |
| 9 | Normally closed 1 |

Connector: DB9P (male)

Relay: Takamisawa RZ-5-C

This relay closes whenever the summary alarm LED for the corresponding encoder or decoder illuminates.

RS232 Interface Connector - Encoder and Decoder:

1	Not used
2	Transmitted data (output) *
3	Received data (input) **
4	Not used
5	Signal ground
6	Not used
7	Not used
8	Not used
9	Not used

Connector: DB9P (male)

* The decoder section outputs data on this pin. The received data pin (pin 3) is not used by the decoder.

** The encoder section receives data on this pin. The transmitted data pin (pin 2) is not used by the encoder.

The RS232 data consists of 1 start bit, 8 data bits and 1 stop bit. No parity bit is included. The CDQ2000 provides a transparent channel for transmitting 8 bit bytes of information.

Remote Connector:

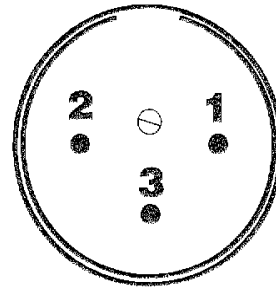
- 1 Remote Enable
- 2 Select A
- 3 Select B
- 4 Select C
- 5 Ground
- 6 Line 1
- 7 Line 2
- 8 Mode
- 9 Type

The Remote Connector allows front panel switches to be controlled remotely. There are eight control lines and one ground line. The control lines are pulled up by 10k resistors to +5v. Grounding a particular line has the same effect as throwing the associated front panel switch up, provided that the remote enable line is grounded. If the remote enable line is not grounded, the CDQ2000 ignores the rest of the control lines.

Bit Rate	Select A	Select B	Select C
56k	Grounded	Grounded	Grounded
64k	Open	Grounded	Grounded
96k	Grounded	Open	Grounded
112k	Open	Open	Grounded
128k	Grounded	Grounded	Open
192k	Open	Grounded	Open
256k	Grounded	Open	Open
384k	Open	Open	Open

Audio Input Connector:

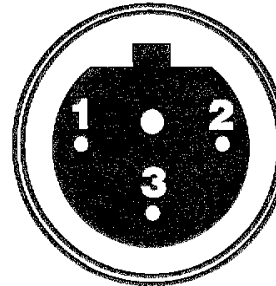
- 1 Ground
- 2 + input
- 3 - input



Connector: XLR female

Audio Output Connector:

- 1 Ground
- 2 + output
- 3 - output

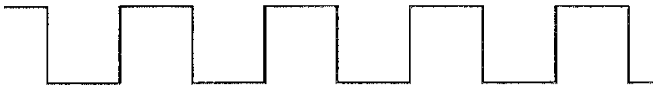


Connector: XLR male

Appendix B

Digital Audio Timing

Transmit Timing B



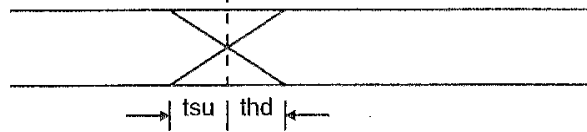
Transmit Data



Transmit Timing B



Transmit Data



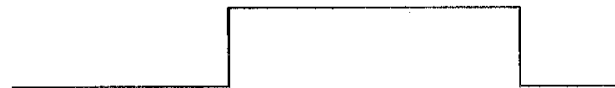
Receive Timing B



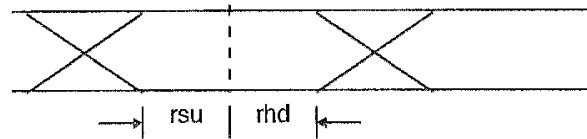
Receive Data



Receive Timing B



Receive Data



$rsu = rhd = tsu = thd = 1 \text{ us}$

Appendix C

CDQ2000ED

Encoder Dip Switches

1	Ancillary data baud rate b2				
2	Ancillary data baud rate b1				
3	Ancillary data baud rate b0				
		b2	b1	b0	Rate
		<u>Dn</u>	<u>Dn</u>	<u>Dn</u>	<u>300</u>
		Dn	Dn	Up	1200
		Dn	Up	Dn	2400
		Dn	Up	Up	3600
		Up	Dn	Dn	4800
		Up	Dn	Up	7200
		Up	Up	Dn	9600
		Up	Up	Up	19200
4	Up	ISO Checksum			
	Dn	CCS Checksum (pre hifa)			
5	Digital interface selection b1				
6	Digital interface selection b0				
		<u>b1</u>	<u>b0</u>	<u>I/F type</u>	
		Dn	Dn	V.35	
		Dn	Up	X.21	
		Up	Dn	Not defined	
		Up	Up	RS422	

Loopback sources select (CDQ2000ED only)

7	Up	AES/EBU timing used for loopback source
	Dn	Internal crystal used for loopback source

CDQ2000ED

8		Analog bandwidth b0		
9		Analog bandwidth b1		
		<u>b1</u>	<u>b0</u>	<u>Bandwidth</u>
		Dn	Dn	default for bit rate
		Dn	Up	10,500Hz
		Up	Dn	12,750Hz
		Up	Up	15,000
10	Up	Internal clock output enabled if RS422.		
	Dn	Internal clock output disabled (xmit timing echoed if RS422).		

See the attached figure for the schematic description of encoder dip switch 10.

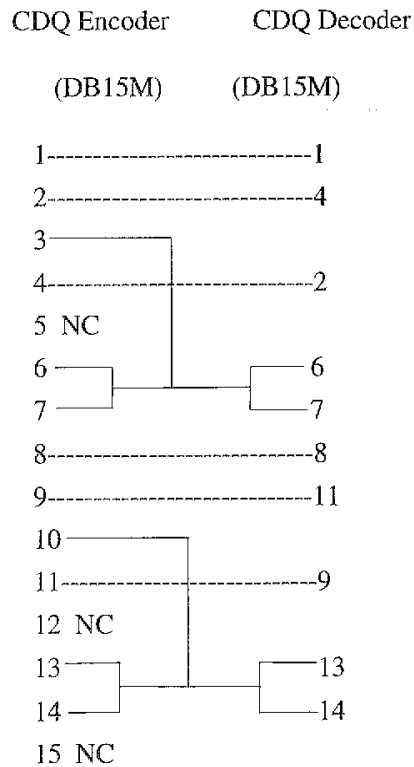
When the internal clock output is enabled, a clock is sent out to the control element output on the RS422 connector. This is an internally generated clock at the selected bit rate and is derived from a crystal oscillator. This clock signal is not used internally for any timing purposes. It is output on the RS422 line connector on pins 3 and 10. It is intended to be connected to pins 7 and 14 of the encoder and to the clock input pins of a separate decoder. Utilizing this internal clock allows the encoder to be connected to a decoder without the need of an external clock source. The CDQ encoder can provide a clock source for itself and for an associated decoder.

If the internal clock is disabled and RS422 is selected, then the transmit clock is echoed on pins 3 and 10. This feature allows precise timing if long (4000 foot) cables are used to provide the clock to the encoder. In this case, the data and the echo clock arrive at the end of the cable at precisely the same time.

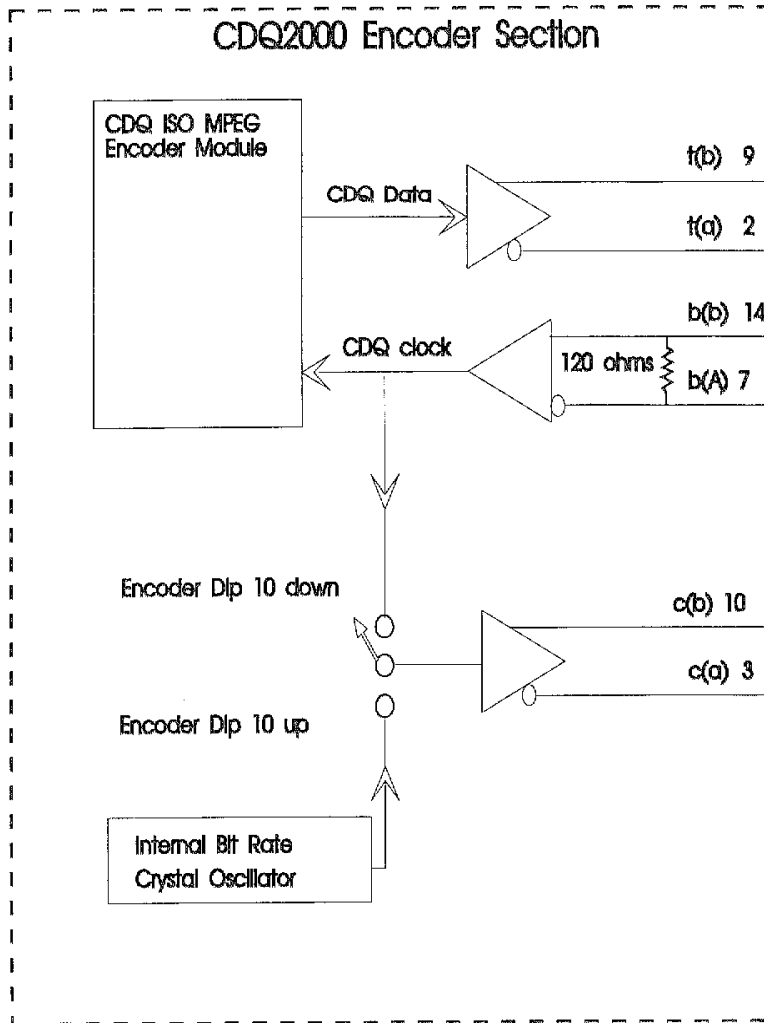
Cable connections for connecting a CDQ encoder to CDQ decoder.

Encoder dip switch 5,6 & 10 must be in the Up position (RS422 selected). On the decoder, dip switch 5 (RS422 selected) must be Up. The cable must be connected to the RS422 line 1 (line 2) on both the encoder and the decoder. Also note which cable end is wired for encoder or decoder (They are not reversible).

The Line 1 switch must be Up and the Line 2 switch must be Down. This selects line 1 in both the encoder and the decoder to carry the data and clock signals. Alternatively, line 2 can be utilized.



Note that the transmit timing echo line is connected to the transmit timing input. This allows the CDQ encoder to receive a clock signal.



RS422 Line
Connector

Encoder Dip Switch 10 Description

Decoder Dip Switches

- 1 Diagnostic/ baud rate control b2
- 2 Diagnostic/ baud rate control b1
- 3 Diagnostic/ baud rate control b0

Baud rate control (dip switch 4 down)

b2	b1	b0	Rate
Dn	Dn	Dn	300
Dn	Dn	Up	1200
Dn	Up	Dn	2400
Dn	Up	Up	3600
Up	Dn	Dn	4800
Up	Dn	Up	7200
Up	Up	Dn	9600
Up	Up	Up	19200

Diagnostic control (dip switch 4 up)

b2	b1	b0	Description
Dn	Dn	Dn	Not Used
Dn	Dn	Up	1000 Hz tone left channel only
Dn	Up	Dn	1000 Hz tone right channel only
Dn	Up	Up	1000 Hz tone right and left channel
Up	Dn	Dn	Decoder memory test
Up	Dn	Up	9600 Hz tone left channel only
Up	Up	Dn	9600 Hz tone right channel only
Up	Up	Up	9600 Hz tone right and left channel

- 4 Up Goes into the diagnostic mode. The diagnostic is controlled by decoder dip switches 1, 2 and 3.
- Dn Normal decoder operation. Dip switches 1, 2 and 3 control the baud rate of the ancillary RS-232 data.
- 5 Not used

- 6 Up Decoder functions separately from encoder
- Dn Decoder functions in co-operation with encoder. This is necessary for the Combined Line mode to automatically operate with one or two lines.

7 Mute output audio

8 Mono output control b0

9 Mono output control b1

b1	b0	Description
Dn	Dn	Mono is output on both channels
Dn	Up	Mono is output on left channel only
Up	Dn	Mono is output on both channels
Up	Up	Mono is output on right channel only

When the CDQ2000 is in mono mode, only the audio present on the left channel of the encoder is sent to the decoder. The above 2 dip switches describe which output channel is used for the mono output.

10 Not used

Appendix D

CDQ2000E

Encoder Dip Switches

See Appendix C encoder dip switch settings.

CDQ2000D

Decoder Dip Switches

- 1 See CDQ2000ED decoder dip switch 1
- 2 See CDQ2000ED decoder dip switch 2
- 3 See CDQ2000ED decoder dip switch 3
- 4 See CDQ2000ED decoder dip switch 4

- 5 Digital interface selection:
Dn V.35
Up X.21/ RS422

- 6 Not Used
- 7 Not used
- 8 See CDQ2000ED decoder dip switch 8
- 9 See CDQ2000ED decoder dip switch 9
- 10 Not used

Appendix E

DB25 to V.35 Adapter Cable

(CCS 601650-L1)

V.35	Name	DB25
R	RDA	3
T	RDB	16
V	RETA	17
X	RETB	12
Y	TETA	15
AA	TETB	14
P	TDA	2
S	TDB	13
C	RTS	4
A	FG	1
B	SG	7 connect to shield at 1 end
H	DTR	20
F	RLSD	8
J	RI	21

Connectors: DB25 - DB25P (male)

V.35 - AMP 200517-2 Block & AMP 201357-1

Paired pins should be connected to paired wires. Unpaired wires may be connected without regard to pairing.

Appendix F

Analog Bandwidth vs. Bit Rate

The analog bandwidth of the CDQ2000 is set by the following rear panel dip switches on the encoder.

Bit Rate	Mode	Analog BW	Analog BW	Analog BW	Analog BW
		Dip 9 8	Dip 9 8	Dip 9 8	Dip 9 8
		0 0	0 1	1 0	1 1
56k	Mono	8.25k	6.00k	6.75k	7.50k
64k	Mono	8.25k	6.00k	6.75k	7.50k
96k	Mono	20.25k	10.50k	12.75k	15.00k
112k	Mono	20.25k	10.50k	12.75k	15.00k
128k	Mono	20.25k	10.50k	12.75k	15.00k
192k	Mono	20.25k	10.50k	12.75k	15.00k
256k	Mono	20.25k	10.50k	12.75k	15.00k
384k	Mono	20.25k	10.50k	12.75k	15.00k
56k	Dual Mono	5.25k	3.00k	3.75k	4.50k
64k	Dual Mono	5.25k	3.00k	3.75k	4.50k
96k	Dual Mono	10.50k	6.75k	7.50k	9.00k
112k	Dual Mono	10.50k	6.75k	7.50k	9.00k
128k	Dual Mono	10.50k	6.75k	7.50k	9.00k
192k	Dual Mono	20.25k	10.50k	12.75k	15.00k
256k	Dual Mono	20.25k	10.50k	12.75k	15.00k
384k	Dual Mono	20.25k	10.50k	12.75k	15.00k
56k	Joint Stereo	6.00k	3.75k	4.50k	7.50k
64k	Joint Stereo	6.00k	3.75k	4.50k	7.50k
96k	Joint Stereo	12.00k	7.50k	9.00k	10.50k
112k	Joint Stereo	20.25k	10.50k	12.75k	15.00k
128k	Joint Stereo	20.25k	10.50k	12.75k	15.00k
192k	Joint Stereo	20.25k	10.50k	12.75k	15.00k
256k	Joint Stereo	20.25k	10.50k	12.75k	15.00k
384k	Joint Stereo	20.25k	10.50k	12.75k	15.00k

These default analog bandwidths may be over-ridden by setting the proper encoder dip switches.

Appendix G

Data Port Timing Considerations

In the standard configuration the CDQ2000 derives its timing from the digital data port. A clock must be provided that is ± 200 ppm from the selected data rate. In the dual line mode, the clocks from each line must be exactly the same frequency, but can have an arbitrary constant phase difference. In the X.21 mode, the transmitter and receiver use a common input clock. RS422 has independent receive and transmit clock inputs.

The transmit clock comes in on the byte timing line and is re-transmitted on the CON line.

It is possible in both the X.21 and RS422 modes to have the CDQ encoder generate a master clock. By switching up dip switch 10 on the encoder, a crystal generated clock of the selected data rate will be sent out on the CON control leads. This clock could then be fed back into the clock input line(s) to sync the unit to it.

By placing a dual unit in Loop Back with switch 10 up, the encoder will lock to the internal clock and transmit data out the transmit data line. The internal clock will come out the CON control lines and the decoder will frame to the outgoing data. This allows the encoder data to be verified as it is transmitted by the decoder.

Appendix H

CDQ2000 Specifications

Audio Path

Frequency Response	20Hz-20kHz 20Hz-20kHz \pm 3dB down 3dB at 20kHz
THD	0.01% at 1kHz
Signal to Noise	88dB referred to +18dBu*
Crosstalk	Less than -85dB
Max input/output levels	+18dBu*
Nominal input/output levels	+8dBu*
Output Impedance	60 ohms balanced(active)
Input Impedance	600 ohm balanced or 20ohms bridged
AD converter resolution	16 bit sigma-delta
DA converter resolution	18 bit sigma-delta
Analog input connector	Female XLR
Analog output connector	Male XLR
Overload indication	+14dBu*
Analog sample rate	48kHz

Digital Path

Data rate (k bit/sec)	56, 64,96, 112, 128, 192, 256, 384
Coding	MPEG ISO mode 2 and 2A
Ancillary data path	9600, 4800...300, none
Digital interface	V.35, X.21(leased), RS422A
Digital access	Single, combined, redundant
Status output	Form C summary alarm
Digital clock accuracy	200ppm

*dBu = dBm into a 600 Ω load

Physical

Size	19in x 1.72in x 10in (E,D) 19in x 3.47in x 10in (ED)
Enclosure	Aluminum
Displays	LED
Power	90-250volts
Configurations	Encoder, Decoder, Encoder/Decoder