DXR.1 Digital Audio Codec

SECTION

About Comrex

Comrex has been building reliable, high quality broadcast equipment since 1961. Our products are used daily in every part of the world by networks, stations and program producers.

Every product we manufacture has been carefully designed to function flawlessly, under the harshest conditions, over many years of use. Each unit we ship has been individually and thoroughly tested. Most items are available off-the-shelf, either directly from Comrex or from our stocking dealers.

Comrex stands behind its products. We promise that if you call us for technical assistance, you will talk directly with someone who knows about the equipment and will do everything possible to help you.

Our toll free number in North America is 800 237-1776. Product information, along with Engineering Notes and User Reports, is available through our website at www.comrex.com. Our email address is info@comrex.com.

Warranty and Disclaimer

All equipment manufactured by Comrex Corporation is warranted by Comrex against defects in material and workmanship for one year from the date of original purchase, as verified by the return of the Warranty Registration Card. During the warranty period, we will repair or, at our option, replace at no charge a product that proves to be defective, provided you obtain return authorization from Comrex and return the product, shipping prepaid, to Comrex Corporation, 19 Pine Road, Devens, MA 01434 USA. For return authorization, contact Comrex at 978 784-1776 or fax 978 784-1717.

This Warranty does not apply if the product has been damaged by accident or misuse or as the result of service or modification performed by anyone other than Comrex Corporation.

With the exception of the warranties set forth above, Comrex Corporation makes no other warranties, expressed or implied or statutory, including but not limited to warranties of merchantability and fitness for a particular purpose, which are hereby expressly disclaimed. In no event shall Comrex Corporation have any liability for indirect, consequential or punitive damages resulting from the use of this product.

SECTION 1. INTRODUCTION

The DXR.1 is a digital audio compression device designed for broadcasters. It employs the ITU G.722 international standard to send and receive 7.5 kHz audio at transmission rates of 56/64 kbps. This unit also provides a full 15 kHz bidirectional feed on 112/128 kbps circuit.

DIGITAL SERVICES

The DXR.1 works on any synchronous 56, 64, 112 or 128 kbps digital transmission system, including ISDN, Switched 56, Digital Data Service (DDS), Fractional T1, digital satellite links and wireless modems.

For part-time audio feeds, dial-up Basic Rate Installation (BRI) ISDN or Switched 56 (SW56) digital lines are ideal since they are billed at a low monthly cost, plus per minute usage which is not much more than standard analog lines. BRI ISDN is available throughout the world. SW56 is a North American “precursor” to ISDN and both services work with one another.

Dedicated 56 kbps or 64 kbps telephone lines (point to point) have been available for quite some time. They may provide a significant cost savings for applications that require full time or near full time exchange of audio. In North America, this is known as Digital Data Service (DDS). Sometimes, Fractional T1 (or E1) is available. This service provides a dedicated link with higher data rates. Portable satellite earth stations and wireless modems may also be used with the DXR.1. Comrex has an extensive engineering note library describing these applications. Contact us for further details.

Note: External terminal equipment is necessary to connect the DXR.1 with the data channel. Please refer to page 8 for more information on terminal equipment.

WHAT COMES WITH THE DXR.1?

The following items are shipped with a new DXR.1:

(1) A/C power cord
(1) DB25 male to DB25 female cable
(1) Operating manual
(1) Warranty card (Please fill out and return.)

Section 2. Setup

Data Connection

This connection is made between the Terminal Adapter or CSU/DSU and the DXR.1 back panel. This is done through the Data in/out connector. The DB25 (25 pin) connector using the EIA530 protocol is the standard data port. Assuming your terminal equipment uses this same connector, you will need a 25 pin straight-through cable (provided by Comrex) to connect to this port. The pin-out of the DB25 connector is:

Pin #	Function
1	Shield
2	TX Data A
3	RX Data A
7	Ground
9	RX Clock B
12	TX Clock B
14	TX Data B
15	TX Clock A
16	RX Data B
17	RX Clock A
20	DTR A
23	DTR B

24 is reserved. Do not connect to it.

If your terminal equipment uses a protocol other than EIA530, a converter cable will be required. Optional adapter cables for V.35 and X.21 connections are available through Comrex. If you want to construct your own adapter cables, the pin-outs are on page 6.

Note: Simply because terminal equipment uses a 25 pin D connector does not mean that it uses EIA530. Implementations of this sort vary widely. Check that the equipment specifically mentions EIA530 or RS530 as the protocol for its data port. Otherwise, it will not work.

Power Connection

The power connection is on the back panel of the DXR.1. The unit is switch selectable for either 115 VAC/60 Hz or 230 VAC/50 Hz. The switch is located on the inside of the unit, on the printed circuit board and is shipped from the factory in the 115VAC/60Hz position.

To change the setting, first make sure the unit is unplugged. Open the DXR.1 case by removing the four screws on each end of the unit and the ten screws on the top. The power selection switch is located directly behind the power connection. For 230 VAC/50 Hz, “230VAC” will be revealed. You must also change the fuse from a 160 mA slo-blo fuse to an 80 mA slo-blo fuse.

To change the fuse, open the cover on the fuse box directly to the right of the power connection. Turning the cover 1/4 turn will allow you to access the fuse. **Make sure the correct power is selected. The wrong decision can destroy your unit.**

Audio Connections

Connect your feed to the Audio In plug. This connection is made through a 3-pin XLR female plug with the following pin-outs:

Pin #	Function
1	Ground
2	Balanced audio high
3	Balanced audio low

It is best when using professional audio gear to connect everything together using balanced audio connections. Sometimes, however, it is unavoidable to connect to consumer equipment with unbalanced connections. If you must use unbalanced audio, here's how to connect to the DXR.1:

Inputs: Pin 2 to audio high, pins 1 and 3 to ground
Outputs: Pin 2 to audio high, pins 1 and 3 to ground

The Audio Out is via a 3-pin XLR male connector with the same pin-outs as the Audio In connector.

Note: The codec provides two distinct and separate channels. You are transmitting to the receive end through the Audio In XLR connector and the receive end may be feeding cues or other programming back to you through the Audio Out XLR connector. Therefore, the Audio Out channel is used to hear what is being sent to you, not what you are sending. For more on this, see the "Applications" section on pages 20-21.

Contact Closure

A Contact Closure jack is located on the back panel. This is available for controlling other equipment, such as starting a tape recorder or turning on certain cue lights. The contact closure is a 1/8" mini jack, and the contact is completed when the Ready LED lights up. More information on the use of this jack is available in the "Applications" section on pages 20-21. The contact closure connection is not designed to switch high power connections. DO NOT connect AC power through the contact closure connection. The switch is designed to carry low voltage (5-15V at low current — less than 50 mA). If this connection is required to switch a high power device (on-air light, selenoid, etc.), buffer it externally with a low control voltage DC relay.

V.35 and X.21 Connections

Pin Connections for Comrex EIA530 (DB-25) to V.35 Cable

EIA530 (DB-25F)	Signal	V.35 (34-pinM)
1	Shield	A
2	Transmit Data A	P
3	Receive Data A	R
7	Ground	B
9	Receive Clock B	X
12	Transmit Clock B	AA
14	Transmit Data B	S
15	Transmit Clock A	Y
16	Receive Data B	T
17	Receive Clock A	V
24	Data Terminal Ready	H

Pin Connections for Comrex EIA530 (DB-25) to X.21 Cable

EIA530 (DB-25F)	Signal	X.21 (DB-15M)
2	Transmit Data A	2
20	Data Terminal Ready A / Control	3
3	Receive Data A	4
6	Data Set Ready A / Indication	5
15, 17	Clock A	6
7	Ground	8
14	Transmit Data B	9
23	Data Terminal Ready B / Control	10
16	Receive Data B	11
22	Data Set Ready B / Indication	12
9, 12	Clock B	13
1	Shield, Ground	Shell

Level Adjustments

With audio being fed into the system, look at the PEAK light on the front panel. This red light should be flashing **occasionally.** There is no set pattern for the flashing, but if it is flashing about 15% of the time, your audio should be received in good fashion. If the light is on constantly, your audio will probably sound clipped and distorted. If the light is not on at all, the level you are sending will be too low, and you may have noise problems.

The solution to both of these problems is to adjust the level. The adjustment is done on the equipment feeding the audio — mixer, tape machine, etc. The DXR.1 is shipped with the audio input and output level set to 0 dBu. If required, it may be adjusted to either -10 or +10 dBu. This is done via internal jumpers. Refer to the “Internal Settings” section on pages 9-10 for more details.

On the DXR.1 there is no need to select the data rate you will be using. The DXR.1 automatically senses the data rate and adjusts the frequency response to match. When sending 56/64 kbps the frequency response will be 7.5 kHz. At 112/128 kbps, the frequency response will be 15 kHz.

To insure the DXR.1 is working properly and will give you no problems when you make your connection with the receive end, you can run a series of tests. These are called loopback tests because the audio you are feeding into the system is fed back to you through a “loop.” The majority of digital equipment provides the capability of loopback testing to isolate problems that may occur. See the “Trouble Shooting” section on pages 11-13 for more information.

Now the DXR.1 is connected and setup. You are ready to communicate with the other end of your link, so place the call through your Terminal Adapter or CSU/DSU. Once your DXR.1 and the codec on the other end have established contact, the green READY light on the front panel will light up. At this point, you may begin transmitting and receiving audio.

Terminal Equipment

Whatever type of digital telephone channel you use, you will need some sort of device to link the DXR.1 to the data channel. For ISDN, this is known as a Terminal Adapter (TA) and for Switched 56, it is known as a Channel Service Unit/Data Service Unit (CSU/DSU). We are happy to recommend a specific TA or CSU/DSU that is appropriate for your application. At the minimum, any terminal equipment you choose should have:

- Synchronous data port capable of V.35, RS530, X.21, or other balanced data protocol
- Dial pad
- Local and remote loopback capability
- V.120 rate adaptation (used for 56/64 conversion)

Other nice but not required features:

- Dual data ports (for Terminal Adapters)
- RS232 remote dialing
- For Terminal Adapters, BONDING or IMUX capability (required for 15 kHz transmission)
- Memory dial

With dial-up digital networks the receiving end of the call answers automatically. This allows unattended operation at one end of the link.

Section 3. Internal Settings

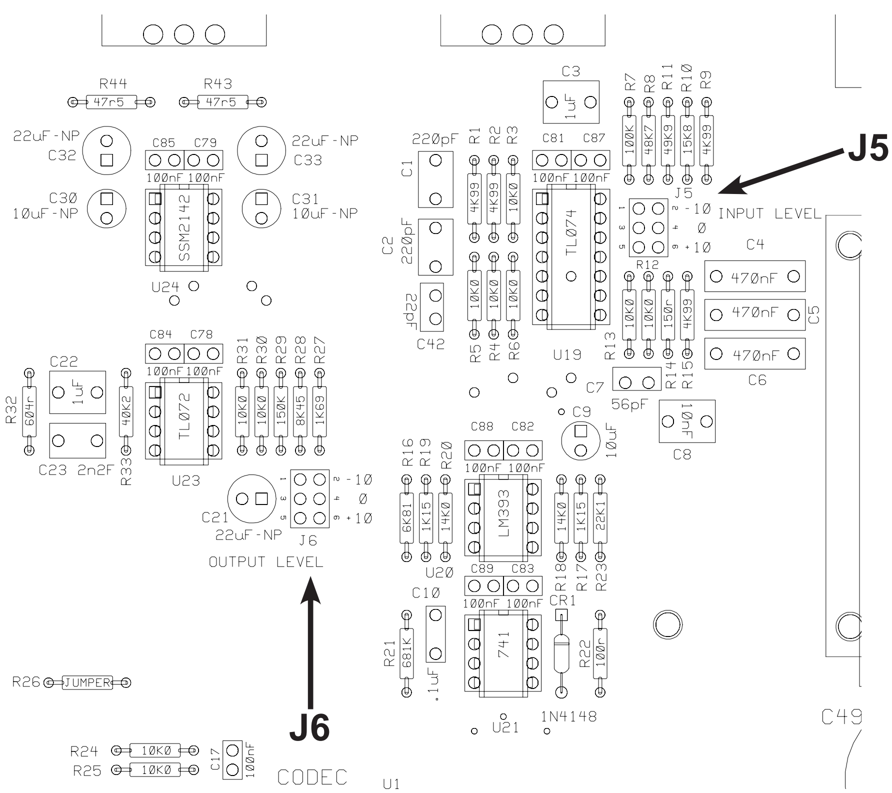
The DXR.1 has been factory configured to meet the majority of applications. However, in this section, we provide information on settings which may be changed to meet special needs. These settings include:

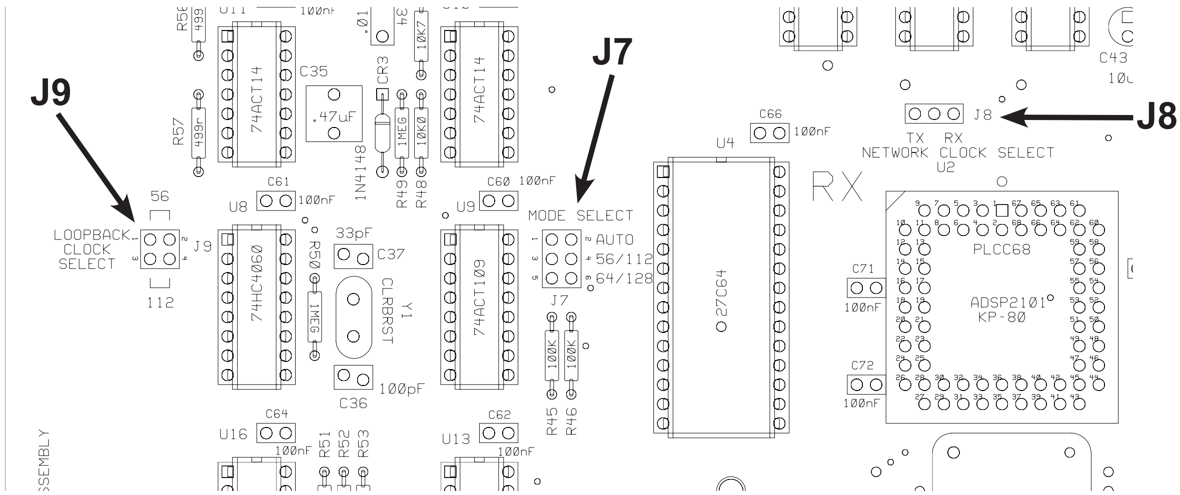
- Audio Input/Output Levels
- Clock Loopback Data Rate
- Mode Select
- Clock Select

Changes are made via internal jumpers on the main PC board. In order to access these jumpers, you must remove the top cover of the chassis. Please remember to disconnect power from the DXR.1 before removing the cover. To change the setting, remove the hood that fits over the pins on the jumper block, and then slide it over the new pin settings.

Audio Input (J5) / Output Levels (J6)

The audio input and output levels are factory adjusted to 0 dBu. If required, it can be modified to either -10 or +10 dBu. This is accomplished by moving the hood on jumpers J5 and J6.





Clock Loopback Data Rate (J9)

The DXR.1 automatically detects the clock rate coming from the Terminal Adapter. For loopback testing purposes, the DXR.1 uses 56 kbps which transmits data at 7.5 kHz. If it is critical that you hear your loopback test at 15 kHz, the jumper may be moved to the 112 kbps position. There is normally no need to move this jumper.

Mode Select (J7)

The DXR.1 automatically detects the data rate and adjusts the audio bandwidth being sent. However, it is possible to "hardwire" the chosen data rate. Moving the jumper to the center position, selects 56/112 kbps. The opposite end selects 64/128 kbps.

Clock Select (J8)

This jumper allows the transmit/receive clock to be changed. It is factory configured to sample the RX clock but may be moved to the TX position to sample the TX clock. This modification is not usually required.

SECTION 4. TROUBLE SHOOTING

The Comrex DXR.1 coding algorithm eliminates redundancy in audio. For this reason, the DXR.1 cannot be subjected to traditional specifications of distortion and signal-to-noise ratio. Most tests done with the codec should be by subjective listening between the original source material and codec-processed audio. Because of the algorithm's dynamic processing properties, tests done with tones tend to prove little.

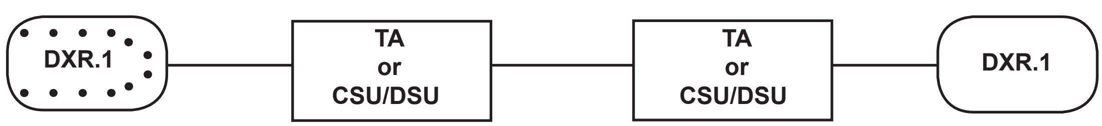
Unlike analog technology, which might work but just be a little off, digital technology tends to either work perfectly or not at all! The trick is to isolate the source of the problem to either the telephone network or the equipment attached to it, so you know where to turn for a solution. We will start with testing the first link in the chain, the DXR.1, then the Terminal Adapter or CSU/DSU and finally the network.

DXR.1 LOOPBACK

This test examines the codec independent of other equipment. Audio must be fed into the unit through the AUDIO IN plug, and you must be able to monitor the audio coming out. Power must be connected to the DXR.1.

Locate the CLOCK AND DATA LOOPBACK switches on the rear panel. Move both of these switches to the up position. This activates two things. It puts the local clock into use to drive the signal, and it connects the encode and decode channels.

DXR.1 Loopback



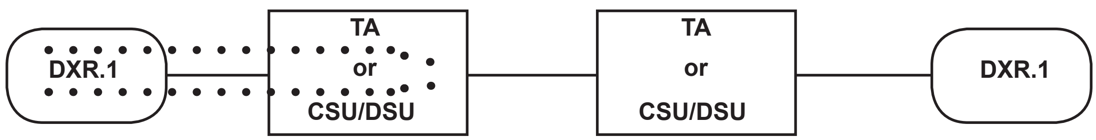
The READY light on the front panel will be green, and you will hear the same audio that you are feeding into the DXR.1. If you do not hear audio, or it is distorted, check your connections. Make sure the power light on the front panel is ON and the peak light is flashing occasionally. If there is still a problem, please contact Comrex to arrange for repair of the DXR.1.

Terminal Adapter or CSU/DSU Loopback

Check the manufacturer's directions for exact instructions on engaging the loopback function. This may be called "local" or "DTE" loopback. Move both loopback switches on the DXR.1 to the normal (down) position. This allows the CSU/DSU or Terminal Adapter to control the testing.

The audio will be going from the source, through the codec and coming back to the codec from the Terminal Adapter or CSU/DSU without ever getting on the network.

TA or CSU/DSU Local Loopback

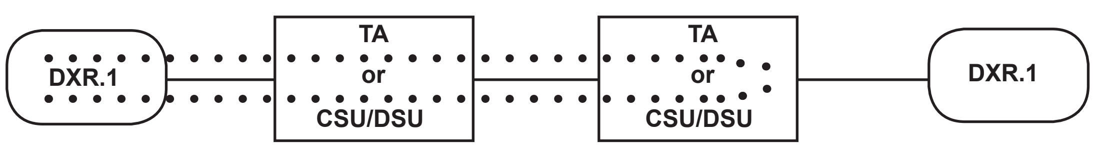


The audio you hear through Audio Out should be the same as what you are feeding into the system, and the DXR.1 Ready LED should be lit. If the codec checked out fine in local test, and the Terminal Adapter or CSU/DSU local loopback fails, the problem is with the Terminal Adapter, CSU/DSU or cable. Contact the manufacturer for assistance.

Remote Loopback

Remote loopback allows audio to be sent down the network, received by the far end Terminal Adapter or CSU/DSU and returned to you. This test will check out the network but requires the capability of remote loopback on your Terminal Adapter or CSU/DSU. The terminal equipment must be of the same type (both ISDN, both SW56, etc.) on either end of the connection. If the types are incompatible, remote loopback will not be possible. Make sure both switches on the DXR.1 are in the normal (down) position before you begin.

TA or CSU/DSU Remote Loopback



If you can hear your own audio with remote loopback engaged and the Ready light is lit, then you know for sure the DXR.1, the Terminal Adapter or CSU/DSU and the network, as well as the remote Terminal Adapter or CSU/DSU are working fine.

Of course, you are always welcome to contact us if you encounter a failure of the system. However, please remember that if the failure lies either in the Terminal Adapter or CSU/DSU, the cable or the network, there is little Comrex can offer in terms of assistance. Most manufacturers of digital equipment have diagnostic tests built into their equipment similar to those we have illustrated. In addition, most digital service carriers have customer hotline numbers for you to report a problem and help you locate the trouble spot on the digital network. We suggest you keep a list of these numbers handy for the occasions when you run into trouble.

SECTION 5. TECHNICAL DETAILS

G.722 ALGORITHM

The codec is a system that encodes and decodes audio signals for transport over digital networks. At the transmit end, the information is encoded and it is decoded at the receive end. Simple. Well, not so simple. As with most things in the world, if everyone created their own method of doing things, nothing would work together. Something as simple as the standardization of power plugs means that we don't think twice about buying appliances or electronic components. But we do think twice about what format our videotape is in — VHS or BETA. But at least it is a small field from which to choose. The same thing is happening with the compression algorithms used to encode and decode audio signals.

International standards bodies have formed to create standards. There are different standards available (such as VHS and BETA), and it is up to you to select which one you will implement. It is also up to you to insure that the vendor you select is implementing the standard with no changes (that can mean your equipment will not work with other manufacturers' equipment, and you will be boxed into a corner).

The international standard known as CCITT G.722 specifies the algorithm that codecs use to convert analog to digital signals and vice versa. The DXR.1 follows this standard very carefully and can communicate with G.722 codecs from other manufacturers.

THEORY OF OPERATION

The Comrex DXR.1 performs a digital algorithm in real time on sampled digital input audio. The unit is based on two high speed computer chips, known as digital signal processors (DSPs). The idea behind the codec (and any other DSP-based device) is to perform functions on analog signals which have been divided into samples taken at discrete times. These samples are then "quantized" (assigned a fixed value) and fed as a stream of binary numbers into the DSP.

The basic assumption of the codec is that digitized audio contains more information than is needed to reproduce it in analog form. By eliminating this redundant information, more audio information may be stored or transmitted.

As mentioned before, the input to the DSP portion of the codec is a series of discrete time samples. Each portion of the codec link (transmitter and receiver) contains a computer program which can predict the next sample based on previous values processed. This function is performed identically in the transmitter and receiver. The difference between the transmitter and receiver is that only the transmitter knows the true value of the next discrete time sample. Since it already possesses an approximation of this value, it can calculate the difference between the two numbers it possesses. This difference is what the transmitter sends to the receiver. The receiver uses this difference to calculate the true value. Since the difference signal contains less than the data sample, data rate is conserved.

In human speech (and most other audio), much more energy exists in the lower part of the audio spectrum than in higher frequencies. Therefore, the codec reproduces audio more accurately at the lower end than at the higher end. Using digital filters, audio is divided between high and low sub-bands, and each sub-band is sent through the encoder-decoder combination separately. The lower band can then use up the majority of the bits available, leaving only a few for the relatively less complicated high band.

Discrete time sampling and quantization of an analog waveform are known as Pulse Code Modulation (PCM), since the codec algorithm uses differences between samples. The predictors adapt automatically with changing values of previous input samples, so we call the algorithm used Adaptive Differential PCM, or ADPCM. When we add the concept of dividing and conquering individual bands, the process becomes Sub-band (SB) ADPCM. SB-ADPCM is defined as an international standard by the CCITT as recommendation G.722. The text of this specification is public information and is a good source for further information on this algorithm.

Synchronization

The transmitting codec forms its outgoing data into "words," each consisting of seven or eight characters. The receiving codec is able to decode and decompress data intelligibly because it has identified the beginning and end of each "word" it receives. This process of identifying and aligning with the correct word order is called synchronization.

The DXR.1 uses a "self synchronizing" technique which allows the encoder to use the entire channel for audio data. With no overhead for synchronization data, the decoder can determine the sync position by performing an algorithm on the raw, incoming data. It takes about one second for the decoder to "find sync" and begin decoding data. The Ready light on the front panel is an indication that the decoder is "in sync."

Every half second, the DXR.1 re-checks to make sure that it is still in sync. If the network causes an error that makes the data stream line up differently, the codec can determine this and "re-sync" within one second. Remember, because the DXR.1 is fully duplex, it is simultaneously encoding/compressing outgoing information and decoding/decompressing incoming information.

ISDN AND SWITCHED 56

Throughout this manual, we refer to two data rates: 56 and 64 kbps. These rates and multiples of them are the most common you will experience with digital telephone services. When you place a normal phone call on the public telephone network, the link is established in a particular pattern. It is an analog signal from you to the central office. (The central office is a telephone facility located at the end of your local loop. It usually contains a switch connected to trunks.) A 64 kbps data link is established between the central office nearest you and the one nearest the called party. From the central office nearest the receiving end, the information is converted back to analog for delivery to the called party. This type of transmission is perfect for voice communication because the bandwidth your voice produces is very narrow. But as the electronic age progresses, higher bandwidths are needed to relay information such as pictures and high quality audio. The digital portion of the phone line has to be extended to the user.

Most digital telephone circuits are based on the concept of using this same digital telephone network channel and extending the digital portion to the user, eliminating the analog section. Basic Rate Interface Integrated Services Digital Network (BRI ISDN) provides the user with access to two of these channels, which are multiplexed onto the same pair of wires and sent to the user. These two channels (called “B” channels) are completely separate. They may be dialed independently and may be used for voice or for different types of data transmission. In North America, some of the public telephone networks use a form of signaling which limits the user bandwidth to 56 kbps channels. In this case, both ends of the telephone link must set their equipment for this lower data rate. In some areas, the 56 kbps (or multiples of it) setting may be “safer,” allowing all calls to complete properly regardless of where they are located.

Also in North America, since ISDN has been slow to proliferate, simpler capabilities are available in a service called Switched 56. As the name implies, Switched 56 is limited to 56 kbps, but can inter-operate successfully with an ISDN line set for this speed. Multiple Switched 56 lines would be required to achieve higher data rates.

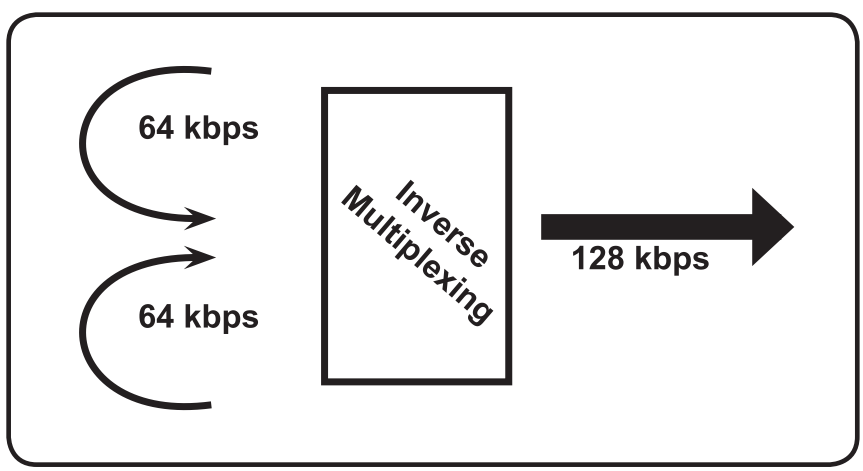
Digital Data Service (DDS) has been available for quite some time in North America. This service provides a point to point dedicated 56 kbps or 64 kbps telephone line. DDS may yield significant cost savings for applications that require full time or near full time exchange of audio. Fractional T1 (or E1), a dedicated link capable of higher data rates (384 kbps), is sometimes available. Wireless modems and portable satellite earth stations may also be used with codecs.

Inverse Multiplexing

Inverse Multiplexing, or IMUXing for short, sounds complicated but is actually quite simple. It means combining two or more lower data rate channels into one, higher data rate channel. It is an extremely important concept when working on digital phone lines like Switched 56 and ISDN, as digital transmission channels on these services come in chunks of 56 or 64 kbps. These chunks have very little to do with each other normally. They may be routed differently throughout the telephone network and incur substantially different transmission path delay. Even the two "B" channels of a Basic Rate Interface ISDN installation offer no guarantee that both calls will be routed along the same path. On a North American coast-to-coast linkup, for example, the first "B" channel connection may be routed via Texas and the second via Michigan.

The IMUX must be able to measure the time delay between the two digital channels and delay the fastest so that it arrives synchronously with the slowest. This procedure is called "aggregation" and is performed differently with different IMUX protocols.

When using BRI ISDN, you will find that several reasonably priced Terminal Adapters have an IMUX built into them, usually using a protocol called "BONDING." These IMUXes work quite well and may be used with the DXR.1 to send 15 kHz over ISDN lines.



Mix-Minus

Even the simplest remotes are a two-way process. The remote site must send its audio to the studio and receive a return feed to monitor the programming. This return feed may be done over a radio station's regular transmitter (with an AM or FM radio at the remote), a special radio link or a telephone circuit. This feed may just go to headphones at the remote, and it may also be put on speakers for the local audience.

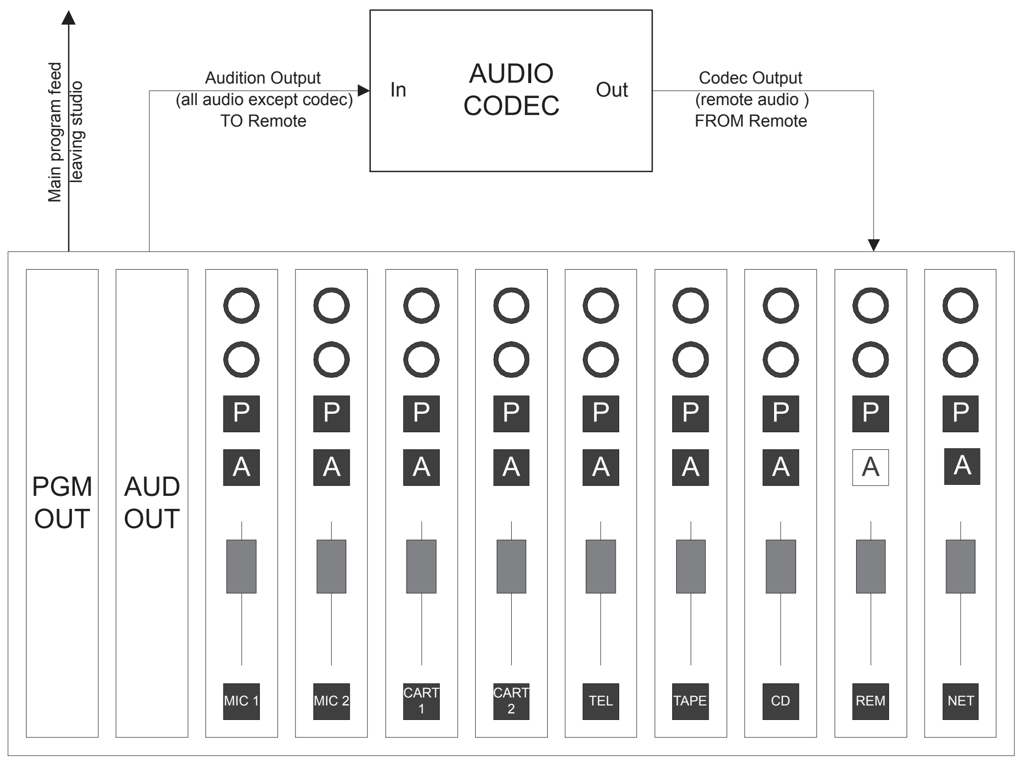
The problem comes when there is a time delay in getting audio to and/or from the studio. In this case, the remote talent hears a delayed version of their voice in the headphones and may find this very distracting. Even a remote done with simple equipment or a frequency extender on plain phone lines may have this problem on a long-distance call. All remotes using ISDN, Switched 56 and POTS codecs will have delays each way as signals are processed from analog to digital, compressed, uncompressed and converted back to analog audio. Some digital compression schemes, such as G.722, result in shorter delay times, but there will still be a "reverb" effect in headphones at the remote site, if their audio is sent back from the studio. In any of these cases, it may not be possible for the remote people to listen to an off-air or program channel feed.

The solution is **mix-minus**. A mix-minus feed has a mix of all of the programming on the radio station (or network) **minus** the audio from the remote. In other words, the station or network doesn't send the remote audio back to the remote. At the remote end, this mix-minus feed is converted back to an "air monitor" by mixing in the local audio from the remote.

For radio stations, in addition to fixing the time delay problem, using a mix-minus feed has two other advantages. First, if the station uses a 6-7 second delay to allow editing of phone calls, pre-delay audio can be sent to the remote site. Second, if there is a PA system at the remote, they will be able to run the speaker levels higher with the mix-minus audio. This is because the remote microphone audio is not running through the station's audio processing, and the levels stay under the control of the remote operator.

The simplest way to do one mix-minus feed in a typical radio studio is to use the Audition or second program channel. On many audio consoles, each fader's output may be sent to both Program and Audition. If your board will allow those feeds simultaneously, just set all of the modules to Program and Audition, **with the exception of the one carrying the remote audio. Set that one to Program only.** The Audition channel will then be a mix of everything on the console except the remote. That will be your mix-minus, and it should be sent to the remote site.

One caution — make sure that audio is being sent to and from any telephone modules you may have in the console. They may have been designed to work with only one channel at a time, either Program or Audition, but not both. If so, you will have to check with your "tech guy" or the board manufacturer for advice. If you use multiple audio codecs, you should investigate the Comrex Mix-Minus Bridge. This will allow you to expand one Program/Audition setup to handle five codecs or other remote audio devices. It also provides IFB (talkback) to remote sites.



"I'm Using Mix-Minus, and I Still Hear an Echo!"

If you are doing a call-in talk show on the road, the remote people may complain of hearing an echo when a caller is put on the air. With the telephone pot down, everything is OK. The culprit is the telephone hybrid being used to put callers on the air. Some of the remote audio is "leaking" through the hybrid and mixing with the caller audio. Modern digital hybrids do a much better job of preventing this than the older units that had to be manually "tweaked" for each call. If you are using a digital hybrid and having this problem, dig out the manual and redo the hybrid's initial setup.

Section 6. Applications

The DXR.1 provides a cost saving alternative to satellite feeds or dedicated circuits. They are ideal for applications requiring high quality (7.5 or 15 kHz) mono audio. There is very little processing delay, making the DXR.1 a good choice for live programming such as talk, news and sports, where real-time cueback is essential.

Network Distribution

Sports reports are often recorded and made available to many stations. Each station dials into the servicing station through their own codec. Once the receive and transmit codecs have gone through a handshaking ritual to insure they are compatible, the DXR.1 Ready light on the transmit end will light up. Through the use of the contact closure jack, a tape recorder is started, and the sports report is transmitted to the receiving station. Once the connection is broken, the switch is opened, and the tape recorder is returned to ready state — waiting for the next call.

Auto Recording

The same technique can be used to record information. Reporters in the field may complete stories at any hour of the day. By calling into the DXR.1, the handshake between codecs is accomplished and the recorder is started via the contact closure jack. This provides for unattended gathering of stories for either direct broadcast or editing at a later date.

Automation Tones

Broadcast networks sometimes use sub-audible tones in the range of 20 to 50 Hz to allow affiliates to automate their network cutaways and rejoins. 25 Hz is a popular frequency, and the combination of low-frequency roll-off in the DXR.1 and our customer's equipment may provide an output too low for a tone decoder expecting full line level. Since the G.722 processing in the DXR.1 "favors" the lower audio frequency band by allocating more bits in the coding process, the automation tones are handled very well in the digital processing. There are two points in the DXR.1 analog section, however, that will restrict the low frequency performance.

Referring to the DXR.1 board schematic included at the end of the manual, capacitor C4 couples the output of U19D to resistor R12, which sets the input impedance to U19C. The factory-supplied value of C4 is 470nF (0.47mF). Increasing this value to 2.2mF, a commonly available value for non-polarized capacitors, will improve the low frequency response of the encoder section. Larger values are fine, but we recommend a non-polarized unit. In the decoder, a change would only have to be made if the DXR.1 audio output was terminated in a true 600 ohm load. If so, the 22mF output coupling capacitors, C32 and C33, which are connected to U24, can be eliminated by adding a jumper across each of them.

Just be aware that you may present a small d-c offset voltage to your equipment. The alternative would be larger non-polarized electrolytic capacitors designed for audio use. If the DXR.1 is feeding a bridging load (>5K ohms, typical of most console inputs), the 22mF capacitors may be left in place.

If you make these changes and are still having problems, you should check that the automation tone level at the DXR.1 input is actually up to the program line level. Small audio transformers in any equipment you may have between the tone generator and the DXR.1 may either reduce the level of the tones or add distortion products. Many transformers will not handle 25 Hz tones at line level, despite the manufacturer's claims. The same reasoning applies at the receiving end in any equipment between the ISDN codec and the automation tone decoder.

SECTION 7. GLOSSARY

CSU/DSU Channel Service Unit/Data Service Unit. On some data networks, these are two separate devices. On most networks used with the codec, this is a box which sits between the codec and the data circuit, used to interface and condition the data coming on and off the network. This box may also contain diagnostic testing functions and indicators, and in the case of switched services, will perform all your dialing functions. A CSU/DSU is required for all SW56 and DDS circuits and is not included with your codec.

DATA PORT The physical and electrical protocol used by the codec and the TA or CSU/DSU to transfer data between each other.

DATA RATE Analog transmission media is specified in bandwidth (usually in Hertz) and signal to noise (usually in dB). Since the principles behind digital transmission are so different, media are specified in different parameters. Rather than how much analog information is passed, a digital user is concerned with how many bits per second can be sent down the channel.

DIGITAL VS. ANALOG An analog electrical signal (or sound or light, etc.) is noted by a fundamental change in character with respect to the information being conveyed. For example, an AM radio station changes the amplitude of a carrier signal to varying degrees depending on the amplitude of the music it is carrying. A digital signal is always in one of two states (on or off), but varying at a rate fast enough that information encoded into numbers (quantized) can be transferred. Another way to distinguish analog from digital is that an analog signal has an infinite number of “degrees” of changes which convey information. A digital signal has only two. One of the largest advantages of digital transmission is that as long as a receiver can distinguish between the two states in the signal, noise will have no effect on it.

DSP Digital Signal Processing. The concept of sampling analog waveforms in discrete time and manipulating these samples using algorithms which would be difficult or impossible in the analog domain.

DTE/DCE Data Terminal Equipment/Data Computer Equipment. To avoid confusion, the data protocols mentioned above designate equipment and ports as either DTE or DCE. The TA or CSU/DSU is ALWAYS the DCE, and the codec is ALWAYS the DTE. Plugging two DTEs together will not establish communication between them, since the DCE provides all the clocks required to run the data around.

Duplex

On plain old telephone service (POTS), the audio transmission can be considered “half duplex,” since if both parties speak at the same time, their voices will intercept on the single pair of wires on each end of the call. Most digital systems are duplex or “4 wire,” allowing simultaneous and independent data (or encoded audio) to pass in each direction. Some systems may be “simplex” which pass digits only in one direction.

ISDN

Integrated Services Digital Network. The worldwide standard for digital telephony. ISDN actually describes a complex set of international standards concerned with digital telephony. This service is currently available in two configurations. Basic Rate Interface (BRI) provides the user with two independent 64 kbps switched channels. Primary Rate Interface (PRI) allows user data rates approaching those of T1 (1.54 Mb/s).

Loopback

Analog signals are easy to test. One simply probes the point of interest with an oscilloscope and checks for the proper signal. High speed digital signals complicate things because they can’t be measured easily by traditional test equipment. To make tests easier, digital equipment often
comes with loopback capability. Borrowed from the telephone repairmen who had no wish to trace miles of circuits, loopback diagnostics allow you to “instruct” a piece of equipment in your digital link to “echo” any information sent to it in the reverse direction. When properly looped back, the codec should echo (from the receiver output) any audio sent into the transmitter. By enabling loopback at different points on a network (i.e. TA or CSU/DSU, codec, remote TA or CSU/DSU loopback) the defective portion can more easily be determined.

Noise

In this application, noise is anything present between the codecs other than the binary signal being transmitted. Digital signals are by nature immune to noise until the noise level approaches the point where a “1” looks like a “0” and vice versa. Digital circuits are susceptible to two different type of noise. The first type is just background noise which occasionally reaches a sufficient level to cause bit errors. The other type, known as “burst errors,” cause incorrect data to be sent for a distinct amount of time. Bit errors cause little noticeable degradation to ADPCM audio, due to the nature of the algorithm. Burst errors cause severe distortion and may cause loss of synchronization.

PCM/ADPCM Pulse Code Modulation (PCM) is the technique used by CD players and other devices to "digitize" audio. The codec converts PCM to Adaptive Differential PCM in order to conserve media bandwidth.

Switched 56 In North America, this service is very similar to DDS but allows the user to place "calls" between several points and cut costs by only paying for part time service. Considered to be a precursor to ISDN.

Synchronization Many data networks simply provide a "bit stream" to the user, without any information on how to divide these bits into "words." Some networks use overhead bits to determine the "start" and "stop" of each word. If the codec is to give you the best possible use of your data channel, it can't afford the overhead required to provide alignment information, if it isn't provided by the network. For this reason, the codec's receiver takes in raw, unframed data and analyzes it to determine the proper word alignment. This "auto synchronization" scheme allows the receiver to frame words with no other information. In the case where you are communicating with a less sophisticated codec (from another manufacturer), the transmitter may be configured so that output words line up with network timing information, if available. This assures that the other codec will receive words aligned with the network timing. Under most circumstances this won't be necessary or desirable.

Terminal Adapter (TA) This can be thought of as a CSU/DSU for an ISDN line. Its function is actually to adapt non-ISDN equipment to the ISDN user rate. It may also provide you with a choice of bearer services, which determine the type of ISDN call you will make. For the most part, the codec requires the placement of a 56 or 64 kbps clear channel data call.