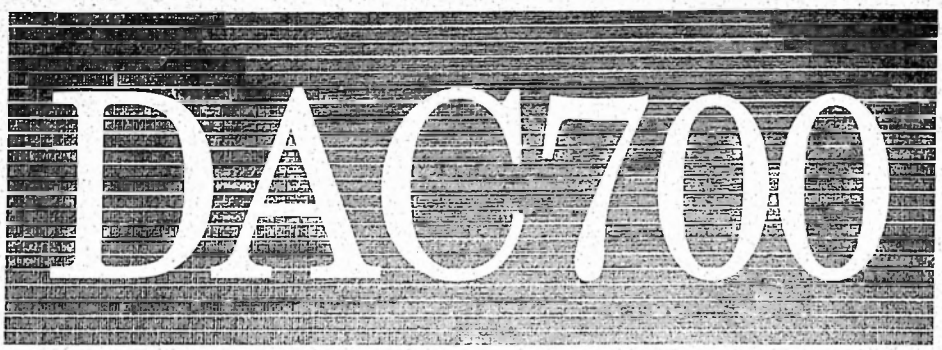


Digital Audio Codec/Multiplexer Subsystem

Installation and Operation Manual

ComStream®

10180 Barnes Canyon Road • San Diego, CA 92121 • (619)458-1800

CPN 01-0479-401
Rev. A 02/95

DAC700

Digital Audio Codec/Multiplexer Subsystem

Installation and Operation Manual

ComStream
A Spar Company

10180 Barnes Canyon Road • San Diego, California 92121 • (619) 458-1800

CPN 01-0479-401
Rev. A 02/95

NOTICE

This publication and its contents are proprietary to ComStream Corporation (ComStream) and are intended solely for the contractual use of its customers for no other purpose than to install and operate the equipment described herein. This publication and its contents shall not be used or distributed for any other purpose and/or otherwise communicated, disclosed, or reproduced, in any way whatsoever, without the prior written consent of ComStream.

For the proper installation and operation of this equipment and/or all parts thereof, the instructions in this manual must be strictly and explicitly followed by experienced personnel. All of the contents of this manual must be fully read and understood prior to installing or operating any of the equipment, or parts thereof. **FAILURE TO COMPLETELY READ AND FULLY UNDERSTAND AND FOLLOW ALL OF THE CONTENTS OF THIS MANUAL PRIOR TO INSTALLING AND/OR OPERATING THIS EQUIPMENT, OR PARTS THEREOF, MAY RESULT IN DAMAGE TO THE EQUIPMENT, OR PARTS THEREOF, AND TO ANY PERSONS INSTALLING AND/OR OPERATING THE SAME.**

ComStream does not assume any liability arising out of the application or use of any products, component parts, circuits, software, or firmware described herein. ComStream further does not convey any license under its patent, trademark, copyright, or common-law rights nor the similar rights of others. ComStream further reserves the right to make any changes in any products, or parts thereof, described herein without notice.

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

WARNING

Shock Hazard!

Do Not Open The DAC700 Equipment!
Service Only By ComStream!

Gefährliche Spannung!

Öffnen des Gerätes und Service nur durch ComStream!

The DAC700 contains no user-serviceable parts. Do not attempt to service this product yourself. Any attempt to do so will negate any and all warranties.

Table of Contents

List of Figures

List of Tables

ComStream
A Spar Company

Preface

Using this Guide

Welcome to the ComStream world of satellite-based communication systems and networks. This guide is your sourcebook for using ComStream's DAC700 Codec/Multiplexer. It describes the installation, operation, and performance specifications of this product.

The chapters in this guide provide step-by-step instructions for a variety of tasks and activities, including setting up, mounting, and operating the DAC700. The chapters also provide an overview of system operations, as well as technical specifications and troubleshooting procedures.

Getting Started

If you are new to satellite communications or are unfamiliar with either ComStream products or the DAC700 codec/multiplexer, you should read the following chapters before unpacking or installing this product:

- Chapters 1, 2, and 3 for an overview of the DAC700
- Chapter 3 for DAC700 installation procedures
- Chapters 4 and 6 for remote monitor and control operation and maintenance and troubleshooting
- Appendix B for a list of abbreviations and acronyms
- The glossary for definitions of commonly used terms

If you are an experienced user familiar with the DAC700 and ComStream products, you may wish to begin with:

- Chapter 1 for an overview of the DAC700
- Chapter 3 for the installation procedure
- Other chapters as needed

Conventions Used in this Guide

This guide is designed to help you find and use information quickly and easily. To take full advantage of this design, please take a moment to review the specific formats.

Locating Information

There are several tools located in this guide to help you quickly locate information. The table of contents, located at the beginning, provides you with an outline of the chapters and major topics contained within them.

The glossary and index are located at the end of this guide. The glossary contains technical terms and system commands for easy reference. The Index can be used to help you quickly locate information.

Special Paragraphs

Throughout this guide you will find two special paragraphs designed to help you identify important information. These paragraphs are:

NOTE: This identifies information for the proper operation of your equipment, including helpful hints, shortcuts, or important reminders.

CAUTION: This identifies information that requires careful attention in order to prevent equipment damage and/or injury to the operator.

This symbol is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance.

Keyboard Entries

Each activity or task is presented in a series of numbered, step-by-step instructions. Commands or information that you type into the system appear in a different, bold type:

Type **`ben 1`** and press Enter.

Keys that are pressed in combination appear with a plus sign (+). For example:

Control+Shift+3

You hold down the Control and Shift keys and press the number 3.

Warranty Statement

ComStream warrants that its products are free from defects in material and workmanship at the time of shipment and that they conform to applicable specifications. In no event will ComStream be liable for consequential misuse or damages.

The ComStream DAC700 Codec/Multiplexer subsystem is warranted against any above-mentioned defects that appear within one year of shipping date.

Should it be necessary to make a claim against this warranty, the buyer shall first notify ComStream's Customer Service Department to define the nature of the problem. When returning products, please be aware of the following:

1. Products returned to ComStream, whether for upgrade, warranted or out-of-warranty repair work, or maintenance, must comply with the ComStream Return Procedure (located on the next page).
2. Products shall be forwarded to ComStream, transportation prepaid.
3. Products returned to ComStream freight collect or without a return material authorization (RMA) number will NOT be accepted.
4. ComStream shall not accept any responsibility for returned products that are improperly packaged and/or damaged in shipment. If possible, please use original shipping and packing materials.
5. Original product identification markings and labels must not be removed, defaced, or altered. Further, to preserve the warranty, the product should not be subjected to abuse, improper installation or application, alteration, accident, or negligence in use, storage, transportation, or handling.
6. Any returned product shall be completely evaluated in an attempt to duplicate the problem so that appropriate corrective action and repair may be completed. Following repair, the product shall be thoroughly tested for compliance with appropriate specifications. This process will be handled in an expedient and prompt manner but may be subject to available labor and material resources.

The ComStream warranty, as stated herein, is in lieu of all other warranties, expressed, implied, or statutory.

For further information, please contact
ComStream Customer Service at
619-657-5454.

Return Procedure

If it is necessary to return a product for out-of-warranty repair, upgrade, or any modification, the following procedures must be followed:

1. Contact ComStream Customer Service, located in the United States, via phone or fax:
   - Phone 619-657-5454
   - Fax 619-657-5455
2. Speak to a ComStream customer service representative about any questions, issues, or problems. Quite often equipment problems can be corrected over the phone, which keeps your equipment in service and avoids unnecessary and costly downtime.
3. Should it be necessary to return a product to ComStream for any reason, the ComStream customer service representative will issue you a return material authorization (RMA) number. To issue an RMA number, the ComStream representative will need the product's serial number, model number, and a description of the problem.
4. You may be returning a product for either repair, upgrade, or modification. If you are returning the product for:
   - Repair, please include a complete description of the problem, the operating conditions which caused the problem, and any circumstances that may have led to the problem. This information is essential for ComStream repair technicians to reproduce, diagnose, and correct the problem.
   - Upgrade or modification, please include a complete description of the current configuration and the desired change(s). This information will allow a ComStream customer service representative to provide a formal quote for the upgrade.
5. Include a purchase order for any upgrade or out-of-warranty repair work being performed. ComStream will begin repair work after a PO is received.
6. Reference the RMA number on all paperwork that accompanies the equipment, and write the RMA number clearly on the outside of the shipping container.
7. Ship your module in the original shipping carton and packing (or its equivalent), prepaid, to the following address.

   ComStream, A Spar Company
   10180 Barnes Canyon Road
   San Diego, CA 92121 USA

   RMA Unit number

   Do not include product accessories such as Installation and Operation guides or rack-mount brackets.

CAUTION: When handling or shipping static-sensitive equipment, observe antistatic procedures and always use antistatic bags for shipment. Upon request, ComStream will provide you with ESD bags for your use.

All equipment upgrade and repair requests will be completely evaluated and the required work performed in an expedient and prompt manner. The equipment will then be thoroughly tested for compliance with appropriate specifications.

Revision History

This guide is periodically updated and revised. The following table lists the revision number and date and provides a description of the type of revision made to the guide.

To determine if you have the most current documentation, you can compare the revision information at the bottom of each page to those listed in the Revision History table below. For documentation updates, call ComStream Customer Service (located in the United States) at 619-657-5454 or fax your request to 619-657-5455.

NOTE: Revision A is always the first release to ComStream customers.

Table 1. Revision History

Revision	Date	Pages
Rev. A	02/95	Initial release

Customer Support

We hope this guide provides all of the information and instructions you need to operate the DAC700 Codec/Multiplexer.

However, in the event that you need further assistance, or if problems are encountered, ComStream has set up a Customer Support Line for your use. Please feel free to contact ComStream Customer Support, located in the United States, by phone or fax at the following numbers:

- Phone 619-657-5454 Monday through Friday, 8:00 a.m. to 5:00 p.m. Pacific time
- Fax 619-657-5455

COMSTREAM
A Spar Company

Chapter 1: Overview

Introduction

The DAC700 audio codec/multiplexer subsystem is a key component of the ComStream digital audio broadcast system. The following equipment is all that is needed to provide a complete CD-quality, digital audio satellite uplink:

- DAC700
- ComStream CM701 satellite modem
- Radio Frequency (RF) terminal
- Antenna

The DAC700 can be set up in an audio broadcast system in many ways. Two of the most common ways are:

- As an unprotected or protected encoder/decoder configuration, in which the DAC700 transmits compressed International Standards Organization/Motion Picture Experts Group (ISO/MPEG) (Musicam) audio, data, and cue controls. This data is then broadcast to, and can be received by, the ComStream family of audio receivers, the ABR200/75.
- In a full duplex codec configuration, the DAC can operate point-to-point with a DAC700, a CM701 modem, and an RF terminal at each end of the link. In this configuration, audio, data, and control data can be transmitted in both directions simultaneously. Asymmetric transmit and receive rates are supported.

This chapter provides an overview of a typical satellite digital audio distribution network as well as an overview of the DAC700.

Satellite Digital Audio Distribution Network Overview

A satellite broadcast network consists of the following three major subsystems shown in Figure 1-1.

- Satellite transmission uplink station
- Satellite link
- One or more remote satellite receivers

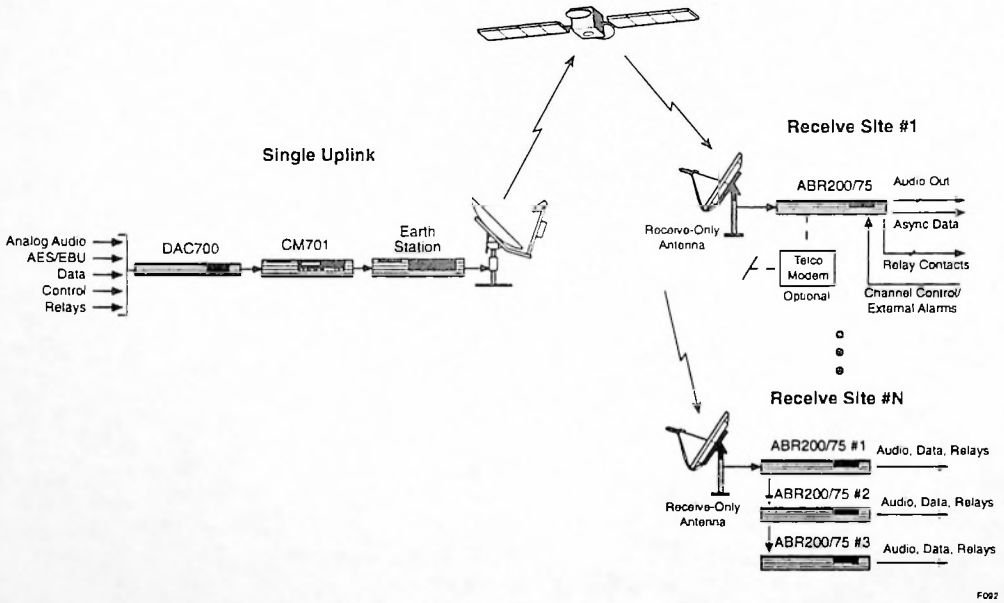


Figure 1-1. Single Channel to Multipoint Audio/Data Satellite Network

The hub or satellite transmission uplink station is the facility where the audio to be transmitted is collected and uplinked to the satellite. This facility consists of the following:

- Audio encoder/multiplexer, such as the DAC700
- Digital modem, such as the CM701
- Earth station, such as the DT7000
- Antenna, such as those provided by Prodelin
- Optional audio network management system, the ANMS
- Optional terrestrial link to provide dial-up diagnostics and performance monitoring

The satellite link consists of a commercial telecommunications satellite in geosynchronous orbit above the earth. Two radio frequency bands that are primarily used are C-band and Ku-band.

The third major subsystem, the remote satellite receiver, includes:

- Satellite antenna subsystem
- Interfacility link cable
- Satellite audio receiver

A number of configurations are possible depending on the size of the network. Networks are often a single channel uplink to multipoint downlinks. On the uplink side, a DAC700 unit connects to a ComStream CM701 modem and RF terminal. On the downlink side, there are a number of ComStream ABR receivers and RF terminals. This configuration is shown in Figure 1-1.

For larger networks, a multichannel uplink to multipoint downlink architecture is used, as shown in Figure 1-2. On the uplink side, two or more DAC700 units connect to CM701 modems and RF terminals. On the downlink side, there are multiple ABR receivers. In this configuration the audio network management/control system (ANMS) is recommended. The ANMS communicates with the entire network of audio satellite receivers via the audio codex/mux unit and provides the system control from the uplink.

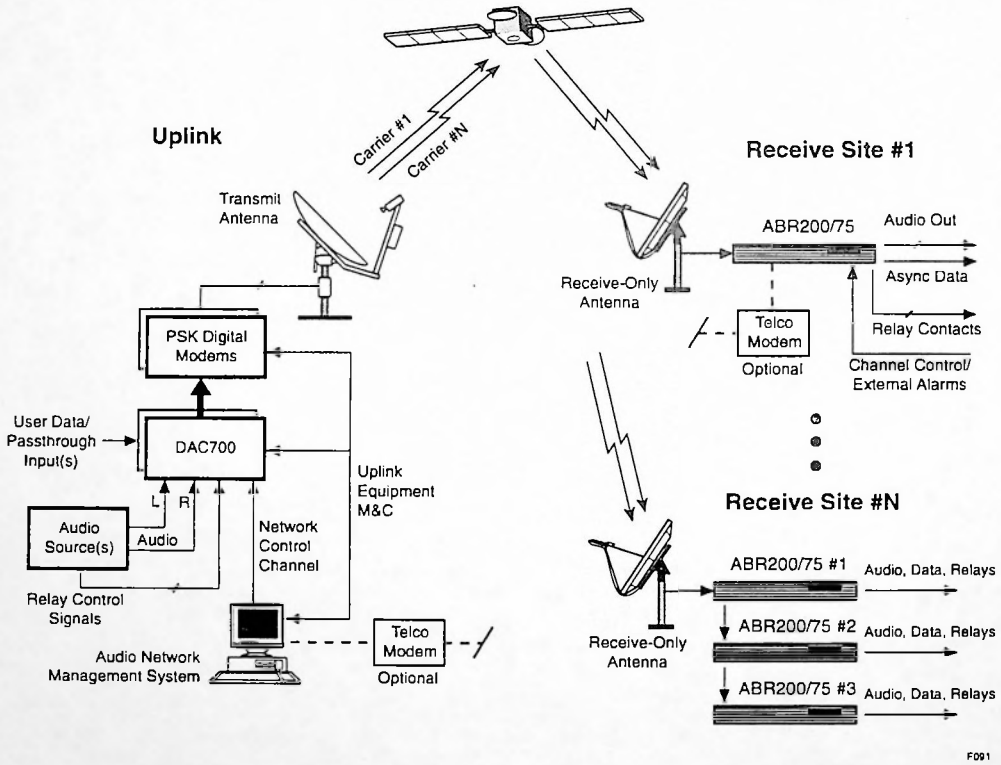


Figure 1-2. Multichannel to Multipoint Audio/Data Satellite Network

DAC700 Features

The DAC700 is a multirate audio encoder/decoder. The DAC700, shown in Figure 1-3, receives either analog audio or AES/EBU input from a studio sound source and translates it to digital audio. A user data port can be connected to a personal computer so that information such as air logs or station traffic can be transmitted with the audio signal. The unit can also transmit relay/control information to the network of ABR receivers. When used with ComStream uplink racks, the unit provides a protection mode that allows it to shadow and swap out channels even if they operate at different encoding and transmission rates.

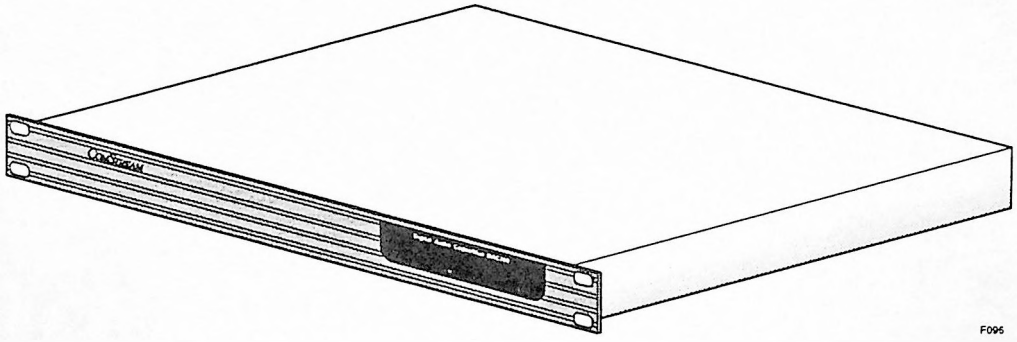


Figure 1-3. DAC700 Audio Codec/Mux Unit

Several capabilities give the DAC700 unmatched versatility for audio and data distribution. The DAC700:

- Is a full duplex, multirate ISO/MPEG (Musicam) encoder and decoder that can transmit and receive mono, dual mono (stereo), or joint stereo analog audio
- Provides full duplex, selectable Audio Engineering Society/European Broadcast Union (AES/EBU) digital input and output for mono, dual mono (stereo), or joint stereo audio
- Is an integrated, multichannel, asynchronous data multiplexer/demultiplexer that can transmit and receive composite user data at rates of up to 9600 bps
- Allows remote control from a desktop computer, portable terminal, or terminal emulator using either an RS-232 or optional multidrop RS-485 connection
- Has a data passthrough mode for direct user data transfer without the internal multiplexer overhead

- Provides 16 individual TTL opto-isolated inputs that can be mapped to an audio receiver's eight output relays for control of external studio units such as a console control or event sequencer
- Provides eight dry contact relay outputs from the onboard decoder to external units such as a studio console control or event sequencer
- Offers users the selection of four different clock sources, one internal and three external
- Has built-in diagnostics that include local and remote loopback of audio/data, operating status, fault alarms, and output status relay
- Provides front panel operating status and fault indicators

The remaining portion of this guide describes in detail the steps necessary to install, configure, and operate the DAC700 audio codec/mux within a network environment.

Chapter 2: Functional Description and Theory of Operation

Overview

The DAC700 accepts either analog audio or AES/EBU formatted digital audio from a studio source. The DAC700 can encode and decode this audio as well as ancillary data using the ISO/MPEG (Musicam) audio compression algorithm. The DAC700 has three operating modes for system audio:

- Mono
- Dual mono (stereo)
- Joint stereo

The DAC700 has the flexibility to have either 600 ohm balanced or >100 Kohm audio input impedance. The factory internally configures the DAC700 for >100 Kohm input impedance. It can also be configured to 600 ohm balanced operation by using an optional cable.

The DAC700 encoder multiplexes ancillary data along with the digitized audio. Ancillary data comprises the following types of data, which provide users with a number of capabilities:

- User data – provides the capability to download pricing information, music play lists, or E-mail files in a "one-to-all" network distribution
- Relay control data – offers precise uplink control of remote receivers and radio studio equipment such as cart machines
- Network control command – provides uplink control of some or all downlink receiver sites for channel or format changes

For more information on the types of ancillary data, refer to *Chapter 3: Installation and Initial Startup.*

The DAC700 decoder section can be configured to accept data either from the onboard encoder or from an external ISO/MPEG (Musicam) encoder (that is, another DAC700). When set up to receive data from the onboard encoder, the decoder section of the DAC700 allows the user to monitor the audio and data that is being transmitted to the remote sites.

The relay/control interface supports both 16 Transistor-Transistor Logic (TTL) opto-isolated sensor inputs and eight decoder output relay contacts. The 16 TTL sensor inputs go to the unit controller for formatting into the system relay control messages. The eight separate form A (SPST) relay contacts are used by the onboard decoder to control external audio or other station equipment.

The DAC700 unit can connect via the Monitor & Control (M&C) port in either RS-232 or RS-485 format to an external portable terminal, desktop computer, or terminal emulator. The computer can be used to issue control, configuration, and diagnostics commands. By means of the DAC700's extensive query command capability, the computer can also be used to monitor unit codec/mux operating status and fault and alarm information.

The clock module produces all the clocks and event timing necessary for the unit. Two separate clock generator circuits are available to support independent, asynchronous transmit and receive operations.

This chapter presents a description of the following:

- DAC700 chassis
- DAC700 front panel indicators
- ISO/MPEG (Musicam)
- Transmission channel signal format
- DAC700 major functional modules
- DAC700 loopback modes

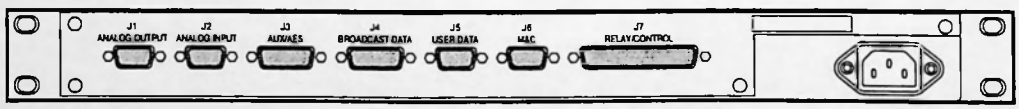
DAC700 Chassis

The DAC700 consists of a single Circuit Card Assembly (CCA) enclosed in a chassis. The chassis consists of a standard 19-inch wide enclosure, one rack unit high. The front view of the DAC700 is shown in Figure 2-1.

F093

Figure 2-1. DAC700 Front Panel

For ease of installation, all input and output connections are made on the rear panel, shown in Figure 2-2.



F090

Figure 2-2. DAC700 Rear Panel

Front Panel Indicators

The front panel of the DAC700 has six indicators located on the right side of the panel, as shown in Figure 2-3. The indicators are backlit to show active operation status or faults in the DAC700 unit.

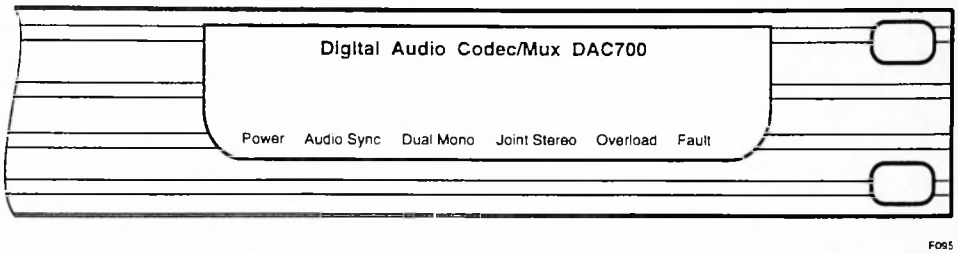


Figure 2-3. DAC700 Front Panel Indicators

The green Power LED indicates the unit is powered on and the power supply is functioning correctly.

The Audio Sync illuminates green to indicate the decoder has ISO/MPEG (Musicam) music frame synchronization.

The Dual Mono is a green LED that indicates the current mode of operation:

- On indicates the audio/digital encoder is performing independent stereo coding in dual mono mode.
- Off indicates the unit is operating in mono mode or joint stereo mode. Refer to the following description of Joint Stereo mode.

The Joint Stereo is a green LED.

- On indicates the audio/digital encoder is in joint stereo coding mode.
- Off indicates the unit is operating in mono mode or in the dual mono operating mode. Refer to the previous description of the Dual Mono mode.

NOTE: When both the Dual Mono and Joint Stereo indicators are off, the unit is in mono operation.

The Overload LED illuminates red to indicate that one or both audio input channels are above +18 dBu. Clipping of the input audio channels occurs at +18 dBu and higher. Refer to *Chapter 6: Maintenance and Troubleshooting* for overload definition.

The Fault indicator illuminates red when a decoder or encoder fault condition exists.

Examples of such conditions are: Receive (Rx) or Transmit (Tx) Phase Lock Loop (PLL) out of lock, and Digital Signal Processor (DSP) hardware errors such as memory fault, watchdog time-out, or muted audio.

The fault indicator function mimics the alarm relay and is masked using the alarm mask (AM) command. See *Chapter 4: Remote Maintenance and Control Operation*.

ISO/MPEG (Musicam)

The DAC700 uses the ISO/MPEG (Musicam) compression algorithm. As the most tested, documented, and reviewed audio compression algorithm in the world, ISO/MPEG (Musicam) has been demonstrated to be a superior compression algorithm.

The importance of the audio compression algorithm is 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 kbps for a CD) can now be reduced by the Musicam algorithm to about 200 kbps as a result of major progress in the development of source coding techniques which utilize knowledge of the human ear. This means that the average quantization of the audio signal at a sampling rate of 48 kHz would be approximately 2 bits per sample in the mono channel, instead of the 16 bits per sample used in CDs.

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. ISO/MPEG (Musicam) also offers the flexibility of independently adjustable digital bit rates (56 kbps, 64 kbps, 112 kbps, 128 kbps, 192 kbps, 256 kbps, 384 kbps ...) as well as embedded data (user data) within the audio bit stream.

ISO/MPEG (Musicam) Compression Concepts

The main principle of ISO/MPEG (Musicam) is the reduction of redundancy and irrelevance in the audio signal. Every audio signal contains irrelevant signal components, such as determination of timbre and localization, which have nothing to do with the identification of the audio signal.

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. Masking thresholds are based on masking tones.

A masking tone is simply a high amplitude audio signal occurring over a relatively narrow frequency span and is often called a masker. A number of these masking tones typically occur at several different frequencies in an audio signal. 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.

The parts of an audio signal below the global masking threshold are inaudible to the ear. 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 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 subbands.

The quantization of the analog time samples required for each individual subband is dependent on the minimum masking value in each subband. This minimum masking level is a measure of the allowed quantization noise which is just below the level of perceptibility. Subbands with desired signals well below the masking threshold, and thus irrelevant for the human ear, do not need to be transmitted.

In each 24 ms period, a calculation of the masking threshold is performed for each subband. This threshold is then used to compute the best psychoacoustical 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 signals changes continuously due to the changing masking threshold.

If there are an insufficient number of bits to completely hide the quantizing-induced noise, 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 a level. The allocation of the extra bits is crucial and allows multiple encode-decode cycles as well as postproduction 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 subband and quantizing the result using 6 bit sampling. The scale factors have a dynamic range of 120 dB, which is sufficient for future encoding of quantized Pulse Code Modulation (PCM) signals using up to 20 bit sampling while still retaining their dynamic range. All necessary information is encoded into Musicam frames, each of which represents about 24 ms of real-time audio.

All of the complex calculations of the ISO/MPEG (Musicam) algorithm are performed by the encoder. Decoders are designed to be universal. ISO/MPEG (Musicam) decoders that correctly decode and play back audio information that has been encoded by a range of ISO/MPEG (Musicam) encoders can be constructed. This aspect to the ISO/MPEG (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

Four fields of use are commonly described in discussing performance considerations:

- Contribution
- Distribution
- Emission
- Commentary

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

Distribution grade systems are used to transmit audio between two storage devices or from a central production center to individual stations or transmitters. However, the number of subsequent encode-decode cycles is limited to only a few. Distribution grade systems are used when the number of audio compression-decompression cycles are limited.

Emission grade systems are used when only one compression-expansion cycle, such as broadcast to the general public, is 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.

Quality vs Bit Rate

The ISO/MPEG (Musicam) design allows the digital bit rate, analog bandwidth, and quality to be generally related by the following formula:

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

According to this formula, 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 quality improves.

The ISO tests in Stockholm in May 1992 have demonstrated that at a digital bit rate of 256 kbps 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 kbps per stereo channel) was that MPEG Layer 2 is transparent. MPEG Layer 2 scored 5 on the Mean Opinion Score (MOS) scale, where the lowest is 1 and the highest is 5.

The qualities the ISO/MPEG (Musicam) Layer 2 algorithm provides at various bit rates is shown in Table 2-1.

Table 2-1. Quality Grades and Bit Rates

Quality	Bit Rate
contribution	384 kbps (stereo, Layer 2)
distribution	256 kbps (stereo, Layer 2)
emission	192 kbps (stereo, Layer 2A)
commentary	64 kbps (mono, Layer 2)

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

Tolerance to Transmission Errors

The ISO/MPEG (Musicam) 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 kbps per stereo channel, the length of a 24 ms frame is 6,144 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 ms of audio would be lost.

An error in the data part of the frame can range from imperceptible to just barely noticeable. This is due to the fact that 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 portion of the frame is unprotected, and any error occurring in the data part of the frame remains. The ISO/MPEG (Musicam) system has a defined error strategy. If an error is detected in the header, the last frame (24 ms) 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 error 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 the point of excessive error rates, the audio output mutes.

Tolerance to Multiple Processing

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

The quantizing noise raises with each encode/decode cycle and after a sufficient number of cycles, the noise level becomes perceptible. The degradation process is gradual and depends on the level of the quantizing noise on the original.

Table 2-2 lists the approximate number of total encode/decode cycles before the noise becomes significant.

Table 2-2. Bit Rate and Encode/Decode Cycles

Mode	Bit Rate	Number of Cycles
Mono	128 kbps	5
	64 kbps	1
Dual Mono (Stereo)	384 kbps	15
	256 kbps	5
Joint Stereo	192 kbps	2
	128 kbps	1

These are approximate and the exact number is highly dependent on the source material.

Postproduction Processing Effects

Modeling postproduction processing of compressed audio is complicated. For example, an equalizer changes the level of a range of frequencies, while limiting and compression are nonlinear processes. Very little test data is available to ascertain the effects of postprocessing. Private communications with the IRT suggest that ISO/MPEG (Musicam) Layer 2 is robust against the effects of postprocessing, and the degree of robustness depends on the compression rate. In particular, 384 kbps audio is unaffected by postprocessing, while 128 kbps audio is somewhat sensitive to postprocessing.

Transmission Channel Signal Format

The ComStream digital audio satellite broadcast system uses a single RF carrier to distribute audio, data, relays, and control information from the uplink to all downlinks. This multiservice data stream uses time division multiplexing (TDM) to receive, process, and transmit all these types of information.

Multiple audio channels can be transmitted using an RF carrier for each mono, dual-mono (stereo), or joint stereo channel service. This technique is known as frequency division multiplexing (FDM). Signal parameters between two RF carriers can be completely different. One carrier may be operating monaural audio at 64 kbps using Quadrature Phase Shift Keying (QPSK) modulation, and a second dual mono at 256 kbps using Bi-Phase Shift Keying (BPSK) modulation. Figure 2-4 illustrates various digital audio carriers that might occur on a single transponder.

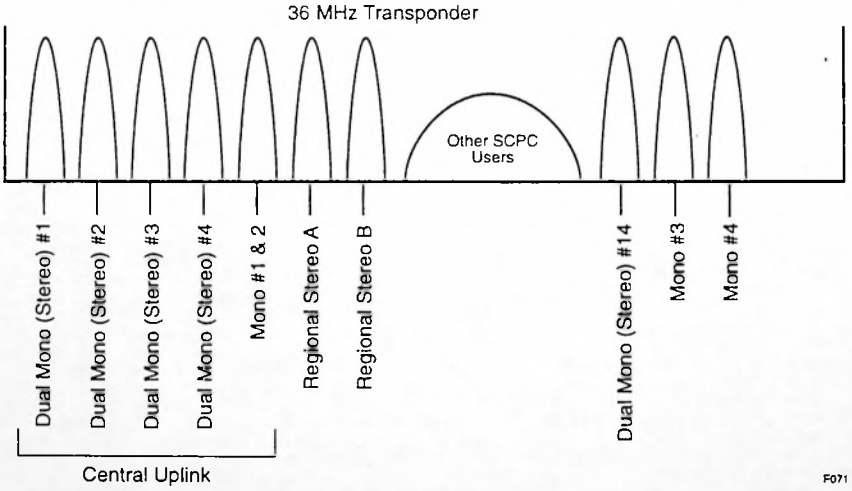


Figure 2-4. Multichannel Single Transponder System

The TDM frame structure is provided by the ISO/MPEG (Musicam) audio standard, as shown in Figure 2-5. The frame header provides all information regarding the audio rate, compression mode (mono, dual-mono (stereo), joint stereo), sampling rate (48 kHz), and ancillary data size. For error detection, a Cyclic Redundancy Check (CRC) checksum is included within the frame header.

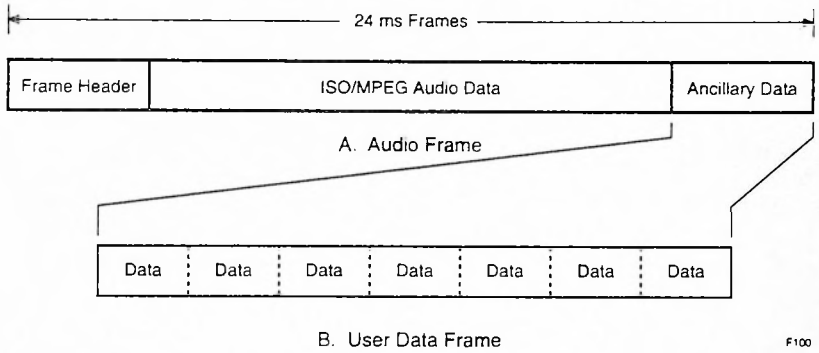


Figure 2-5. DAC700 Channel Format

The audio data is the digitally compressed data generated by the ISO/MPEG (Musicam) encoder section of the DAC700. The ancillary data stream comprises three types of control blocks:

- Network control
- User data
- Relay (cue) signaling

As shown in Figure 2-4, the ancillary data is embedded into the outbound digital stream that is then sent to the modem for the uplink. At the ABR receiver, the ancillary data is stripped out by the MPEG decoder and sent to a control microprocessor for additional processing.

Proper Signal Discrimination of Narrow Band RF Signals

The acquisition range of the ABR200/75 receiver works over a very large frequency range, approximately 2 MHz. With the possibility of having multiple carriers over that range, a means to uniquely identify one RF carrier from another RF carrier is needed to ensure the correct RF signal is received and processed as desired.

Discrimination is provided by the DAC700 when it embeds a unique identifier into the relay/control block of the ancillary data of the ISO/MPEG (Musicam) compressed audio output stream. This unique identifier has two components: a network ID and a channel ID. The programmable values for the IDs are generated by the DAC700 codec/mux at regular intervals (multiples of 100 ms).

For a given uplink, each codec/mux is programmed with a unique channel number, with each carrier typically having the same network ID. The network and channel IDs can be user-programmed during the DAC700's installation using the NI and CI commands, respectively. These commands are detailed in *Chapter 4: Remote Monitor and Control Operation*.

NOTE: For single uplinks it is recommended that, as a minimum, the network ID should be reprogrammed from its default value of 1.

Please note that the RF channel configuration at the uplink must match the downlink receiver's configuration or the receiver will not acquire and output the signal.

For example, if a channel ID other than 1 is used, say 16, then ensure that the downlink ABR receivers have a channel configuration #16 defined for the proper RF receiver frequency, symbol rate, and demodulation type (that is, CC 16,11700000,128000,1). If CC 0, CC 1, or any other channel configuration number other than CC 16 is programmed into the receivers, they will NOT acquire the channel.

For multiple carriers from a single uplink, a good practice is to keep the network ID the same for all muxes. The channel IDs should be consecutively ordered 1, 2, 3, and so on to correspond to the channel configuration numbers that are programmed into the downlink ABR units (CC 1, CC 2, CC 3, and so on).

Table 2-3 provides an example of these different configurations and their RF channel spacing (one carrier is incorrect). Assume QPSK operation and 128 kbps transmission rate.

Table 2-3. Channel Identifier and Frequency Allocation Combinations

Carrier No.	Uplink Site	RF Freq. MHz	NI	CI	CC/FC Format ABRs
A	X	11700.0	2	1	CC1,...FD 1,2,1,7
B	Y	11700.2	1	1	CC1,...FD 1,1,1,7
C	Z	11700.8	1	1	CC1,...FD 1,1,1,7
D	Z	11701.0	1	2	CC2,...FD 2,1,2,7 FD 3,1,2,5
E	Z	11701.2	1	3	CC2,...FD 4,1,3,7

NOTE: With this configuration of carrier frequencies and ID numbers, it is possible that carriers B and C will be incorrectly received, since they are within 600 kHz of each other and do not have unique ID numbers. All other carriers will operate properly.

DAC700 Major Functional Modules

Figure 2-6 is a block diagram of the DAC700. The major functional modules consist of:

- Audio input interfaces
- Audio output interfaces
- Broadcast data interface
- User data interface
- Passthrough interface
- Monitor and control interface (M&C)
- Relay/control interface
- Audio/data encoder
- Audio/data decoder
- Data multiplexer
- Unit controller
- Clock generator
- Power module

A brief description of each functional module is provided in the following paragraphs.

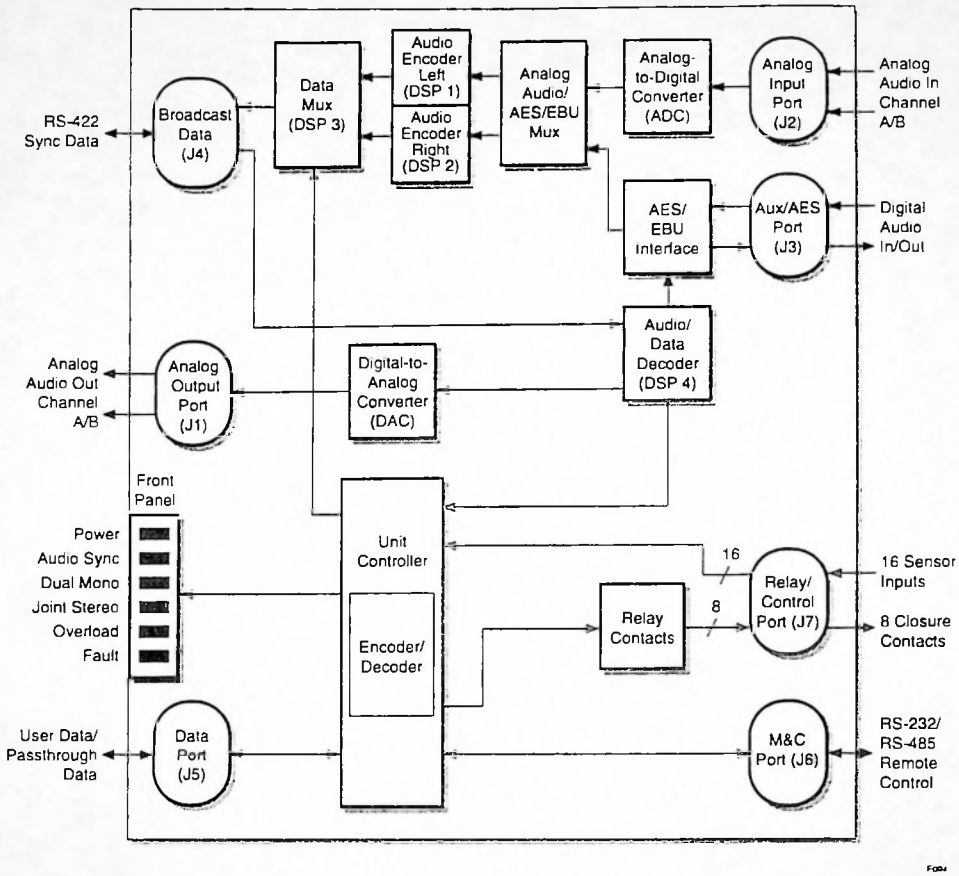


Figure 2-6. DAC700 Functional Block Diagram

Audio Input Interfaces

Analog audio is input as two channels (A/B) through the analog (audio) input port, as shown in Figure 2-7. The audio is buffered, amplified, and converted into 16 bit linear PCM data by the Analog-to-Digital Converter (ADC). The ADC is a dual channel oversampling device operated at a 48 kHz sample rate.

Predigitized AES/EBU audio may also be input through the AUX/AES port from an external device. Input AES/EBU data rates of 56 K and 112 K are not supported.

The AES/EBU serial audio data is converted into multiplexed 16 bit linear PCM data compatible with the ADC output.

The audio data is sent to the audio/data encoder (DSP 1/DSP 2). A 2:1 selector controls which audio data source is input: either via the ADC from analog (audio) input port or the AES/EBU interface from the AUX/AES port. Selector operation is controlled by the unit controller.

The audio/data encoder (DSP 1/DSP 2) converts the audio data to ISO/MPEG (Musicam) digital audio. The audio data then goes to the data multiplexer (DSP 3) where ancillary data is added to form a composite stream that is transmitted via the broadcast data port to the modem.

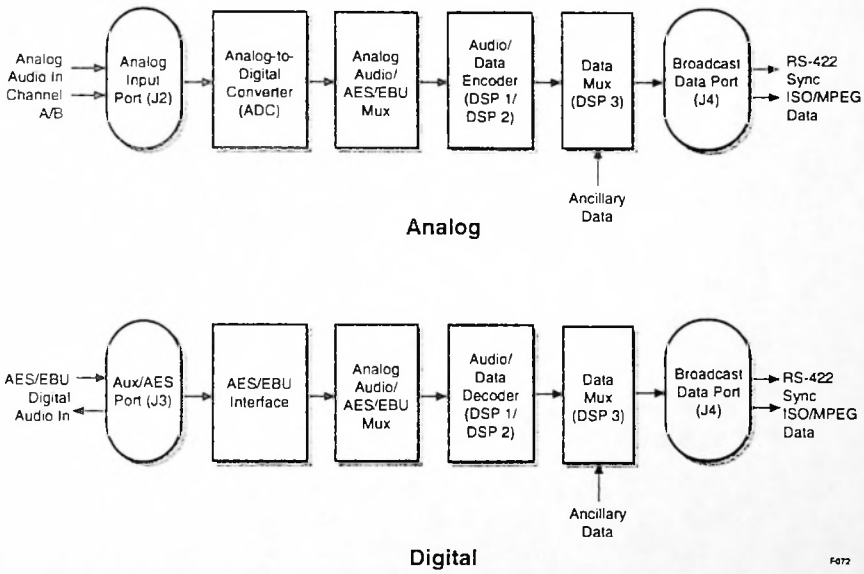


Figure 2-7. Audio Input Interfaces

Audio Output Interfaces

Transmitted data is received via the broadcast data port as shown in Figure 2-8. The audio/data decoder (DSP 4) separates the data into ISO/MPEG (Musicam) digital audio and ancillary data streams. The audio data stream goes to the Digital-to-Analog Converter (DAC).

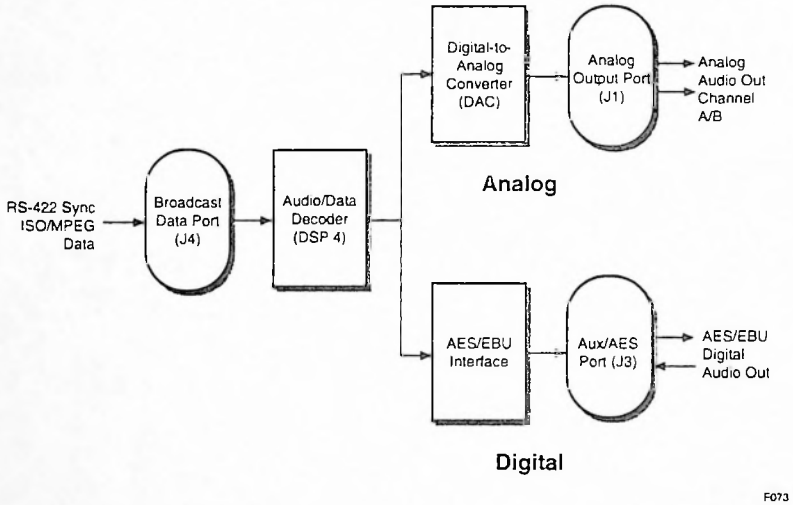


Figure 2-8. Audio Output Interfaces

The DAC is a dual channel, oversampling device operating at a 48 kHz sample rate. It converts the digital audio into analog audio that is transferred via the analog output port to the external sound equipment.

Digitized audio may also be output. The audio/data decoder converts the signal to 16 bit linear PCM data and sends it to the AES/EBU interface. The resulting AES/EBU audio data (in serial form) is output via the AUX/AES port. Both the analog audio and the AES/EBU digital format are output at the same time.

The appropriate transmit and receive timing signals for the audio input and output interfaces are provided by the clock generator module described later in this chapter.

Broadcast Data Interface

The broadcast data interface consists of a full duplex, synchronous RS-422 port. The transmit and receive data channels operate independently at rates from 56 to 384 kbps. The broadcast data port connects directly to the CM701 modem, or equivalent, for satellite transmission. The transmit and receive data rates are selectable via the M&C port.

Data Interfaces

The data interface consists of two asynchronous, RS-232 serial channels. Each serial channel operates independently. The audio data stream goes to the Digital-to-Analog Converter (DAC).

Data Channel 1

Data channel 1 is for user data. The DAC700 processes, formats, and transfers user data following the ComStream proprietary protocol.

The maximum user data rate depends on the system's data transfer specifications. User data rates are from 300 to 9600 bps. Since user data has the highest priority in the encoding process, at lower transmission rates (64 K, 96 K, and 112 K), the higher the user data rate, the more bits will be taken away from the audio content.

On the transmit side, user data is input to the unit controller, divided into blocks, and transferred in packet form to the data multiplexer (DSP 3) along with relay/control blocks and network control blocks. It multiplexes the ancillary data packets with the digital audio into a composite stream. Then this composite data is output via the broadcast data port, as shown in Figure 2-9.

The data block length can be user-defined with the BL command. Valid lengths are from 30 to 255 bytes with a system default of 26 bytes.

For more information on block types, refer to *Chapter 3: Installation and Initial Startup*.

NOTE: The ABR200/75 can only accept block lengths of up to 50 bytes.

Blocks are closed and transmitted based on one of the following:

- Maximum block length

  When the maximum block length is reached, the current block is transmitted via broadcast data port.

- Time-out

  If the maximum block length has not been reached in a given period of time, the block is closed and transmitted. The length of time is called the user data block time and is user-defined.

- Termination character

  The user may define a special end-of-block terminating character, which may also be sent with the user data or stripped off.

Data channel 1 features are programmed with the Data Port 1 Configuration (UC), User Block (UB), and User Termination (UT) commands. (See *Chapter 3: Installation and Initial Setup*).

On the receive side, composite data is demultiplexed by the audio/data decoder (DSP 4) into separate digital audio and ancillary data streams. The ancillary data is transferred to the unit controller where it is demultiplexed into user data, relay control messages, and network control commands and the user data is output via data channel 1.

Figure 2-9 diagrams user data processing.

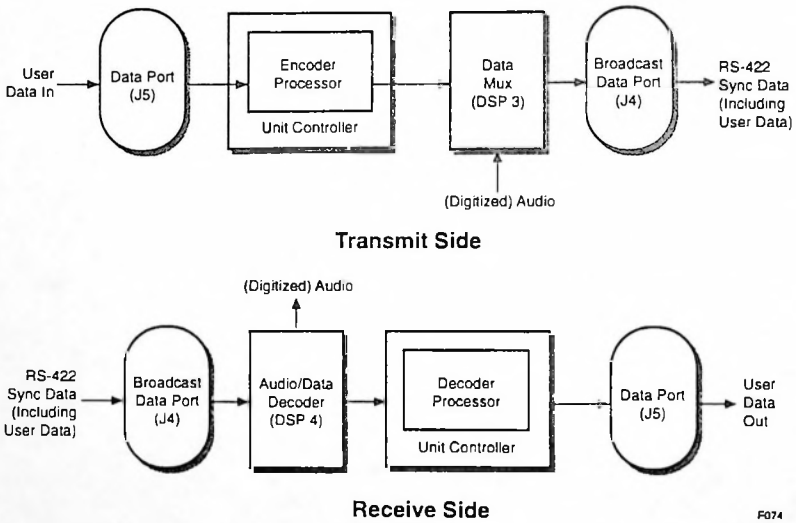


Figure 2-9. User Data Processing

Data Channel 2

Data port channel 2 allows the user to send "passthrough" data. The data rate is programmable and is embedded into the output ancillary data stream. The passthrough mode allows the user to transfer *preformatted* data to minimize the overhead of the onboard multiplexer. In this mode, an external source such as the ComStream Smart Node performs the formatting normally done internally by the DAC700. The passthrough mode can be selected at any time independent of user data, relay control messages, or control channel information. Passthrough data rates vary from 300 to 9600 bps.

NOTE: Passthrough data can be any of the allowable data types for the system: user data, relay control, and network control commands.

NOTE: It is not recommended to send a block type via data channel 1 when sending the same block type in the passthrough mode via channel two.

On the transmit side the passthrough data is input to the unit controller (encoder processor), as shown in Figure 2-10. The controller transfers the data in formatted form to the data multiplexer (DSP 3). The passthrough and ancillary data is multiplexed with the digital audio into a composite stream. Then the composite data is output via the broadcast data port and transmitted over the satellite in remote loopback mode.

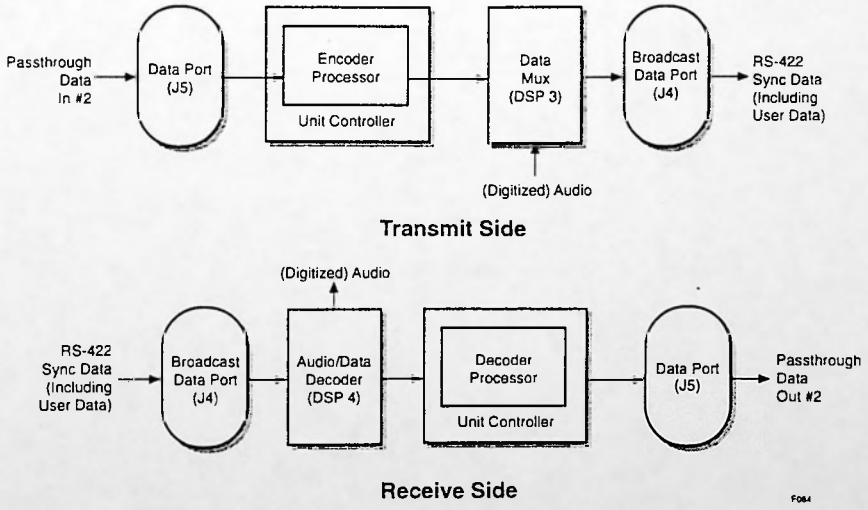


Figure 2-10. Data Passthrough Processing

The DAC700 transmits all blocks successfully received, based on ComStream proprietary protocol definition. Blocks not successfully received are discarded.

On the receive side, the loopback composite data is demultiplexed by the audio/data decoder (DSP 4) into separate digital audio and ancillary data streams. The ancillary data is transferred to the unit controller where it is demultiplexed and output via channel 2 of the data port.

NOTE: Only passthrough data will be accepted through channel 2.

Implementation

Two dual channel Universal Asynchronous Receiver/Transmitters (UARTs) are used to implement the data port. One UART is used for user data and a separate one is used for passthrough data.

Monitor and Control Interface

The monitor and control interface provides two ports for external monitoring and control of the DAC700 configuration and operation. This interface supports both RS-232 or RS-485 asynchronous channels via the M&C port.

The DAC700 connects via the M&C port to an external portable terminal, desktop computer, or terminal emulator. Control and status information from the remote computer is input to the unit controller (encoder/decoder processors), as shown in Figure 2-11.

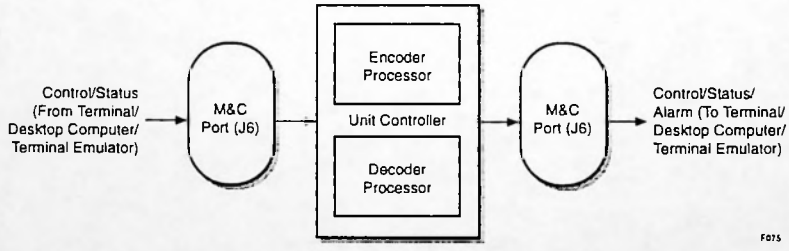


Figure 2-11. Monitor and Control Processing

Relay/Control Interface

The relay/control interface supports 16 TTL opto-isolated sensor inputs, that are used to create the relay control messages broadcast to the network in the ancillary data stream. These 16 relay inputs are mapped to eight output relays at each audio decoder within the network. This is on both the audio receivers and the DAC700. The relay contacts can be used to control external audio or other station equipment.

NOTE: Because only eight contact closures are available on each audio decoder, a mapping must be made as to which of the 16 possible control inputs activate the closures. For the DAC700 decoder, this mapping is performed by using the contact mapping (CM) command (see *Chapter 4: Remote Monitor and Control Operation*). The default for both the DAC700 and the ABR audio receiver decoders is to have the first eight inputs control the eight closures.

Figure 2-12 illustrates the 16-to-8 mapping function that occurs for the relay/control of the DAC700 decoder section.

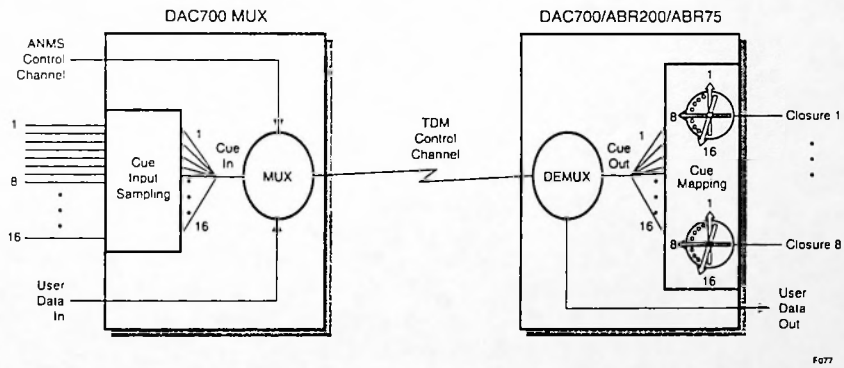


Figure 2-12. Contact Mapping Diagram

The sensor inputs operate at TTL logic levels, internally pulled up to +5 volts. The inputs become active by providing a dry contact closure to the signal ground on the relay/control port. Contact polarity can be set as either normally open or normally closed for each relay.

All inputs are sampled at a user-defined rate set using the RM command (see *Chapter 4: Remote Monitor and Control Operation*). Any change in the state of any input line is sensed within two sample periods and transmitted over the control channel.

The input levels are transmitted continuously over the control channel at a specified rate. The relay contact closures at the receiving end directly track the logic levels at the transmitting end. At the default sampling rates, pulsed signals are reproduced faithfully to the nearest 50 ms.

Audio/Data Encoder

The audio/data encoder (DSP 1/DSP 2) (Figure 2-6) consists of two DSPs that perform transmit audio ISO/MPEG (Musicam) processing. The first DSP handles the left channel and the second DSP handles the right channel of both the analog and digital audio input. These two DSPs convert the audio data into 16 bit linear PCM data and transfer it to the data multiplexer (DSP 3) for further processing via a shared memory architecture.

Data Multiplexer

The data multiplexer (DSP 3) (Figure 2-6) inputs and processes 16 bit linear PCM audio data and asynchronous ancillary data from the encoder processor. Data transfer is via a simplex interface that operates at a programmable rate set by the unit controller. The ancillary data is multiplexed along with the 16 bit linear PCM audio data into a composite data stream. Then the composite data is transferred synchronously via the RS-422 network interface for transmission by the CM701 modem over the satellite link.

Audio/Data Decoder

DSP 4 (Figure 2-6) performs the onboard ISO/MPEG (Musicam) decoding of the received (loopback or external) digital stream. A single DSP demultiplexes the incoming composite data stream into 16 bit linear PCM audio data and asynchronous ancillary data. The ancillary data is output to the unit controller via a simplex interface that defaults to 38.4 kbps. Then the decoded ISO/MPEG (Musicam) audio data is sent to both the DAC and AES/EBU interfaces. The audio is available at both the analog output port and the AUX/AES port.

ISO/MPEG (Musicam) data input to the decoder is selected from two sources:

- Broadcast data port during external baseband input operation
- Audio/data encoder when operating in PCM loopback or normal operation mode

Loopback testing operation allows the audio/data decoder to monitor the functional integrity of the encoder. Should the encoder function abnormally or fail, the decoder will lose frame synchronization.

Unit Controller

The unit controller provides centralized control of both the encoder and decoder processes. The controller utilizes a Motorola MC 68010 running at 12 MHz. Included with the microprocessor is 64 K RAM, 128 K EPROM, and 16 K battery-backed RAM within the DAC700.

Encoder Processor

The encoder processor of the unit controller has parallel read/write registers to monitor and control the analog and digital AES/EBU encoder. It also sends numerous event control and status bits throughout the system to select clock rates, operating modes, and device/circuit configurations.

Decoder Processor

The decoder processor of the unit controller receives the ancillary data from the decoder DSP. This information includes user data and network control messages. The information is handled in a manner identical to the encoder processor. The decoder processor also receives relay control data and controls the eight decoder output relay contacts.

Clock Generator

The clock generator module in Figure 2-13 produces all the clocks and event timing necessary for the system. Two separate clock generator circuits are provided to support independent, asynchronous transmit and receive operations. There are four selectable clock sources available to the user.

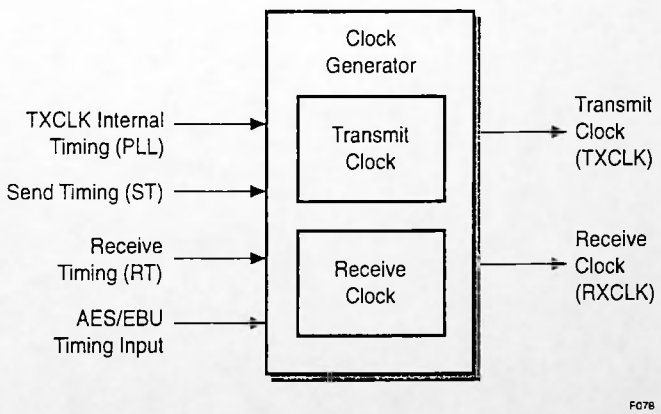


Figure 2-13. Clock Generator

The transmit clock generator circuit can derive clocking from one of four sources, shown in Table 2-4. The TT (Transmit Timing Selection) command selects which timing source to use. For more information on the TE command, refer to *Chapter 4: Remote Monitor and Control Operation*.

Table 2-4. Timing Source

TT	Timing Source	Figure
0	Internal 12.288 MHz crystal oscillator (PLL)	2-17
1	External from "send timing" (ST), RS-422 input **from** the broadcast data port (that is, from the modem send timing)	2-15
2	External from "receive timing" (RT), RS-422 input **from** the broadcast data port (that is, from the modem send timing)	2-16
3	AES/EBU (digital) clock input (External)	2-17

For internal timing (TT 0), the DAC700 uses its onboard 12.288 MHz PLL to generate the terminal timing that is sent to the modem with the transmit data. Figure 2-14 diagrams the internal timing source.

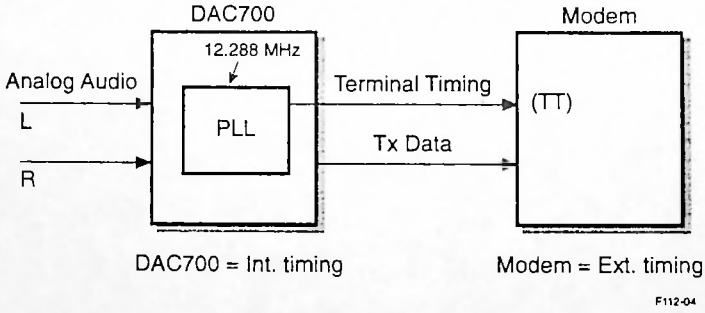


Figure 2-14. TXCLK Internal Timing (PLL)

For "send timing" (TT 1), the DAC700 uses the clock received from the modem to send back with the transmit data. This clock is derived from the modem's onboard PLL, as shown in Figure 2-15.

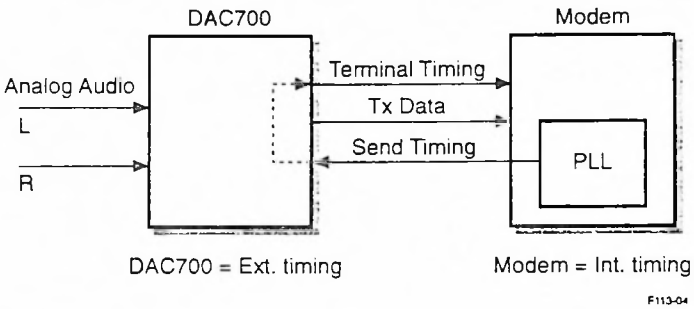


Figure 2-15. Send Timing

For "receive timing" (TT 2), the DAC700 uses the clock received from the modem to send back with the transmit data. This clock is derived from the active demodulator on the modem, as shown in Figure 2-16.

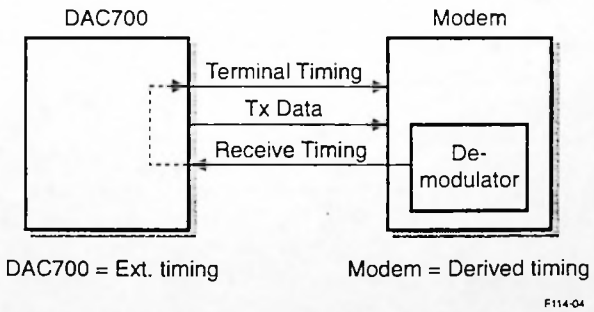


Figure 2-16. Receive Timing

For AES/EBU timing (TT 3), the DAC700 derives the clock from the AES/EBU digital input data stream and sends this to the modem with the transmit data, as shown in Figure 2-17.

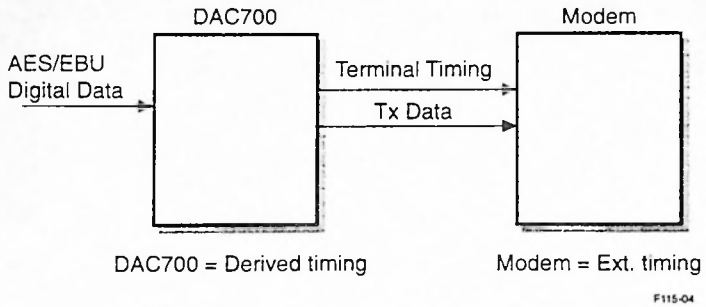


Figure 2-17. AES/EBU Timing Input

NOTE: If the AES/EBU stream is not present, the DAC700 indicates a fault and stops sending a clock to the modem.

Protected Mode

While in LB 2, the protected mode of the DAC700 controls the terminal transmit timing clock (TTCLK). This is based on the frame synchronization status of the audio/digital decoder within the DAC700. If the decoder is synchronized, then the TTCLK is output at its prescribed rate. If the audio/digital decoder loses synchronization, then the TTCLK is held low until it is restored. A delay of up to several seconds is provided to prevent race conditions with external transmission equipment. The unit controller enables the TTCLK and determines the delay value. The TE (Set Protected Mode) command enables or disables this feature.

Power Module

An autoswitching, universal input voltage power supply provides all internal DC voltages in the unit.

Sources for Decoder Input

There are two possible sources of input to the onboard ISO/MPEG (Musicam) decoder. The Audio Sync indicator shows when the decoder has ISO/MPEG (Musicam) frame synchronization but does not show the source of the signal. There are three defined loopback modes, LB 0, LB 1, and LB 2, that are set or read using the LB command.

Following is a description of each loopback mode. For more information on the LB command, refer to *Chapter 4: Remote Monitor and Control Operation*.

LB 0—External Baseband Input Operation

In LB 0 an ISO/MPEG (Musicam) stream from an external source is brought in through the broadcast data port and then decoded, as shown in Figure 2-18. This mode can be used to monitor the uplink, since you are using the DAC700 as a decoder only.

NOTE: An independent ISO/MPEG (Musicam) stream can also be sent out the broadcast data port.

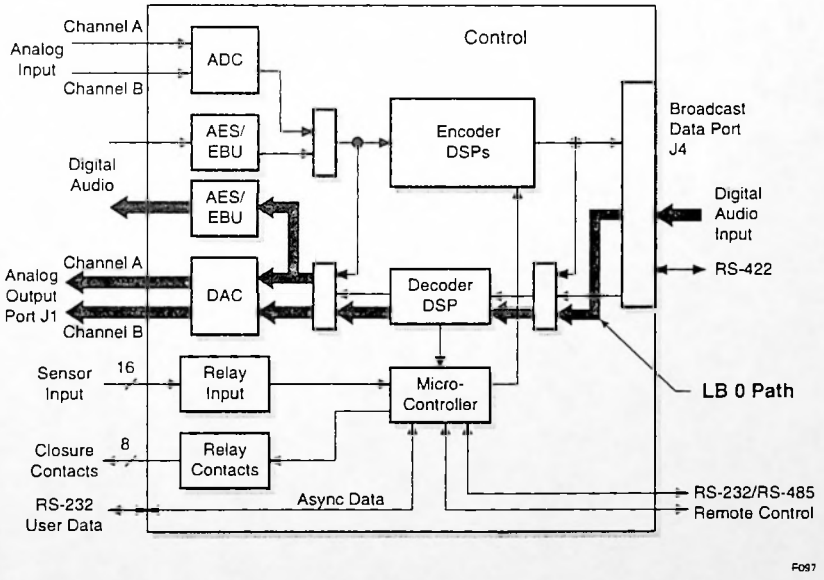


Figure 2-18. ISO/MPEG (Musicam) Codec DAC700 Block Diagram for LB 0

LB 1—PCM Loopback Operation

In LB 1 the data is looped from the output ADC through the DAC, as shown in Figure 2-19, bypassing the DSP encoder/decoder. This mode can be used to verify the ADC performance while bypassing the algorithm.

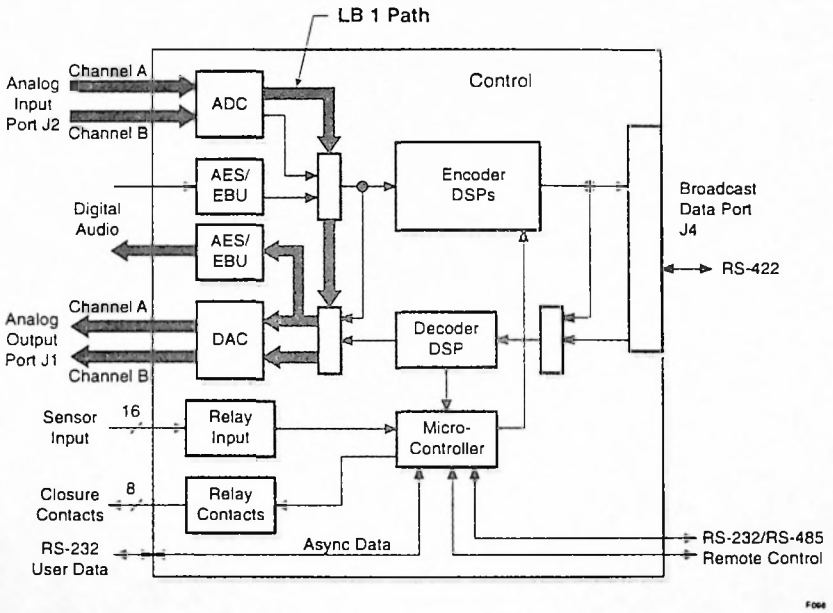


Figure 2-19. ISO/MPEG (Musicam) Codec DAC700 Block Diagram for LB 1

LB 2—Normal Operation

In LB 2 the data is received from the analog input port as shown in Figure 2-20, encoded, and then sent out the broadcast data port. From the encoder, data is also sent to the decoder. This mode is primarily used for normal system operation.

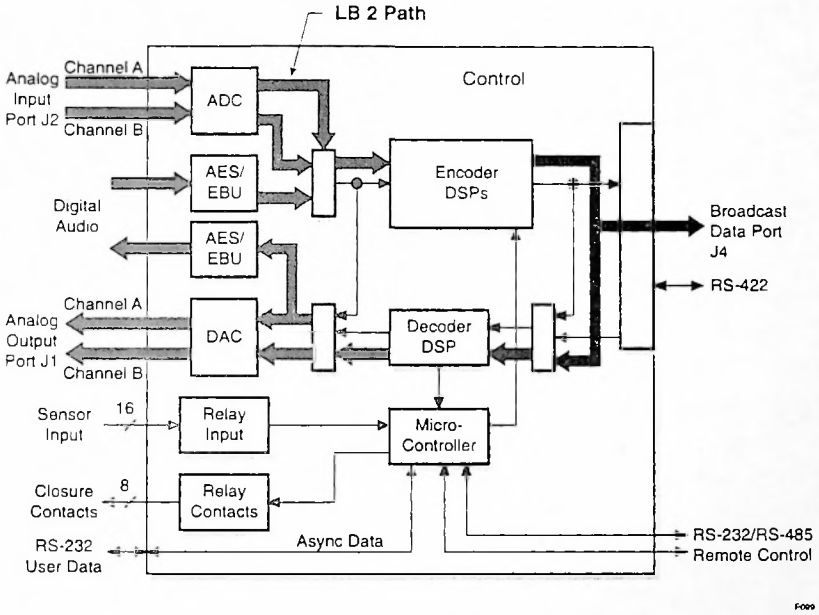


Figure 2-20. ISO/MPEG (Musicam) Codec DAC700 Block Diagram for LB 2

ComStream
A Spar Company

Chapter 3: Installation and Initial Startup

Introduction

This chapter describes the steps to install and initialize the DAC700. It also describes the initial configuration procedures as part of the installation process.

Overview

The overall steps for installing and starting up the DAC700 are shown in Table 3-1. The table also shows the recommended installation sequence. You should become familiar with this sequence before installing the unit. Be aware of any unusual conditions or contingencies that might affect the installation process.

Table 3-1. Installation Sequence

Sequence
Site planning and preparation
Location requirements
Safety considerations
External connections
Initial power-up
Installation check
Firmware version
Initial configuration
Audio bandwidth configuration
User data or passthrough mode configuration
Encoder/decoder ancillary data configuration
User relay configuration
Network and channel definitions
Timing source configuration

Site Planning and Preparation

The purpose of site planning is to specify where the components are to be located and to identify any special installation or operational requirements. Time spent in site planning prevents unnecessary complications during installation.

Location Requirements

Location requirements for the DAC700 unit are:

- Standard 19 in equipment rack, one unit (1.75 in) high
- Adequate ventilation

  Mount the DAC700 in a manner that ensures adequate ventilation at all times. The minimum airflow clearance space required is three inches at the sides and six inches in the rear.
- Facility AC line power as follows:
  - 90 VAC to 264 VAC
  - 47 to 63 Hz
  - 40 w service with circuit breaker and integral ground protection
  - one or more lightning and surge protectors
  - cable run to AC power connector on rear of unit with 3 ft (1 m) relief length

  ComStream recommends an uninterruptible power supply (UPS), a line conditioner, or both where power is subject to surges, unstable or noisy conditions, or outages.
- Adequate distance between the DAC700 unit and electromagnetic interference (EMI) sources such as AC power generating equipment
- Proximity of 25 ft (7.62 m) or less to equipment providing specified signal I/O levels for RS-422, RS-232, or optional RS-485 interfaces.

Mounting the DAC700

The DAC700 works in concert with the CM701 modem. The DAC700 has been thermally designed to be placed in a 19" rack in a pattern of alternating DAC700s and CM701s (CM701/DAC700/CM701/DAC700 and so on).

If the DAC700 is not used with the CM701, it is recommended that one unit of rack space be left open above each DAC700 unit to ensure adequate air flow. The rack must be ventilated so that the internal ambient temperature does not exceed the DAC700's maximum ambient temperature of 50°C.

Safety Considerations

Carefully read and follow all safety, use, and operating instructions before operating the DAC700. Heed all warnings and cautions contained in this manual. Retain these instructions for future reference.

FOLLOW STARTUP PROCEDURE

Do not plug in the DAC700 until you have connected the system and read this chapter.

ROUTE POWER CORDS SAFELY

Route power cords so they are not walked on or pinched. Pay particular attention to cords and connections at the plugs, receptacles (such as power strips), and the point where they exit from the DAC700 and attach to other equipment. Do not place any items on or against power cords.

NO STACKING

Do not place or stack any objects on top of the DAC700. Other equipment may be placed in a rack or on a shelf above or below the DAC700, but never stacked directly on top of the encoder itself.

PROTECT AGAINST LIGHTNING AND POWER SURGES

When the satellite communication equipment is installed, have the professional installer ground the system to protect against voltage surges and built-up static charges.

KEEP OBJECTS OUTSIDE

Touching internal DAC700 parts is dangerous to both you and the unit. Never put any object, including your fingers, through DAC700 slots or openings, as this could result in touching dangerous voltage points, short-circuiting parts, electric shock, or fire.

If an object falls into the DAC700, unplug the unit and contact ComStream Customer Service, as serious damage could occur to the DAC700 or its components.

SERVICE THE DAC700

Do not attempt to service the DAC700 yourself, as there are no user-serviceable parts inside. Opening or removing covers may expose you to dangerous voltages or other hazards, as well as void your warranty.

If you need service, contact ComStream Customer Service to obtain qualified service personnel.

PERFORM SAFETY CHECKS

Upon completion of any service or repairs to the DAC700 or its option cards, ask the service technician to perform safety checks to verify that the system is in safe operating condition.

External Connections

This section describes the physical and electrical connections to the DAC700.

CAUTION: Ensure that ALL cables are installed before connecting to an AC power source.

All external connections to the DAC700 are made through the rear panel connectors. The DAC700 has eight possible connections, including the power plug, on the rear panel. The location of these connectors is shown in Figure 3-1.

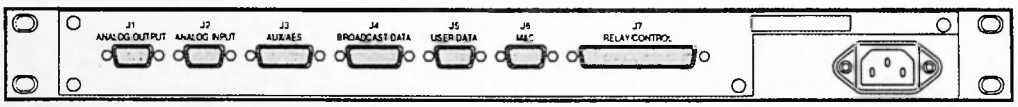


Figure 3-1. DAC700 Rear Panel Connectors

The pinouts for these interfaces are detailed in *Appendix A: Interface Pinouts.*

NOTE: To ensure compliance with emission standards, all signal cables connected to the receiver should be shielded. The shield must be electrically attached to the mating connector.

Analog Output

The analog output port has two balanced output channels from the decoder: one right channel and one left channel.

NOTE: Ensure that phase alignment is maintained between studio equipment and the encoder. If a phasing error occurs when operating in the joint stereo mode, common mode (background) noise may be accentuated.

Analog Input

This port has two balanced input channels for the encoder: one right and one left audio channel. The maximum input level before distortion occurs is +18 dBu (dBm). Above this level, the DAC700 hard clips the signal.

Audio input impedance is configured internally for >100 Kohm audio input impedance at the factory. An optional cable (ComStream Part No. 05-0548-001) can be used to configure it for 600 ohm balanced operation.

AUX/AES

This port provides two digitized AES/EBU audio channels: an *input* channel *to* the encoder and an *output* channel *from* the decoder. Both the input and output channels are serial digital with balanced line, transformer-coupled interfaces.

The AUX/AES port also provides three preset inputs that can be programmed to automatically switch the DAC to operate in any of eight predetermined formats. See *Chapter 4: Remote Monitor and Control Operation.*

In addition, two status closure contacts provide codec alarm relay fault status, and the remote control reset initializes the M&C port in the default configuration.

Broadcast Data

The broadcast data port provides the composite audio and multiplexed data to and from the satellite modem. It is a full duplex, synchronous, RS-422 interface.

The transmit and receive channels operate independently at data rates from 56 to 384 kbps. Transmit timing for data output may be derived from four sources. Receive timing for data input may be derived from two sources. See *Chapter 2: Functional Description and Theory of Operation* for timing definitions.

User Data

The port consists of two separate, asynchronous, RS-232 serial ports. Each serial channel operates independently. Rates can be selected via the M&C port. Data 1 channel provides transmit direction hardware flow control and operates at rates from 300 to 9600 bps. Data 2 channel is used as an input/output path for the data passthrough mode and operates at rates from 300 to 9600 bps.

M&C

The M&C port can be used to connect the DAC700 unit to an external portable terminal or desktop computer. The computer is used to issue user control, configuration, and diagnostic commands to the unit. The computer can also be used to monitor unit operating status, and fault and alarm information.

The port supports a standard RS-232 interface or an optional RS-485 interface:

- The standard RS-232 interface provides duplex, asynchronous, serial data with an operating range from 300, 1200, 2400, 4800, 9600 to 19200 bps. The defaults for RS-232 interface are 1200 bps, odd parity, 7 data bits, and 1 stop bit.
- The RS-485 interface provides duplex, asynchronous, multidrop packet data configured for party-line operation. This port is used in conjunction with the remote control computer. The operating rates are 300, 1200, 2400, 4800, 9600, or 19200. The RS-485 interface defaults are 2400 bps, odd parity, 7 data bits, and 1 stop bit.

NOTE: A proprietary ComStream packet protocol is used for communicating via this interface. See *Chapter 5: ComStream Packet Protocol Overview*.

The default RS-485 address of the M&C port on the DAC700 is 31, but it can be configured to any bus address from 1 to 31 using the RA command. See *Chapter 4: Remote Monitor and Control Operation*.

Relay/Control

The relay/control port is a multifunction interface that supports 16 TTL opto-isolated sensor inputs and eight relay contact outputs. The 16 TTL sensor inputs go to the unit controller for formatting into relay control messages. In turn, these messages are sent through the system's ancillary data channel to directly control an external control console or event sequencer connected to the relay output port of the receivers.

The sensor inputs operate at TTL logic levels, internally pulled up to +5 v. The inputs become active by providing a dry contact closure to the signal ground on the relay/control port.

Power Cord

For international applications, use an unterminated HAR-approved (up to 250 VAC at 6 A) AC power cord with an IEC-compatible female plug on one end and three stripped and tinned bare wires on the other end. This cord complies with the following international color code:

Line	Brown
Neutral	Blue
Ground	Green/Yellow

If an unterminated power cord is supplied with the unit, the appropriate certified termination plug must be installed. Table 3-2 lists the required certifying agencies for some countries.

If the color code described does not correspond to the colored markings identifying the terminals on the plug you are installing, do the following:

- Connect the green and yellow wire to the terminal in the plug marked by the letter E or by the earth symbol (⏚), or colored green and yellow.
- Connect the blue wire to the terminal marked with the letter N or colored black.
- Connect the brown wire to the terminal marked with the letter L or colored red.

Refer to Table 3-2 for a list of the required certifying agencies for various countries.

Table 3-2. International Certifying Agencies

Country	Agency
Australia	SSA
Austria	OVE
Belgium	CEBEC
Canada	CSA
Denmark	DEMKO
Finland	FEI
France	UTE
Germany	VDE
India	ISI
Ireland	IIRS
Italy	IMQ
Japan	MITI
Netherlands	KEMA
New Zealand	SECV SECQ SECWA EANSW ETSA HECT SANZ
Norway	NEMKO
Republic of So. Africa	SABS
Spain	AEE
Sweden	SEMKO
Switzerland	SEV
United Kingdom	ASTA BSI

Initial Power-up

This section describes the steps to complete the initial AC power-up of the DAC700 unit.

To power up the DAC700:

1. Make sure the DAC700 is properly installed in the equipment rack and the system cables are properly connected to the unit.

NOTE: Ensure that ALL cables are installed before connecting to an AC power source.

2. Turn on power for the remote control terminal, desktop computer, or terminal emulator and wait for it to initialize.

   Make sure the computer COM port provides RS-232 communication and is connected to the unit M&C port. Configure the port for 2400 baud, 7 data bits, 1 stop bit, and odd parity.

3. Turn on the unit by connecting the power cord to the AC power jack on the unit.

When the power-up initialization cycle has been completed, as a minimum the green power LED will be illuminated. Other LEDs may also be illuminated depending on your system configuration.

Installation Check

Once the DAC700 has been powered up, verify the unit connections for the audio, user data, and relay/control ports. This can be accomplished by communicating with a receiver locked to the uplink RF carrier or using an ASCII terminal to query the decoder section of the DAC700 to confirm the setup and perform several diagnostic tasks.

Use the DB-9-to-DB-25 adapter cable (ComStream Part No. 03-0120-093) to connect the DAC700's M&C port to the DB-25 connector on your terminal. The terminal should be configured for 2400 baud, 7 data bits, 1 stop bit, and odd parity.

NOTE: When communication is successfully accomplished, a > prompt appears on the terminal, indicating it is ready to accept commands.

To establish communication with the DAC:

- At the ASCII terminal connected to the DAC700, press the ENTER key on the terminal.

  The DAC700 should respond with an ASCII login request string.

NOTE: Refer to *Chapter 4: Remote Monitor and Control Operation* for an alphabetical listing of commands with proper command syntax, valid parameters, and default values.

Once the communications link with the DAC700 is established, use the following steps to perform the installation check:

1. Using a terminal emulation program, verify that the default values are correct by entering the appropriate command for each value. For a list the commands and their correct values, see Table 4-4 in *Chapter 4: Remote Monitor and Control Operation.*
2. Test for fault-free operation.
   a. Using the terminal emulation program, enter **`LB 1`** to put the unit in PCM loopback mode. Allow the unit to operate in loopback for 5 to 10 minutes error free.

NOTE: There are also two other loopback modes available. Select the appropriate mode for your testing. (See *Chapter 2: Functional Description and Theory of Operation.*)

---

   b. After operating in loopback mode, enter **`FL`** and check for any fault codes.
      - If no faults are displayed, enter **`LB 0`** to put the unit into normal operating mode.
      - If any faults are displayed, refer to *Chapter 6: Maintenance and Troubleshooting.*
3. Check audio performance.
   a. Using the terminal emulation program, enter the **`AT`** command with the appropriate parameter.

      The DAC700 provides two internally generated audio test tones for checking audio performance and the option of checking the left channel, the right channel, or both. (See *Chapter 4: Remote Monitor and Control Operation.*)

   b. After checking audio performance, enter **`AT 0`** to return to normal operation.

Firmware Version

Check the firmware version number as part of initial installation. The number should be the same as all other stations in the network. Different version numbers indicate possible software incompatibility and might result in unusual or intermittent operation.

1. Enter **DC**.
2. Verify that the firmware version number matches the version number of other stations in the network.

If the number does not match, notify the network operator or ComStream Customer Service as soon as possible.

Initial Configuration

Specify the initial configuration parameters for the DAC700 using the procedures that follow. Use only the procedure or procedures that support your hardware configuration and the operating mode or modes you intend to use in the network.

The configuration process requires you to use a number of commands. This section gives the general syntax for commands used during configuration. *Chapter 4: Remote Monitor and Control Operation* provides an alphabetical listing of all commands with a full description of the command, command syntax, valid parameters, and default values.

NOTE: Read the command description in *Chapter 4: Remote Monitor and Control Operation* before using the command to configure the DAC700.

Audio Bandwidth Configuration

The audio bandwidth configuration depends on three factors:

- Transmission rate
- Operating mode (mono, dual mono (stereo), joint stereo)
- User-selectable bandwidth

For a table giving the relationship between these factors, see Table 7-1 in *Chapter 7: Technical Specifications and Port Information*.

Using Table 7-1 in *Chapter 7: Technical Specifications and Port Information*, select the transmission rate, operating mode, and bandwidth for the audio configuration of your network.

Use the following steps to configure the unit audio bandwidth to match your selections.

1. Set the encoder analog bandwidth. Command format is:

   AB 0 {to select bandwidth A}

   AB 1 {to select bandwidth B}

   AB 2 {to select bandwidth C}

   AB 3 {to select bandwidth D}

2. Set the encoder transmit mode. Command format is:

   TM 0 {to select mono}

   TM 1 {to select dual mono}

   TM 2 {to select joint stereo}

3. Set the encoder transmit rate. Command format is:

   TD *transmission_rate*

   Replace *transmission_rate* with 56000 to select 56 K, 64000 to select 64 K, and so on.

4. Set the decoder receive channel rate. Command format is:

   RD *receive_channel_rate*

   The value for *receive_channel_rate* should be the same as for *transmission_rate*.

User Data Configuration

Use the following steps to configure the user data feature.

1. Configure data port 1. Command format is:

   UC *baud,parity,data_bits,stop_bits*

2. Set the user data time or send interval. Command format is:

   UB *time_interval*

3. Set the user data termination character. Command format is:

   UT *termination_character*

   Replace *termination_character* with the decimal value of the ASCII character to be used.

Passthrough Mode Configuration

Use the following steps to configure the passthrough mode.

1. Configure data port 2. Command format is:

   PP *baud,parity,data_bits,stop_bits*

2. Enable the data passthrough mode. Use the PT command as shown:

   PT 1 {to enable data port 2}

   PT 0 {to disable data port 2}

Encoder/Decoder Ancillary Data Rate Configuration

Ancillary data is made up of three components: relay blocks, network control message blocks, and user data blocks. Each of these information types has a specific packet structure or format.

- Relay blocks include the input relay information, the network ID, and channel ID information. These blocks have the highest priority of the three. Relay blocks are sent out at user-programmable intervals of multiples of 100 ms. The length of the interval is programmed in the DAC700 with the RM command. (See *Chapter 4: Remote Monitor and Control Operation.*)
- Network control message blocks contain commands generated by the ANMS and sent to the receivers through the satellite transmission. Network control messages have the second highest priority.
- User data blocks are originated by the user and input into the DAC700 for broadcast transmission to all receivers.

The ancillary data rate is the "bandwidth" set aside for this information and is programmed using the AG command. The bandwidth is dynamic and is only as wide as it needs to be. The default is 38.4 kbps.

NOTE: The rule of thumb is to minimize the user data rate as much as possible. Because user data can vary from 300 to 9600 bps, at lower transmission rates—112 kbps down to 56 kbps—audio degradation may occur if user data exceeds roughly 5% of the bandwidth.

NOTE: Judging the effect that data bandwidth has on music quality is very subjective. The guideline given in this manual is a recommendation only.

To configure encoder/decoder ancillary data:

1. Set the ancillary data rate for the encoder/decoder. Command syntax is:

   AG *baud,parity,data_bits,stop_bits*

2. Set the mux/demux mode. Command format is:

   DM 0 {for simplex mode}

   DM 1 {for full duplex mode}

For a definition of simplex and duplex modes, refer to *Chapter 4: Remote Monitor and Control Operation* .

User Relay Configuration

Configure user relays to provide control of cueing or event operations for a studio console, event sequencer, or ABR receiver. For example, user relays can turn specific receivers on at a set time.

Use the following steps to configure these relays.

1. Set the contact sense polarity for the eight relay channels. Command syntax is:

   CS *relay_1,relay_2,relay_3,relay_4,relay_5, relay_6,relay_7,relay_8*

   For each instance of *relay*, substitute:

   0 {relay normally open}

   1 {relay normally closed}

NOTE: This is a DAC700 decoder function only.

2. Map transmit relays to receive relays. Command syntax is:

   CM *relay_1,relay_2,relay_3,relay_4,relay_5 relay_6,relay_7,relay_8*

NOTE: This is a DAC700 decoder function only.

3. Define the relay message interval. Command syntax is:

   RM *interval*

   Interval is the time between relay control message updates.

4. Configure the relay input port. Command syntax is:

   XM *active_input_mask,polarity_mask,debounce*

Network and Channel Definitions

Define channel and network ID numbers as follows:

1. Define channel ID numbers. Command syntax is:

   CI *n*

2. Define network ID numbers. Command syntax is:

   NI *n*

Timing Source Configuration

The DAC700 uses the timing reference to support independent, asynchronous transmit and receive operations. The correct timing source is essential for accurate unit internal timing and external interface timing. For more information on timing sources, see *Chapter 2: Functional Description and Theory of Operation.*

To select the timing source, use the TT command as shown below:

TT 0 {to select internal—12.288 MHz crystal oscillator}

TT 1 {to select external—from send timing}

TT 2 {to select external—from receive timing}

TT 3 {to select AES/EBU timing input}

ComStream
A Spar Company

Chapter 4: Remote Monitor and Control Operation

Overview

The DAC700 M&C port provides a means to configure and monitor the status of the entire ISO/MPEG (Musicam) audio codec/multiplexer unit. The unit is capable of communicating in RS-232 ASCII message format and in a multidrop RS-485 network packet protocol format. When operating in the RS-485 packet protocol format, network control messages can also be communicated through this port.

The command formats are compatible with existing ComStream command formats. However, for RS-232 operation, there is no need for an ESCAPE character to precede the command. If an ESCAPE character is input by the user, the ESCAPE character is ignored and the command executes as entered.

Command Syntax

For RS-232 ASCII commands, the following generic formats are used.

- To write or change a parameter associated with the command designated by a two-character mnemonic, XX, the formats are:

  XX parameter

  XX parameter 1,parameter 2,parameter 3

  Separate multiple parameters with a comma only; do NOT add spaces.

- To read or query a parameter value for command, the format is:

  XX ?

  The space before the question mark is optional.

- To repeat the previous command, use a carriage return only.
- To access help, press ?. Help provides a summary of available commands with a short explanation of each.

Commands may be entered in either upper- or lowercase. All commands are followed by a carriage return.

Command Error Codes

Command errors occur when a command has been mistyped, is inappropriate, or cannot be immediately executed. The operator normally receives one of the error messages shown in Table 4-1.

Table 4-1. Error Codes and Descriptions

RS-232 Mode	RS-485 Mode	Description
Invalid Command	ER1	Indicates that an invalid command has been entered
Invalid Parameter	ER2	Indicates that the associated parameter is outside the expected range
Not Available	ER3	Indicates that the command is currently not available (for example, a function is not operating that would provide the expected results)

Command Summary

Table 4-2 displays the DAC700 commands in a summary form. Further details are provided in the next section.

Table 4-2. DAC700 Command Summary

Command	Description
AB	Analog bandwidth
AG	Aggregate (ancillary data rate)
AM	Alarm mask for status relay activation
AS	Algorithm select
AT*	Audio test tone mode
BL	Block length
CF	Clear fault accumulation
CI	Sets channel ID for relay control messages
CM	Contact map definition for local relay closures
CO	Contact control for local relay closures
CQ	Contact status query for local relay closures
CS	Contact sense for local relay closures
DC	Display configuration
DD	Selects decoder ancillary data rate
DM	Duplex mode
FL	Accumulated fault status
IN	Master reset, reinitializes unit to default values
LB	Loopback mode selection
LR*	Left/right channel toggle command
MU*	Audio mute command
NI	Sets network ID for relay control messages
PC	Sets RS-232 M&C port configuration
PD	Preset definition for redundancy protection
PM	Set preset mode
PP	Passthrough port configuration
PS	Preset select
PT	Passthrough mode
RA	Sets RS-485 remote control device address
RC	Sets RS-485 M&C (remote control) port configuration
RD	Receive channel data rate
RE	Resets the unit
RM	Selects relay message interval
SP	Set protection mode
ST	Real-time fault status
TD	Transmit channel data rate
TM	Select Musicam encoder (transmitter) mode
TT	Select transmit timing clock source
UB	User data send interval
UC	User data port configuration
UT	User termination
XM	Relay input mapping

* AT, LR, and MU affect only DAC700 Decoder Audio Function.

Detailed Command Descriptions

Following is a list of commands with a detailed description of each used to configure and use the DAC700. The items shown in italics are variable characters to be replaced by the specific parameter information being entered. Because of space constraints, defaults sometimes appear on more than one line. In actual use, these defaults would be on one line separated by commas.

Table 4-3. Alphabetical Command List

Command	Function	Syntax/Description	Default
AB	Analog Bandwidth	AB *n* This command sets audio bandwidth. *n* specifies bandwidth. Valid values for *n* are: ***n*** **Bandwidth** 0 Bandwidth A 1 Bandwidth B 2 Bandwidth C 3 Bandwidth D The selected bandwidth is dependent on transmission rate (TD) and operating mode (TM). For the relationship between bandwidth, transmission rate, and operating mode seeTable 7-1 in *Chapter 7: Technical Specifications and Port Information.*	0
AG	Aggregate Encoder-decoder Ancillary Data Rate	AG *baud,parity,data_bits,stop_bits* AG ? This command sets or reads the encoder/decoder ancillary data rate. The valid values for the parameters are: **Parameter** **Valid Values** Baud 1200, 2400, 4800, 9600,19200, 38400 Parity N-none Data bits 8 Stop bits 2 Defaults 38400,N,8,2	38400,N,8,2
AM	Alarm Mask	AM *n* AM ? This command sets or reads the alarm relay mask. *n* is the decimal number that represents the bit map of the faults to be monitored by the status relay as shown below. **Fault Number** **Fault Condition** **Decimal Weight** 1 TX PLL not locked 1 2 RX PLL not locked 2 3 Decoder not framed (out of sync) 4 4 Audio Overload, Left Channel 8 5 Audio Overload, Right Channel 16 6 Network or channel ID fault 32	255

Table 4-3. Alphabetical Command List (continued)

Command	Function	Syntax/Description	Default
AS	Algorithm Select	AS *n* AS ? This command selects or reads the algorithm to either ISO/MPEG or Musicam. The valid values for *n* are: ***n*** **Algorithm** 0 Musicam 1 ISO/MPEG	0
AT	Audio Test Tone Mode	AT *n* AT ? This command sets or reads the decoder tone generation mode. *n* specifies the channel or channels over which the tone is generated and the tone frequency. An *n* of 1 to 6 generates a tone continuously until the user stops it by entering AT 0. The valid values for *n* are: ***n*** **Tone generation** 0 Normal operation 1 Left audio channel, 1,000 Hz tone 2 Right audio channel, 1,000 Hz tone 3 Both audio channels, 1,000 Hz tone 5 Left audio channel, 9,600 Hz tone 6 Right audio channel, 9,600 Hz tone 7 Both audio channels, 9,600 Hz tone	0
BL	Block Length	BL *n* BL ? Sets or reads the maximum block length used for the protocol of the ancillary data where *n* is the length of the block in bytes. Valid values are 30 to 255. NOTE: ABR200/75 receivers can only accept block lengths of up to 50 bytes.	30
CF	Clear Fault	CF *n* CF ? The clear fault command clears the fault specified by *n* in the fault history register. If *n* is zero (0), all faults are cleared.	0
CI	Channel ID	CI *n* CI ? Sets or reads the channel ID number that is sent out as part of the relay control message. *Chapter 2: Functional Description and Theory of Operation* provides a description of the channel ID. Valid range is 1 through 255.	1

Table 4-3. Alphabetical Command List (continued)

<table>
<tr><th>Command</th><th>Function</th><th>Syntax/Description</th><th>Default</th></tr>
<tr><td>CM</td><td>Contact Map</td><td>CM map values
CM ?
This command allows the codec to be configured to provide a mapping of relay contacts at the transmitting codec to relay contacts at all receiving decoders, both internal and over the satellite network.
The map is a series of eight parameters separated by commas, which indicates a correspondence of transmitting relays to receive relays. The mappings are designated by decimal integer values. The first transmitting relay designator corresponds to receive relay 1, the second corresponds to receive relay 2, and so on in order through receive relay 8.
Example:
CM 3,2,1,4,15,6,6,7 performs the relay mapping as follows:
Transmitter Relay — Receive Relay
3 — 1
2 — 2
1 — 3
4 — 4
15 — 5
6 — 6
6 — 7
7 — 8</td><td>1,2,3,4,5,6,7,8</td></tr>
<tr><td>CO</td><td>Relay Contact Control</td><td>CO string
CO ?
This command allows the receive relays to be temporarily activated/deactivated for test purposes.
The string in the command line is an 8 byte character string which controls the state of each relay contact. The first character controls relay number 1, the second controls relay number 2, and so on through relay 8. The valid values in the string are:
Value — State of Relay
0 — Deactivates a relay
1 — Activates a relay
X — Do not care (normal control channel processing)
This command is implemented before the relay sense command processing, thereby permitting a logical control of the contacts, not physical contact closure.</td><td>XXXXXXXX</td></tr>
</table>

Table 4-3. Alphabetical Command List (continued)

Command	Function	Syntax/Description	Default
CQ	Contact Status Query	CQ ? This command queries the physical state of the relay contact closures. The value returned is an eight-character value, each character representing the status of an individual relay. The first character corresponds to the first relay, the last character corresponds to the last relay. A '0' for a relay indicates that the relay is open, a '1' indicates that the relay is closed.	—
CS	Contact Sense	CS *string* CS ? This command controls the normal (deactivated) position for the DAC700 decoder relay closures. The *string* is an 8 byte character string with each character position controlling the normal state of a relay. Each character is separated by a comma. The first character controls relay number 1, the second character controls relay number 2, and so on in order through relay number 8. A '1' for an individual character indicates that the relay is normally closed and that the relay is activated by opening it. A '0' for an individual character indicates that the relay is normally open and that the relay is activated by closing it.	0,0,0,0,0,0,0,0
DC	Display Configuration	DC This command instructs the receiver to display its software version.	—
DD	Decoder Ancillary Data Rate	DD *baud,parity,data_bits,stop_bits* DD ? This command sets or reads the decoder ancillary (Aux) data (baud) rate. The valid values for the parameters are: **Parameter** — **Valid Values** Baud — 300, 1200, 2400, 3600,4800,7200, 19200, 384000, Parity O-odd, E-even, N-none Data bits — 7 or 8 Stop bits — 1 or 2 **NOTE:** This value is automatically set when the AG value for the encoder is modified.	38400,N,8,1

Table 4-3. Alphabetical Command List (continued)

Command	Function	Syntax/Description	Default
DM	Duplex Mode	DM *n* DM ? This command sets or reads the mux/demux operation to full duplex or simplex mode. In duplex mode, the DAC700 encodes and sends the digitally compressed data out and also loops it back through its onboard decoder. The signal is then available as needed in monitoring. In simplex mode, the DAC700 encodes and sends the digitally compressed data out without looping this information back through its onboard decoder. The valid values for *n* are: ***n*** **Mode** 0 Simplex mode 1 Full duplex mode **NOTE:** DM should be set to 0 (simplex) if PP is set to a rate > than 9600 for Passthrough.	1
FL	Fault Status	FL This command displays *accumulated* faults. The faults are bit-mapped registers, with each bit and associated weight assigned to a particular fault condition. The output value is the sum of all set fault weights. The table below provides a description of each fault and its weight. Example: If the Audio Overload, Left Channel and Audio Overload, Right Channel are active, the FL command returns the value 48. **Fault Condition** — **Weight** Encoder PLL not locked — 1 Decoder PLL not locked — 2 Decoder not framed (out of sync) — 4 Audio Overload, Left Channel — 16 Audio Overload, Right Channel — 32 Network or channel ID fault — 64	—
IN	Master Reset	IN 9346 This command resets the nonvolatile parameters to their factory default settings, shown in Table 4-5. A related command, RE, resets the entire unit and reinitializes all parameters, but does NOT reset the nonvolatile parameters.	—

Table 4-3. Alphabetical Command List (continued)

Command	Function	Syntax/Description	Default
LB	Loop Back Mode	LB *n* LB ? This command sets or reads the codec loop back mode. The *n* specifies the mode as shown below. ***n*** **Loopback Mode** 0 External baseband input operation 1 PCM loopback 2 Normal operation In external baseband input operation, an ISO/MPEG (Musicam) stream from an external digital audio is brought in through the broadcast data port and decoded. From the decoder, data is sent to the DAC, converted to analog audio, and sent out the analog output port. An independent ISO/MPEG (Musicam) stream can be sent out the broadcast data port. In the PCM loopback operation, analog audio is converted into 16 bit linear PCM data and transferred directly from the analog-to-digital converter output to the digital-to-analog converter input. The PCM data is processed through the DAC, converted into analog audio, and sent to the analog output port. In the normal operation, analog audio is converted into 16 bit linear PCM data and transferred directly from the audio/data encoder output to the decoder input. The PCM data is processed through the decoder, converted into analog audio, and sent to the analog output port.	2
LR	Left/Right Channel Toggle	LR *n* LR ? This command sets or reads the sense of the left and right audio channel output on the DAC700 decoder only. The values for *n* are: ***n*** **Description** 0 Normal sense 1 Reverse sense 2 Left input on both left and right output 3 Right input on both left and right output **NOTE:** This will not affect the sense of the audio transmitted to the receiver.	0
MU	Audio Mute	MU *n* MU ? This command sets or reads the muting of both the analog and digital audio outputs of the DAC700 decoder. The valid values for *n* are: ***n*** **Description** 0 Unmuted, audio outputs enabled 1 Muted, audio output disabled 2 Left channel muted, right channel unmuted 3 Right channel muted, left channel unmuted **NOTE:** This will not affect the audio transmitted to the receiver.	0

Table 4-3. Alphabetical Command List (continued)

Command	Function	Syntax/Description	Default
NI	Network ID	NI *n* NI ? Sets or reads the network ID number that is sent out as part of the relay control message. Valid range is 1 through 255.	1
PC	RS-232 Monitor and Control Port Configuration	PC *baud,parity,data_bits,stop_bits* PC ? Sets or reads the RS-232 M&C port configuration. Valid values for the parameters are: **Parameter** — **Valid Values** Baud — 1200, 2400, 4800, 9600 Parity — O-odd, E-even, N-none Data bits — 7 or 8 Stop bits — 1 or 2	1200,O,7,1
PD	Preset Definition	PD *preset_number,network_id,channel_id,encoder_mode, ag_value,transmit_channel_rate,receive_channel_rate* PD ? This command sets or reads the preset definition. *Preset_number* identifies each unique definition. The valid range of preset numbers is 1 to 8. The preset definition can be used to preset the configuration parameters for each channel into the DAC700 for that channel. To do this, the user defines the configuration for each channel using the PD command, then assigns the preset definition for that configuration to a particular DAC700 using the Preset Select (PS) command. The preset definition can also be used in redundancy protection. In redundancy protection, an extra DAC700 and modem pair serves as a backup to any one channel of a DAC700 and modem pair. When a user-defined fault is detected, a channel switchover automatically occurs. This switch initiates the replacement of the faulty channel's DAC700 and modem pair with the spare DAC700 and modem pair. Switchovers are controlled by the Preset Mode of the DAC700. To use this mode, three commands need to be configured: Preset Definition (PD), Preset Mode (PM), and Preset Selection (PS).Using the PD command, the user enters resets for all the channels in the system into a spare DAC700 for redundancy protection. *CONTINUED*	1,1,1,1, 38400, 128000, 128000

Table 4-3. Alphabetical Command List (continued)

Command	Function	Syntax/Description	Default
PD		*CONTINUED* To enable the preset mode, the user sets the PM command for each DAC700 to 1. When a fault is detected and the modem protection switch performs a channel switchover, the address of the faulty channel is sent to the spare DAC700 and a signal routing switch. With PM equal to 1, the spare DAC700 receives the address, selects the preset for that channel, and uses these parameters to reconfigure itself to mirror the faulty channel. Upon receiving the faulty channel's address, the signal routing switch selects the faulty channel's audio input and routes it to the spare DAC700 to complete the switchover. Once the switchover is complete, the signal from the spare codec/modem pair proceeds to the RF terminal to be transmitted to the satellite, via a satellite antenna, and then to the receiving sites.	
PM	Preset Mode	PM *n* PM ? This command sets or reads the preset mode. Valid values for *n* are: **n** **Description** 0 Mode is disabled 1 Presets are controlled from the three hardware input lines 2 Presets are controlled by the PS command In mode 1, the presets are controlled by the three TIL input lines on the AUX/AES port P3. The mappings of these pins to the preset definitions as follows: **Preset No. Pin 3 Pin 4 Pin 5** 1 0 0 0 2 0 0 1 3 0 1 0 4 0 1 1 5 1 0 0 6 1 0 1 7 1 1 0 8 1 1 1	0
PP	Data Port 2 Configuration (Pass-through)	PP *baud,parity,data_bits,stop_bits* This command sets or reads the data port 2 configuration. Valid values for the parameters are: **Parameter** **Valid Values** Baud 300,1200,2400,4800,9600 Parity O-odd, E-even, N-none Data bits 7 or 8 Stop bits 1 or 2	9600,O,7,1

Table 4-3. Alphabetical Command List (continued)

Command	Function	Syntax/Description	Default
PS	Preset Selection	PS *n* PS ? This command sets or reads the current preset. Valid range for *n* is 1 through 8. If preset mode = 1, then this command is read only and reports back the current hardware preset selection number. When a preset is selected, the associated preset definition parameters are loaded into their respective command values for unit reconfiguration.	1
PT	Passthrough Mode	PT *n* PT ? This command sets or reads the passthrough mode. An *n* of 0 disables the mode; an *n* of 1 enables the mode.	0
RA	Remote Port Address	RA *n* RA ? This command sets or reads the remote port address for the unit. Valid address range is 1 through 31.	31
RC	Remote Control Port Configuration RS-485	RC *baud,parity,data_bits,stop_bits* RC ? This command sets or reads the RS-485 M&C (remote control) port configuration. Valid values for the parameters are: **Parameter** — **Valid Values** Baud — 1200, 2400, 4800, 9600 Parity — O-odd, E-even, N-none Data bits — 7 or 8 Stop bits — 1 or 2	2400,O,7,1
RD	Receive Channel Rate	RD *n* RD ? The RD command sets or reads the receive channel rate operation for the decoder. The Receive Channel Rate should be the same as the Transmit Channel Rate (TD). Valid values for *n* are: 56000, 64000, 96000, 112000, 128000, 192000, 256000, 384000	128000
RE	Resets the Unit	RE This command forces the entire unit to reset and reinitialize all parameters. It is identical to a hardware reset. Unlike the IN 9346 command, the RE command does NOT reset the nonvolatile parameters. The following parameters are set: • AT 0 • CO X • CF 0 • LB 0	—

Table 4-3. Alphabetical Command List (continued)

Command	Function	Syntax/Description	Default
RM	Relay Message Interval	RM *n* RM ? This command sets or reads the time interval in 100 ms increments between relay control message updates. Valid range for *n* is 0 (off) through 100 (10 seconds).	1 (100 ms)
ST	Real-time Status	ST This command reads *real-time* faults. The faults are bit-mapped registers, with each bit and associated weight assigned to a particular fault condition. The table below provides a description of each fault and its weight. **Fault Condition** — **Weight** Encoder PLL not locked — 1 Decoder PLL not locked — 2 Decoder not framed (out of sync) — 4 Audio Overload, Left Channel — 16 Audio Overload, Right Channel — 32 Network or channel ID fault — 64	—
TD	Transmit Channel Rate	TD *n* TD ? This command sets or reads the transmit channel data rate operation for the decoder. Valid values for *n* are: 56000, 64000, 96000, 112000, 128000, 192000, 256000, 384000	128000
TE	Set Protection Mode	TE *n* TE ? This command sets or reads the protection mode. Values for *n* are: ***n*** — **Description** 0 — Disabled 1 — Enabled: the decoder input comes from the encoder's output and the transmit terminal timing output is qualified with the decoder's audio frame sync status.	0
TM	Encoder (Transmitter) Mode	TM *n* TM ? This command sets or reads the encoder mode. Valid values for *n* are: ***n*** — **Transmitter Mode** 0 — Mono 1 — Dual mono 2 — Joint stereo	0

Table 4-3. Alphabetical Command List (continued)

Command	Function	Syntax/Description	Default
TT	Transmit Timing Selection	TT *n* TT ? This command sets or reads the transmit timing mode: *n* specifies the mode as shown below: **n** **Timing Mode** 0 Internal 1 External (from ST input) 2 External (from RT input) 3 AES/EBU Interface	0
UB	User Data Time or Send Interval	UB *n* UB ? This command allows the user to determine the period at which user data is sent out with the aggregate data. *n* specifies the value in milliseconds. Example: An *n* of 500 sets the send interval to 500 ms or 0.5 sec. The valid range for *n* is 0 to 1000. A value of 0 indicates no time interval and causes the characters to be sent as they are received with one character per packet.	500
UC	Data Port 1 Configuration (User Data)	UC *baud,parity,data_bits,stop_bits* UC ? This command sets or reads the user data port configuration. Valid values for the parameters are: **Parameter** **Valid Values** Baud 0 (off), 300, 1200, 2400, 4800, 9600 Parity O-odd, E-even, N-none Data bits 7 or 8 Stop bits 1 or 2	2400,O,7,1
UT	User Termination	UT *n* UT ? The user termination command enables the user to specify an 8 bit character or value which is checked against incoming characters. *n* is the decimal value of the ASCII character to be used as the termination character. If the value is less than zero and matches a character, then the character is discarded from the input user stream and the current user block is terminated. If the character is greater than zero, then the termination character is stored in the current block for transmission, and the block is terminated. A zero value character disables input termination. Example: If N = 65, when ASCII 'A' is detected in the input user data stream, the current user data block is sent out with 'A' as the last character. If N = -65, when 'A' is detected, the block is sent without 'A.'	0

Table 4-3. Alphabetical Command List (continued)

<table>
<tr><th>Command</th><th>Function</th><th>Syntax/Description</th><th>Default</th></tr>
<tr><td>XM</td><td>Relay Mapping</td><td>XM active_input_mask,polarity_mask,debounce
XM ?

This command sets or reads the relay input port configuration. The three parameters are defined as follows:

• Active input mask

The active input mask selects which of the 16 inputs will be used in forming the relay control word.

The mask is specified in a decimal format with 65535 selecting all 16 bits, zero selects no active inputs. Selection numbers other than zero are:

Sensor Input Number / Selection Number
0 / 1
1 / 2
2 / 4
3 / 8
4 / 16
5 / 32
6 / 64
7 / 128
8 / 256
9 / 512
10 / 1024
11 / 2048
12 / 4096
13 / 8192
14 / 16384
15 / 32768

• Polarity mask

The polarity mask selects the sense of the input on a bit basis in decimal format. A '1' selects inverted, and a '0' selects no inversion.

For example, 65535 selects all bits to be inverted before being transmitted. Zero selects all bits to be noninverted. Given active-low hardware inputs, the inverted state, 65535, provides true polarity in the relay message packet.

• Debounce

The debounce selects the time interval applied to all inputs before a relay message is generated.

Valid values are 20 to 1000 ms.</td><td>65535,
65535,20</td></tr>
</table>

The IN command resets the nonvolatile parameters to their factory default settings, as shown in Table 4-4.

Table 4-4. Factory Default Settings

Command	Default Value
AB	0
AG	38400,N,8,2
AM	255
AS	0
AT	0
BL	30
CF	0
CI	1
CM	1,2,3,4,5,6,7,8
CO	XXXXXXXX
CS	0,0,0,0,0,0,0,0
DD	38400,N,8,1
DM	1
LB	2
LR	0
MU	0
NI	1
PC	1200,O,7,1
PD	1,1,1,1,38400,128000,128000
PM	0
PS	1
PT	0
RA	31
RC	2400,O,7,1
RD	128000
RM	1
TD	128000
TE	0
TM	0
TT	0
UB	500
UC	2400,O,7,1
UT	0
XM	65535,65535,20

Chapter 5: ComStream Packet Protocol Overview

Overview

This chapter presents an overview of the ComStream Packet Protocol and includes:

- Packet format and contents
- Extended addressing
- Controller-to-modem command structure and slave responses
- Packet communication modes
- Special packet control commands

In addition specifications for ComStream packet protocol and the physical interface connectors are described.

ComStream Packet Protocol Overview

Packet Protocol Interface

The ComStream packet protocol uses the RS-485 bus interface to control devices which reside on the bus. The RS-485 interface allows for a maximum of 31 devices to be connected to a single bus and is configured as a party line. Each device on the bus is given an address from 1 to 31 and responds to commands and messages containing its specific address. The default address for the DAC700 is "31."

Packet Protocol Messaging

Commands and messages are sent between a controller and individually addressable devices, referred to as slaves, via information packets. Some packets sent from the controller may request a return *acknowledgment packet* from the slave. This acknowledgment packet:

- Indicates if the command has been executed
- Replies with the appropriate error and/or status messages
- Signals that the slave can receive and process another packet

The rate at which the controller can send messages to:

- One specific slave is determined either:
  - Implicitly by the slave's message handling rate specification
  - Explicitly by acknowledgment packets sent from the slave back to the controller
- Different slaves are determined by the baud rate of the communication channel

Packet Format and Content

ComStream packet protocol runs under the standard asynchronous ASCII format with 1 start bit, 7 data bits, odd parity, and 1 stop bit.

Each packet must conform to the format shown in Figure 5-1.

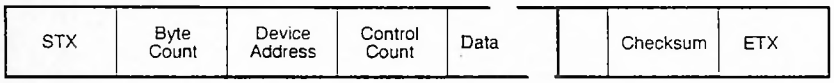


Figure 5-1. Packet Format

Each byte within the packet is defined as follows:

- STX – Signifies the beginning of a packet. Its ASCII value is STX=02_H.
- Byte Count – Represents the total number of bytes in the packet including the STX and ETX. The minimum count is 6, and the maximum count is 127.
- Device Address – Indicates the destination of a packet and is bit mapped as follows:
  - Bits 0-4 signify the slave address

    Although each slave recognizes only those packets which are addressed to it, all slaves will also respond to, but not acknowledge, packets addressed to device address zero.
  - Bit 5 is always 1
  - Bit 6 is the address extension bit. This bit allows a packet to be addressed to a nonprimary device, that is, a device which is not directly connected to the primary RS-485 bus. When set, additional address fields will follow. Address extension bit parameters are:
    - 0 – No address extension
    - 1 – Address extension

    Refer to *Extended Addressing*, which follows this section, for a more detailed explanation.

- Control Byte – Provides control information to the receiving device. It is bit mapped differently for controller-to-slave or slave-to-controller communication. Bit maps are as follows:
  - For controller-to-slave communications:
    - Bit 0 represents the packet acknowledge request as follows:
      - 0 – slave should not acknowledge receipt of packet
      - 1 – slave is required to acknowledge receipt of packet
    - Bits 1-6 are always 0
  - For slave-to-controller communications:
    - Bits 0-5 are always 0
    - Bit 6 is always 1
- Data – Used to transfer information from one device to another, such as commands from the controller to the slave or response information from the slave to the controller. The format of this data varies depending on message content.
- Checksum – Verifies the integrity of the characters contained within a message. The checksum is computed as the sum (module 128) of packet bytes, beginning with the device address (byte) and ending with the last byte of the data field.
- ETX – Signifies the end of a packet. Its ASCII value is ETX=03_H.

All controller or slave-generated packets have a:

- Minimum length of six bytes (*Minimum packet* format omits the data field and consists of STX, Byte Count, Address, Control, Checksum, ETX.)
- Maximum length of 127 bytes including delimiters and checksum

All packets with a length exceeding 127 bytes or that do not meet the prescribed format are discarded.

ComStream Packet Protocol Address Selection

To select the remote address of a DAC700 for ComStream packet protocol mode, use the M&C port to change the Packet Address with the RA command. For more information on the Packet Address command, refer to *Chapter 4: Remote Monitor and Control Operation.*

Extended Addressing

The ComStream packet protocol allows hierarchical addressing. Any device within the packet network may be addressed by providing a full *address path*.

The address field of a packet may be extended indefinitely if the:

- Appropriate address extension bits are set
- Maximum packet length is not exceeded

Intermediate ComStream devices in the addressing chain automatically forward the packet to the next device in the chain. Response packets, originating from the last device in the chain, are forwarded until they reach the controller.

If an acknowledge packet is required, the controller must adjust its time-out period to account for the packet forwarding delay encountered within the device hierarchy.

Packet Protocol Command Structure

All commands are initiated by a controller that sends addressed packets to the slave units on the packet party-line interface. A controller command can be:

- Configuration
- Control
- Status query

All commands are issued by the controller as odd parity, and 7 data bits. All responses are returned as odd parity and 7 data bits, with the exception of BD command responses.

For a listing of specific configuration, control, or query commands, refer to *Chapter 4: Remote Monitor and Control Operation*.

Configuration or control commands

The structure of configuration and control commands consists of:

- Two- or three-letter mnemonic ASCII string
- Single ASCII space character
- Optional ASCII parameter string

Configuration and control commands have the following structure:

```
xxx parameter
```

Status query commands

The structure of a status query command consists of:

- Two- or three-letter mnemonic ASCII string associated with the parameter to be interrogated
- Single ASCII space character
- ASCII question mark (?) character

Status query commands have the following structure:

```
xxx ?
```

Command execution

Commands are executed when the following conditions are met:

- Command is valid
- Parameter value, when required, is within the valid range
- Parameter value or command is compatible with the present modem configuration
- Command or query can be executed immediately

Commands that do not follow these guidelines are not executed and produce an error code. For information on error conditions and valid command parameters, refer to *Chapter 4: Remote Monitor and Control Operation*.

Controller Packet Command Example

To assist you in creating and sending viable packets, a byte-by-byte packet construction for the query command **EM ?** is presented in this section. This EM query packet is constructed by the controller and issued to a slave modem.

The following assumptions are made for this example:

- Slave address is 1
- Extended addressing is not required
- An acknowledgment packet is required
- ASCII codes used in this example include:
  - E is 45_H
  - M is $4D_H$
  - space character is 20_H
  - ? is $3F_H$

The packet to issue an EM query command is constructed as shown in Table 5-1.

Table 5-1. Packet Protocol EM Query Command

Byte Description	STX	Byte Count	Device Address	Control Byte	Data Byte				Check-sum	ETX
					E	M	space	?		
Before Parity	02_H	OA_H	21_H	01_H	45_H	$4D_H$	20_H	$3F_H$	13_H	03_H
After Odd Parity	02_H	$8A_H$	$A1_H$	01_H	45_H	CDH	20_H	BF_H	13_H	83_H

ComStream Packet Protocol Communications Modes

Two communications modes are supported by ComStream packet protocol:

- One-way communication consisting of controller-to-slave transmission with no slave response requested
- Two-way communications consisting of controller-to-slave transmission with a slave-to-controller acknowledgment response

The controller may indicate which communications mode is to be followed on a packet-by-packet basis by transmitting the appropriate control byte to the slave.

One-way communication mode

In one-way communication, information is sent one way only—from the controller to the slave. In this mode, the controller receives no information from the slave and is operating in an *open loop* mode. In this mode, the slave does not respond to the controller nor sends any type of messages informing the controller of commands received, non-executed commands, or error conditions.

Two-way communication mode

Operating in a two-way communication mode, the controller receives a response packet from the slave for every packet transmitted. The data portion of the slave response packet depends on the command issued by the controller.

The slave's response to a configuration or control command will be either:

- Empty packet (no data field) if the command was successfully completed
- Error sequence if the slave was not able to execute the requested modification

The slave's response to a status query will be either:

- Message packet containing status information consisting of the queried parameter, an ASCII space, and the parameter value or status information
- Error sequence if the slave was unable to obtain the queried parameter information

Special Packet Control Commands

Most of the standard modem monitor and control commands are supported by ComStream packet protocol. Commands which result in a response that exceeds the maximum packet byte count of 127 bytes are not supported. All DAC700 commands are supported.

ComStream Packet Protocol Specifications

The interface signal levels and electrical characteristics are in accordance with the IA RS-485, full duplex, tri-state interface bus standards. The bus is configured as a *party line* with a maximum of 32 devices connected to a single bus.

The physical interface connector is a 9-pin, D-type connector. Table 5-2 lists the pinout specifications for an RS-485 device.

Table 5-2. RS-485 Device Pinout Specifications

Pin	Signal	Direction
1	Rcv-	In
5	Gnd	—
7	Rcv-	In
8	Xmit+	Out
9	Xmit-	Out
2,3,4,6,8	—	No Connection

A positive differential at the RCV outputs [Vrcv+>Vrcv-], also known as a space, is intended as a TTL 0, which is considered to be a start bit per EIA specifications.

A positive differential at the Xmit inputs [Vxmit+>Vxmit-], also known as a space, will be interpreted as a TTL 0, which is considered to be a start bit per EIA specifications.

ComStream
A Spar Company

Chapter 6: Maintenance and Troubleshooting

Overview

This chapter provides information about the maintenance of the DAC700. This chapter also provides an alphabetical listing of the following:

- Key performance monitoring and maintenance commands
- Fault conditions
- Status/fault indicator lights

These listings also provide a description of each command and fault condition. Other maintenance topics include cables and connectors.

The last section of this chapter presents information that can help troubleshoot problems that may occur with the DAC700.

Maintenance

The DAC700 does not require scheduled or preventive maintenance. The unit has no internal or external configuration switches, jumpers, or straps that require setting by the technician or operator.

Cleaning the DAC700

The DAC700 may be cleaned, if desired. Before cleaning the DAC700, be sure to unplug it from the wall outlet. Do not use water or any type of abrasive pad, scouring powders, liquid cleansers, aerosol cleansers, or solvents such as alcohol or benzene.

Use only a clean, soft cloth lightly moistened with a mild detergent solution. Wipe all equipment with a clean, soft cloth lightly moistened with water to remove the detergent solution.

Performance Monitoring and Maintenance Commands

The DAC700 has a number of commands for monitoring key system parameters and performing maintenance tasks. By interrogating maintenance parameters on a periodic basis, the overall system performance level can be determined and changed if necessary.

Table 6-1 describes the key performance monitoring and maintenance commands. For more detail on these commands, refer to *Chapter 4: Remote Monitor and Control Operation.*

These commands are implemented using a remote control terminal or desktop computer operating in terminal emulation mode. For more detail on connecting the computer to the DAC700, see *Chapter 3: Installation and Initial Setup.*

Table 6-1. Key Performance Montoring and Maintenance Commands

Command	Function	Description
AT	Audio Test Tone	This command sets the tone generation mode. The DAC700 provides two internally generated audio test tones for checking the audio performance. By entering the appropriate parameter, the user can select an audio test tone and specify the channel to be checked—the left channel, the right channel, or both. The test tone is generated continuously until the operator stops it by entering the appropriate AT command parameter. **NOTE:** The audio test tone is generated internally in the decoder DSP and is, therefore, only testing the decoder section of the DAC700. Use the remote control computer or an oscilloscope connected to the analog output port to track the output level of the test tone. The test tone frequency level should be within ±0.5 dB of the stable level (active balanced output drives down to 50 ohm loads or unbalanced output). If the audio test tone is muted or not present, check to see if the MU (audio mute) command is active. Also check to see if the right and left channels can be reversed. Check this by using the LR (left/right channel toggle) command.
CO	Relay Contact Control	The CO command controls the relay contacts logically so they can be temporarily activated/deactivated without physical contact closure. Use this command to check for correct contact configuration and operation.
DC	Unit Configuration	This command lists the current unit operating parameters. Make sure the configuration parameters match the operating requirements. If appropriate, change the parameters to provide the intended unit performance.
IN 9346	Master Reset	This command resets the nonvolatile parameters to their factory default settings. (See *Chapter 4: Remote Monitor and Control Operation* for a table of factory defaults.) The RE command resets and reinitializes the unit but does NOT reset the nonvolatile parameters.

Table 6-1. Key Performance Montoring and Maintenance Commands (continued)

Command	Function	Description
LB	LOOPBACK	This command can be used to isolate system faults to a failure of the DAC700 unit or of another unit in the system, such as the modem or ABR receiver. Three loopback modes are available: • External Baseband Input Loopback Operation (LB 0) In external baseband input operation, an ISO/MPEG (Musicam) stream from an external source is brought in through the broadcast data port, decoded, and sent out the audio output ports. An independent ISO/MPEG (Musicam) stream can also be sent out the broadcast data port. This mode can be used to monitor the uplink, since you are using the DAC700 as a decoder only. External baseband input audio means that the onboard decoder and the DAC are operating correctly. Abnormal, muted, or absent loopback audio means that either the decoder or DAC are not operating correctly or are improperly configured. • PCM Loopback Operation (LB 1) In PCM loopback operation, analog audio is converted into 16 bit linear PCM data and transferred directly from the ADC output to the DAC input bypassing the DSP encoder/decoder. This mode can be used to verify the ADC performance while bypassing the algorithm. Normal PCM loopback audio means that the audio transmit and receive converters are operating correctly. Abnormal, muted, or absent loopback audio means the ADC, the DAC, or both have either failed or are not properly configured. • Normal Loopback Operation (LB 2) In normal operation the data is received from the analog input port, encoded, then sent out the broadcast data port. From the encoder, data is also sent to the decoder. Normal audio means that the audio transmit and receive sides are operating correctly. Abnormal, muted, or absent loopback audio means that either or both sides are not properly configured or have failed (that is, audio/data encoder or decoder, ADC, or DAC circuits).
RE	Reset Unit	This command resets the entire unit and reinitializes all parameters. It is identical to a hardware reset. Unlike the IN 9346 command, the RE command does NOT reset the nonvolatile parameters. This command should be used only if the technician suspects that one or more user-defined parameters are incompatible with the unit's physical configuration, functional configuration, or both. Refer to the RE command description in *Chapter 4: Remote Monitor and Control Operation* for the parameters affected and their default values.
XM	Relay Configuration	This command sets or reads the current or existing relay parameters. Check the relay input port configuration to determine the current mapping parameters. If change is appropriate, each contact can be configured independently or in combination with other contacts by the relay control messages from the unit controller.

Fault Condition Descriptions

Table 6-2 provides a description of fault conditions. Periodically check the current and accumulated fault codes using the **FL** or **ST** commands at the remote control computer.

Table 6-2. Fault Conditions

Fault	Condition
TX PLL Unlock	The TX PLL unlock fault occurs when the PLL reference is out of phase, exceeds the 25 Hz limit, or is not present.
RX PLL Unlock	This fault occurs when the PLL reference is out of phase, exceeds the 25 Hz limit, or is not present.
Decoder Not Framed	The decoder not framed fault occurs when the encoder output is abnormal, intermittent, or fails causing the decoder to lose frame synchronization.
Audio Overload—Left Channel	This fault occurs when the input level exceeds +18 dBu maximum, causing input clipping, distortion, and noise.
Audio Overload—Right Channel	The audio overload right channel fault occurs when the input level exceeds +18 dBu maximum, causing compression, distortion, and noise.
Network/Channel ID Fault	This fault is generated when the decoder has not received a network/channel ID for a period of 10 seconds. This fault indicates that either the decoder is not functioning properly or the encoder is not transmitting network/channel IDs. **NOTE:** Check the RM value to make sure the relay interval is not zero.

Status/Fault Indicator Lights

Two front panel indicators provide fault alarm information—Overload and Fault lights (LEDs). Check the indicator lights periodically along with the accumulated or real-time fault codes on the remote control computer.

Overload Indicator

The red overload warning indicator illuminates when the analog audio input level exceeds the +14 dBu limit. The right channel analog audio is provided via the analog input port.

Fault Indicator

A red indicator light appears when a summary fault occurs. This indicates one or more of the following failures:

- Audio muted or not present
- Transmit or receive PLL out of lock
- Encoder or decoder DSP errors
- Unit controller watchdog time-out exceeded
- Processor memory failure

Power Indicator

The indicator light is green as long as AC line power is applied to the unit power supply and the internal DC voltages are normal.

The light goes off if one of the following fails:

- AC line power
- Internal DC voltages
- Power supply module

Unreliable operating power can affect operation of the DAC700 unit and can sometimes damage electronic equipment or components. Power disturbances can be caused by a variety of sources, including lightning, switching high power electrical equipment on and off, and utility company activities such as power factor correction. (See *Chapter 7: Technical Specifications and Port Information.*)

Power problems may cause slow or erratic unit operation, audio distortion, intermittent alarms, or loss of configuration. In some cases, even unit or component failure can occur.

Connectors and Cables

Check the connectors on the rear panel of the DAC700 unit for any of the following defects:

- Loose, bent, or broken pins, recessed or missing pins in a connector, resulting in misalignment, improper mating, overheating, intermittent electrical contact, or shorting
- Cracked or broken connector shells, resulting in intermittent electrical contact or shorting
- Bare wire contact with a metal casing, resulting in overheating, intermittent electrical contact, or shorting
- Defective threads, missing guide pins, or key misalignment resulting in improper mating, electrical contact, or shorting

Check the interconnection cables on the unit and associated equipment for any of the following defects:

- Loose, cracked, or missing insulation and connectors on cables resulting in intermittent or no electrical contact or shorting
- Discolored, melted, or burnt insulation resulting from overloads, exceeding voltage/current ratings, or shorting
- Cable routed too close to electromagnetic interference (EMI) sources, resulting in intermittent data losses, excessive signal noise, crosstalk, feedback, or poor quality audio or data transmissions

General Troubleshooting

The general troubleshooting procedure requires using a remote control computer or laptop terminal. Also, the status and fault indicators on the front panel should be used to determine the cause of unit failures. Associated system equipment or facilities fault indications should be used, since they can assist in troubleshooting the DAC700 unit.

1. Check the status and fault indicator lights on the unit front panel for any alarms.
2. Using the remote control computer, check the fault display for any unit fault codes.
3. Check for any fault indications in associated system equipment, such as the modem, ABR receiver(s), RT terminal, control console, event sequencer, etc.
4. Check for any facility services faults, such as AC line power surge, drop, brownout, or loss; excessive temperature or humidity conditions in equipment room or enclosure; air conditioning, cooling fans, or blowers not operating; dirty or blocked air filters, etc.

Unit Troubleshooting

Table 6-3 provides a list of faults with the probable cause and corrective action that applies to the DAC700 unit. The remote control computer, laptop terminal, or terminal emulator is required to make full use of the troubleshooting information in the table. Also the status and fault indicators on the front panel should be used in conjunction with the remote control computer.

Table 6-3. DAC700 Unit Troubleshooting

Fault Indication	Action
Power indicator light (green) off	Ensure the unit is plugged into an active AC outlet. Verify that the line cord is firmly plugged into the rear panel receptacle. Ensure the line cord is not at fault by using a known working cord.
Overload indicator light (red) on	This indicates that the analog audio input level exceeds +18 dBu. Restore input audio to normal peak level.
Fault indicator light (red) on	This is a summary alarm that means one or more faults have occurred. Use the FL command (FL ?) to determine what faults are occurring. For each fault identified, refer to the fault description and suggested actions presented in this table. Ensure the audio is normal. Ensure the Unit controller has not exceeded the watchdog time-out and that the processor memory is functioning normally.
TX PLL unlock fault code	Ensure the transmit PLL reference is present, in phase, and within the 25 Hz limit.
RX PLL unlock fault code	Ensure that the receive PLL reference is present, in phase, and within the 25 Hz limit. Check the timing source. • If the internal source 12.288 MHz crystal-controlled oscillator (master clock) is being used, switch to an external timing source. • If an external source is being used for send timing (ST) or receive timing (RT), switch to an internal timing source.
Decoder not framed fault code	This indicates the audio/digital decoder has lost frame synchronization because the encoder output is abnormal, intermittent, or has failed.
Audio overload left channel fault code	This indicates analog audio input to the left channel exceeds the +18 dBu maximum level. Restore input audio to normal peak level.
Audio overload right channel fault code	This indicates analog audio input to the right channel exceeds the +18 dBu maximum level. Restore input audio to normal peak level.
Encoder DSP alarm code	This code indicates that audio/digital encoder DSP 1/DSP 2 is not working correctly or has failed. Anomalous or failed ADC operation affects downstream DSPs. Check analog audio input. If it is overloaded, too low, or absent, restore it to normal peak level.
Decoder DSP alarm code	Ensure audio/digital decoder DSP 4 operation is normal. Ensure DAC operation is normal.

COMSTREAM
A Spar Company

Chapter 7: Technical Specifications and Port Information

Audio Performance

Operation	Full duplex, encoder operates independently of decoder
Compression Technique	ISO/MPEG (Musicam), Layer II and IIa
Operating Rates (Analog)	56, 64, 96, 112, 128, 192, 256, and 384 kbps
Operating Rates (AES/EBU)	64, 96, 128, 192, 256, and 384 kbps
Operating Modes	Mono, dual mono (stereo), and joint stereo
Frequency Response	15 Hz to 20.0 kHz
Analog Sampling Rate	48 kHz
Analog Input Levels	+0 dBu average program (APL) level recommended, +18 dBu max
Input Impedance	>100 kohm
Output Levels	+18 dBu max; active balanced or unbalanced output drives down to 50 ohm loads; no adjustments
Digital I/O	AES/EBU transformer coupled, balanced I/O
Total Harmonic Distortion	<0.2% at 1 kHz at +8 dBm
Dynamic Range	>90 dB
Signal to Noise	>90 dB at 1 kHz (referenced to +18 dBu)
Idle Channel Noise	< -67 dBm (unweighted)
Crosstalk (two channel)	>85 dB (referenced to +18 dBu)
Stereo Phase Deviation	<1.0°, all frequencies
End-End Gain	±0.5 dB at 1 kHz, no adjustments
End-End Stability	±0.5 dB, 15 to 20 kHz, referenced to gain at 1 kHz, no adjustments
Channel Mute	>98 dB (referenced to +18 dBu)

Available Audio Rates and Bandwidths

Table 7-1 lists the available audio rates and bandwidths.

Table 7-1. Available Audio Rates and Bandwidths

Rate (kbps)	Audio Mode	Analog BW A (kHz) AB 0	Analog BW B (kHz) AB 1	Analog BW C (kHz) AB 2	Analog BW D (kHz) AB 3
56	Mono	10.50	10.50	10.50	10.50
	Dual Mono (Stereo)	8.25	8.25	8.25	8.25
	Joint Stereo	10.50	10.50	10.50	10.50
64	Mono	10.50	10.50	10.50	10.50
	Dual Mono (Stereo)	8.25	8.25	8.25	8.25
	Joint Stereo	10.50	10.50	10.50	10.50
96	Mono	15.00	10.50	10.50	10.50
	Dual Mono (Stereo)	8.25	8.25	8.25	8.25
	Joint Stereo	12.50	10.50	10.50	10.50
112	Mono	20.00	15.00	12.70	10.50
	Dual Mono (Stereo)	8.25	7.50	6.75	6.00
	Joint Stereo	20.00	15.00	12.70	10.50
128	Mono	20.00	15.00	12.70	10.50
	Dual Mono (Stereo)	8.25	7.50	6.75	6.00
	Joint Stereo	20.00	15.00	12.70	10.50
192	Mono	20.00	15.00	12.70	10.50
	Dual Mono (Stereo)	20.00	15.00	12.70	10.50
	Joint Stereo	20.00	15.00	12.70	10.50
256	Mono	20.00	15.00	12.70	10.50
	Dual Mono (Stereo)	20.00	15.00	12.70	10.50
	Joint Stereo	20.00	15.00	12.70	10.50
384	Mono	20.00	15.00	12.70	10.50
	Dual Mono (Stereo)	20.00	15.00	12.70	10.50
	Joint Stereo	20.00	15.00	12.70	10.50

Diagnostics

The following diagnostics are all selectable from the Monitor and Control (M&C) port.

PCM Loopback	PCM output of A/D is fed directly into D/A
Normal Operation	Decoder input comes from encoder output instead of the broadcast data port
Test Tone Output	1,000 Hz and 9,600 Hz; left only, right only, or both channels

Rear Panel Ports

Rear Panel Port Mapping

Rear panel port mapping is shown in Table 7-2.

Table 7-2. Rear Panel Port Mapping

Port Number	Port Name
J1	Analog Output
J2	Analog Input
J3	AUX/AES
J4	Broadcast Data
J5	User Data
J6	M&C
J7	Relay/Control

Analog Input Port

Connector	DB-9 female

Analog Output Port

Connector	DB-9 male

AUX/AES Port

Interface Type	Transformer coupled, balanced I/O
Connector	DB-15 female

Broadcast Data Port

Interface Type	Asynchronous RS-422
Data Rates	56, 64, 96, 112, 128, 192, 256, and 384 kbps
Connector	DB-15 female
Clocking	Internal, external, or derived from AES/EBU
Operating Modes	1. Normal 2. Protected (decoder sync status controls clock presence)

User Data Port

Interface Type	Asynchronous RS-232
Data Rates	300, 1200, 2400, 4800, and 9600 baud
Connector	DB-9 female

Passthrough Port

Interface Type	Asynchronous RS-232
Data Rates	300, 1200, 2400, 4800, and 9600
Connector	DB-9 female

Relay/Control Port

Recoder Output Contact Closures	8, form A, individually controlled
Encoder Inputs	16 TTL opto-isolated inputs
Connector	DB-37 female

M&C Port

Interface Type	RS-232 asynchronous or RS-485 multidrop packet; address is configurable from control port, default is address 31; default interface is RS-232; no handshaking signals
Connector	DB-9 female
Functions	Unit configuration, status, and diagnostics; connects to terminal emulator (RS-232) or ComStream packet protocol (RS-485)

Front Panel LED Status

The front panel LED descriptions are shown in Table 7-3.

Table 7-3. LED Descriptions

Label	Function	LED Color
Power	Power on	Green
Audio Sync	Decoder framing OK	Green
Dual Mono	Audio mode	Green
Joint Stereo	Audio mode	Green
Overload	Left/right audio overload	Red
Blank	Fault status	Red

Mechanical (IDU)

Size	4.5 cm x 42.5 cm x 38 cm (48 cm rack mount); 1.75" x 17 x 15" (19" rack mount)
Weight	<6 kg (12 lbs)
Shipping Weight	24 lbs

Power

Input Voltage (AC)	90 to 264 VAC (auto ranging)
Frequency	47 to 63 Hz
Consumption	<40 w (typical)

Environmental

Temperature	0° to 50°C (operating)
Humidity	0 to 95% noncondensing
Safety	UL 1950, CSA 950, and TUVPS EN 60950
Emission	FCC Part 15B Class A CISPIR 22 Class B

Appendix A: Interface Pinouts

Analog Output Port J1

The pin positions for the analog output port connector are given in Figure A-1.

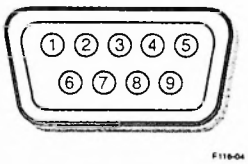


Figure A-1. DB-9 Male Connector

The pinout specifications for the analog output port connector are given in Table A-1.

Table A-1. DB-9 Male Connector

Pin #	I/O	Name	Description
1	O	LO+	Left Audio Output (+)
2	O	LO-	Left Audio Output (-)
3	—	—	Not Used
4	O	RO+	Right Audio Output (+)
5	O	RO-	Right Audio Output (-)
6	O	AGND	Analog Ground
7	—	—	Not Used
8	—	—	Not Used
9	O	AGND	Analog Ground

Analog Input Port J2

The pin positions for the analog input port connector are given in Figure A-2.

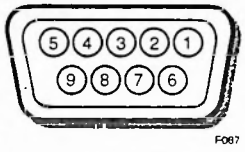


Figure A-2. DB-9 Female Connector

The pinout specifications for the analog input port connector are given in Table A-2.

Table A-2. DB-9 Female Connector

Pin #	I/O	Name	Description
1	O	LI+	Left Audio Input (+)
2	O	LI-	Left Audio Input (-)
3	—	—	Not Used
4	O	RI+	Right Audio Input (+)
5	O	RI-	Right Audio Input (-)
6	O	AGND	Analog Ground
7	—	—	Not Used
8	—	—	Not Used
9	O	AGND	Analog Ground

AUX/AES Port J3

The pin positions for the AUX/AES port connector are given in Figure A-3.

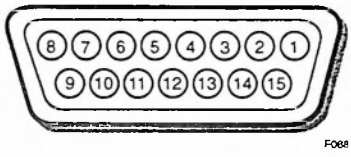


Figure A-3. DB-15 Female Connector

The pinout specifications for the AUX/AES port connector are given in Table A-3.

Table A-3. DB-15 Female Connector

Pin #	I/O	Name	Description
1	O	SG	Signal Ground
2	O	Status+	Status Closure Contact 1
3	I	PS0	Preset Select Bit 0
4	I	PS1	Preset Select Bit 1
5	I	PS2	Preset Select Bit 2
6	I	DIGIN-	AES/EBU Digital Audio In (-)
7	O	SG	Signal Ground
8	O	DIGOUT-	AES/EBU Digital Audio Out (-)
9	I	RST-	Remote Control Reset (short to pin 1, SG)
10	O	Status-	Status Closure Contact 2
11	—	—	Reserved
12	I	Reset	External Reset
13	O	DIGIN+	AES/EBU Digital Audio In (+)
14	O	SG	Signal Ground
15	O	DIGOUT+	AES/EBU Digital Audio Out (+)

Broadcast Data Port J4

The pin positions for the broadcast data port connector are given in Figure A-4.

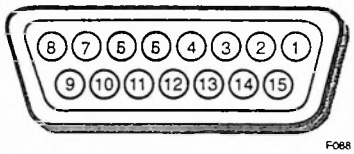


Figure A-4. DB-15 Female RS-422 Connector

The pinout specifications for the broadcast data port connector are given in Table A-4.

Table A-4. DB-15 Female RS-422 Connector

Pin #	I/O	Name	Description
1	O	FG	Frame Ground
2	O	TD+	Transmit Data (+)
3	O	TT+	Terminal Timing (+)
4	I	RD+	Receive Data (+)
5	—	—	Not used
6	I	RT+	Receive Timing (+)
7	I	ST+	Send Timing (+)
8	I	SG	Signal Ground
9	O	TD-	Transmit Data (-)
10	O	TT-	Terminal Timing (-)
11	I	RD-	Receive Data (-)
12	—	—	Not used
13	I	RT-	Receive Timing (-)
14	I	ST-	Send Timing (-)
15	—	—	Not used

User Data Port J5

The pin positions for the user data port connector are given in Figure A-5.

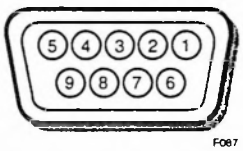


Figure A-5. DB-9 Female RS-232 Connector

The pinout specifications for the user data port connector are given in Table A-5.

Table A-5. DB-9 Female RS-232 Connector

Pin #	I/O	Name	Description
1	I	AUXRTS	Aux Request to Send
2	O	RD	Receive Data
3	I	TD	Transmit Data
4	O	CTS	Clear to Send
5	O	SG	Signal Ground
6	I	RTS	Request to Send
7	O	AUXRD	Aux Receive Data
8	I	AUXTD	Aux Transmit Data
9	O	AUXCTS	Aux Clear to Send

M&C Port J6

The pin positions for the M&C port connector are given in Figure A-6.

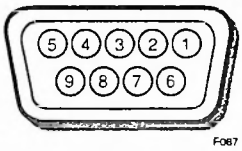


Figure A-6. DB-9 Female RS-232 or RS-485 Connector

The pinout specifications for the M&C port connector are given in Table A-6.

Table A-6. DB-9 Female RS-232 or RS-485 Connector

Pin #	I/O	Name	Description
1	O	TD-	Transmit Data, RS-485 (-)
2	O	RD	Receive Data, RS-232
3	I	TD	Transmit Data, RS-232
4	O	CTS	Clear to Send, RS-232
5	O	SG	Signal Ground
6	I	RTS	Request to Sent, RS-232
7	O	TD+	Transmit Data, RS-485 (+)
8	I	RD-	Receive Data, RS-485 (-)
9	I	RD+	Receive Data, RS-485 (+)

Relay/Control Port J7

The pin positions for the relay/control port connector are given in Figure A-7.

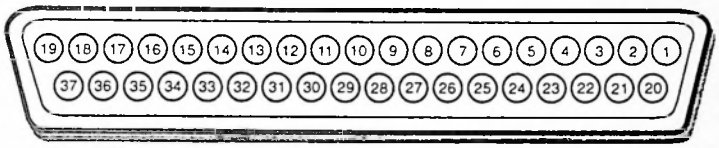


Figure A-7. DB-37 Female Connector

The pinout specifications for the relay/control port connector are given in Table A-7.

Table A-7. DB-37 Female Connector

Pin #	I/O	Name	Description
1	I	S1	Sensor Input 1, opto (cathode)
2	I	S2	Sensor Input 2, opto (cathode)
3	I	S3	Sensor Input 3, opto (cathode)
4	I	S4	Sensor Input 4, opto (cathode)
5	I	S5	Sensor Input 5, opto (cathode)
6	I	S6	Sensor Input 6, opto (cathode)
7	I	S7	Sensor Input 7, opto (cathode)
8	I	S8	Sensor Input 8, opto (cathode)
9	I	S9	Sensor Input 9 opto (cathode)
10	I	S10	Sensor Input 10 opto (cathode)
11	I	S11	Sensor Input 11, opto (cathode)
12	I	S12	Sensor Input 12, opto (cathode)
13	I	S13	Sensor Input 13, opto (cathode)
14	I	S14	Sensor Input 14, opto (cathode)
15	I	S15	Sensor Input 15, opto (cathode)
16	I	S16	Sensor Input 16, opto (cathode)
17	O	OPTOCOM	Opto Common (anode)
18	O	+12 v	+12 v thru 220 ohm resistor
19	O	SG	Signal Ground
20	O	RC1A	Relay Closure Contact 1A
21	O	RC1B	Relay Closure Contact 1B
22	O	RC2A	Relay Closure Contact 2A
23	O	RC2B	Relay Closure Contact 2B
24	O	RC3A	Relay Closure Contact 3A
25	O	RC3B	Relay Closure Contact 3B

Table A-7. DB-37 Female Connector (continued)

Pin #	I/O	Name	Description
26	O	RC4A	Relay Closure Contact 4A
27	O	RC4B	Relay Closure Contact 4B
28	O	RC5A	Relay Closure Contact 5A
29	O	RC5B	Relay Closure Contact 5B
30	O	RC6A	Relay Closure Contact 6A
31	O	RC6B	Relay Closure Contact 6B
32	O	RC7A	Relay Closure Contact 7A
33	O	RC7B	Relay Closure Contact 7B
34	O	RC8A	Relay Closure Contact 8A
35	O	RC8B	Relay Closure Contact 8B
36	O	SG	Signal Ground
37	O	SG	Signal Ground

Appendix B: Abbreviation/Acronym List

The following is a list of abbreviations and acronyms used throughout this manual:

A	Amps
ABR	Audio Broadcast Receiver
AC	Alternating Current
ADC	Analog-to-Digital Converter
AES/EBU	Audio Engineering Society/European Broadcast Union
ANMS	Audio Network Management System (ComStream proprietary)
APL	Average Program Level
ASCII	American Standard Code for Information Interchange
AUX	Auxiliary
bps	Bits per second
BPSK	Bi-Phase Shift Keying
CCA	Circuit Card Assembly
CRC	Cyclic Redundancy Check
CD	Compact Disc
CSA	Canadian Standards Association
DAC	Digital-to-Audio Converter
dB	Decibels
dBm	Decibels referenced to 1 milliwatt
dBr	Decibels above reference number
dBu	Decibels relative to a reference of 0.775 volts RMS
DSP	Digital Signal Processor

EIA	Electronic Industries Association
EPROM	Erasable Programmable Read Only Memory
EMI	Electromagnetic Interference
FCC	Federal Communications Commission
FDM	Frequency Division Multiplexing
ID	Identification
IEC	International Electrotechnical Commission
ISO	International Standards Organization
IXC	Interexchange Carrier
Hz	Hertz
kbps	Thousand bits per second
kHz	Thousand Hertz
LED	Light Emitting Diode
LNA	Low Noise Amplifier
LNB	Low Noise Block downconverter
M&C	Monitor and Control
MHz	Megahertz (million Hertz)
MPEG	Motion Picture Experts Group
MR	Master Reset
PC	Personal Computer
PCM	Pulse Code Modulation
PLL	Phase Lock Loop
PPL	Peak Program Level
PROM	Programmable Read Only Memory
QPSK	Quadrature Phase Shift Keying
RAM	Random Access Memory
RF	Radio Frequency
RMA	Return Material Authorization

Rx	Receive
SR	Status Relay
TDM	Time Division Multiplexing
TTCLK	Transmit Timing Clock
TTL	Transistor-Transistor Logic
Tx	Transmit
TXCLK	Transmit Clock
UART	Universal Asynchronous Receiver/Transmitter
UL	Underwriters Laboratories
UPS	Uninterruptible Power Supply
v	volts
VDE	Verband Deutscher Elektrotechniker
w	watts
XCO	Crystal Controlled Oscillator

COMSTREAM
A Spar Company

Glossary

16 Bit Linear PCM

An audio encoding format in which 16 bits of data for each sample are used with Pulse Code Modulation (PCM).

AES/EBU

AES/EBU is the acronym for Audio Engineering Society/European Broadcast Union and their standard for digital audio interfaces.

Amplify

To strengthen audio signals that have become weaker during processing.

Analog-to-Digital

A device that converts an analog signal to a digital signal.

Ancillary Data

Ancillary data is nonaudio data that is processed by the DAC700. Ancillary data has three components: relay blocks, network control message blocks, and user data blocks. For more detail, see the definitions for each of the components.

Asymmetric

A compression technique that requires intensive processing on the compression end, but little processing on the decompression end.

Buffered

Placed in a temporary storage location before being sent or received.

Codec/Multiplexer

A codec/multiplexer is a piece of equipment that combines the functions of both a codec and a multiplexer.

A codec consists of a series of integrated circuits that perform a specific analog-to-digital conversion. The DAC700 codec converts analog sound signals to a digital bit stream.

A multiplexer simultaneously transmits two or more messages on a single transmission path. The multiplexer also receives messages and controls the communication lines.

DSP (Digital Signal Processor)

A DSP is a specialized digital microprocessor that performs calculations on the digitized signals that have been converted from analog and sends the results on.

Duplex Mode

Duplex mode is an optional configuration for operating the DAC700. In duplex mode, the DAC700 encodes and sends the digitally compressed data out and also loops it back through its onboard decoder. The signal is then available as needed for monitoring performance.

External Baseband Input Operation (LB 0)

In external baseband input operation, an ISO/MPEG (Musicam) stream from an external source is brought in through the broadcast data port, decoded, and then sent out the audio output ports. An independent ISO/MPEG (Musicam) stream can also be sent out the broadcast data port.

Loopback

A type of diagnostic test in which the transmitted signal is returned to the sending device after passing through the DAC700. This enables a technician to compare the output signal with the input signal to get some sense of what is wrong.

The DAC700 has two diagnostic loopback modes: PCM loopback and network loopback. (For more detail, see the definitions of the specific loopback modes.)

Network Control Message Blocks

Network control message blocks are one of three types of ancillary data processed by the DAC700. Network control messages consist of commands generated by the ANMS computer and sent to the receivers in the transmission.

Of the three types of data, network message blocks have the second highest priority.

Normal Operation (LB 2)

In normal operation, the data is received from the analog input port, encoded, and then sent out the broadcast data port. From the encoder, data is also sent to the decoder.

Passthrough Mode

The DAC700 passthrough mode is a selectable configuration for transferring user data. Passthrough mode allows the user to transfer *preformatted* data. Preformatting the data minimizes the overhead of the onboard multiplexer.

The passthrough mode can be selected at any time independent of user data, relay control messages, or control channel information.

PCM Loopback Operation (LB1)

In PCM loopback operation, the data is looped from the output ADC through the DAC, bypassing the DSP encoder/decoder. This mode can be used to verify the ADC performance while bypassing the algorithm.

Point-to-Point

Communication between two points only.

Protected

A protected system has the built-in capability to detect when a fault has occurred in the uplink transmission of one of the channels. When a fault is detected, the system swaps out the faulty DAC700/CM701 with the channel in the rack referred to as the "spare." The spare channel also consists of a DAC700/CM701. Using this technique ensures that the rack is always "protected," unless a fault has previously occurred and the faulty channel has not yet been fixed.

Redundancy Protection

Redundancy protection is an optional configuration in which an extra DAC700 serves as a backup to one or more other DAC700s. Users can preset definitions to match the configuration of the redundant DAC to one or more primary units. Users can also configure the system to automatically swap DAC700s if a primary unit goes out of operation.

Relay Blocks

Relay blocks are one of three types of ancillary data that can be processed by the DAC700. Relay blocks include the input relay information and the network ID and channel ID information. Relay blocks are sent out at user-programmable intervals in multiples of 100 ms. The length of the interval is programmed in the DAC700 with the RM command. (See *Chapter 3: Remote Monitor and Control Operation*.)

Of the three types of data, relay blocks have the highest priority.

RF Terminal

A radio frequency terminal (RF terminal) is composed of a modem, an earth station, a radio, and an uplink dish.

Sampling

The process of converting continuous signals (analog), such as voice or music, into discrete (binary) values.

Sampling Rate

The number of times per second that an analog signal is measured and converted to a binary number.

Shadow

To act as a back-up unit; for example, to provide an extra DAC700 to back up one or more other DAC700s. (See redundancy protection.)

Simplex

In simplex mode, the DAC700 encodes and sends the digitally compressed data out without looping this information back through its onboard decoder.

Subsystem

The major components that make up a system. For example, the DAC700 and CM701 are the uplink modulator subsystem. The ABR200 with an LNB and dish with IFL cable are the receive site downlink subsystem.

Swap

To switch from one operating unit to another unit with the same configuration.

TTL Logic Levels

TTL stands for Transistor-Transistor Logic and references on and off levels. A one equals a high (+5 v).

Unprotected

An unprotected system does not have the built-in capability to switch to a spare piece of equipment when a fault occurs.

User Data Blocks

User data blocks are one of three types of ancillary data that can be processed by the DAC700. User data blocks are originated by the user and input into the DAC700 for broadcast transmission to all receivers.

Of the three types of data, user data blocks have the lowest priority.

ComStream
A Spar Company

Index: DAC700

ComStream
A Spar Company invites you to comment on our manual...

Since our manuals are designed and written for you, we would like your input to ensure that we continually deliver the best in customer documentation. Thank you in advance for taking a few minutes to complete this survey.

Which ComStream manual are you providing input to? _____

Please circle your response to each of the following statements.

	Strongly Agree	Agree	Disagree	Strongly Disagree	Not Applicable
The manual is well organized.	1	2	3	4	N/A
The information is easy to locate.	1	2	3	4	N/A
The information is complete and accurate.	1	2	3	4	N/A
The instructions and procedures are easy to follow.	1	2	3	4	N/A
The illustrations are accurate.	1	2	3	4	N/A
The illustrations help clarify the text.	1	2	3	4	N/A
The number of illustrations is about right.	1	2	3	4	N/A

Please answer these questions.

What did you like about this manual? _____

_____

_____

What would you change in this manual? _____

_____

_____

What would you like to see in this manual? _____

_____

_____

What other comments about the manual do you have? _____

_____

_____

Please tell us a little about yourself.

Is this your first ComStream product? ___ Yes ___ No

How many years have you been installing, operating, or using digital satellite communications equipment? ☐ Less than 1 year ☐ 1-2 ☐ 3-5 ☐ 6-10 ☐ More than 10

Name _____ Title _____

Company _____

Address _____

City/State/Province/Country _____

Postal/Zip Code _____ Phone _____

Fold here, tape at bottom, and mail Thank you

Place
Postage
Here

ComStream
A Spar Company

10180 Barnes Canyon Road
San Diego, CA 92121

Attn: Customer Service