IBM

Series/1

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IBM Series/1

Audio Input/Output Feature

RPQs D02337 and D02338

Custom Feature

Preface

This manual provides general information about the Audio Input/Output Feature card for the IBM Series/1 4987 Programmable Communications Subsystem. This publication includes descriptions of the following RPQs:

- RPQ D02338—Audio Input/Output Feature card and data set cable
- RPQ D02337—Audio Input/Output Feature card, data set cable, audio input cable, and diagnostic group

The intended audience for this publication is customer executives, programmers, and maintenance personnel who will use this information to order products, prepare machine-language code, and supplement other maintenance aids.

This manual is organized as follows:

Chapter 1. Introduction provides a general description and overview of the Audio Input/Output Feature attachment to the IBM Series/1 4987 Programmable Communications Subsystem.

Chapter 2. Operation describes audio data rate considerations, the device control registers for the 407 data set and voice data, vocabulary generation tips, and sample function strings.

Chapter 3. Maintenance Information provides a detailed description of the interface lines to the 4987 and the data set rate switch, and explains the coder/decoder process.

Appendix A. Reference Information describes the data set and audio cables and the wrap connector.

The user of this manual should be familiar with data processing concepts and have a basic understanding of both the IBM Series/1 and the IBM Series/1 4987 Programmable Communications Subsystem and their capabilities.

Prerequisite Publications

IBM Series/1 4987 Programmable Communications Subsystem and 4990 Model 1 Communications Console for the 4987 Description, GA34-0049.

First Edition (August 1979)

Use this publication only for the purpose stated in the Preface.

Changes are periodically made to the information herein; any such changes will be reported in subsequent revisions or Technical Newsletters.

Contents

Chapter 1. Introduction

RPQ D02338 consists of an Audio Input/Output Feature card and a 6.1-m (20-ft) interface cable for the 407 data set or equivalent.

RPQ D02337, in addition to the items in D02338, contains an audio input cable plus the diagnostic group, which includes a wrap connector, a diagnostic diskette, and associated documentation.

The Audio Input/Output Feature, which is contained on a printed circuit card, provides the interface between the IBM Series/1 4987 Programmable Communications Subsystem and the customer's data communication equipment. This feature card accepts either voice inputs or tones generated by pushbutton telephones and responds with voice output through a Western Electric 407 A, B, or C data set with parallel option (or equivalent). See Figure 1-1.

Each feature card occupies one slot in the 4987 subsystem and is capable of servicing one line of a switched or private telephone network.

The major functions contained on this feature card are:

- Coder/decoder
- Data set interface circuitry
- 4987 interface circuitry

The coder/decoder digitizes audio signals received from a tape recorder or telephone network for storage in the Series/1. The coder/decoder also decodes the assembled digital messages and presents audio signals to the 407 data set.

The 4987 subsystem can contain up to two scanner cards, each of which can service eight audio feature cards.

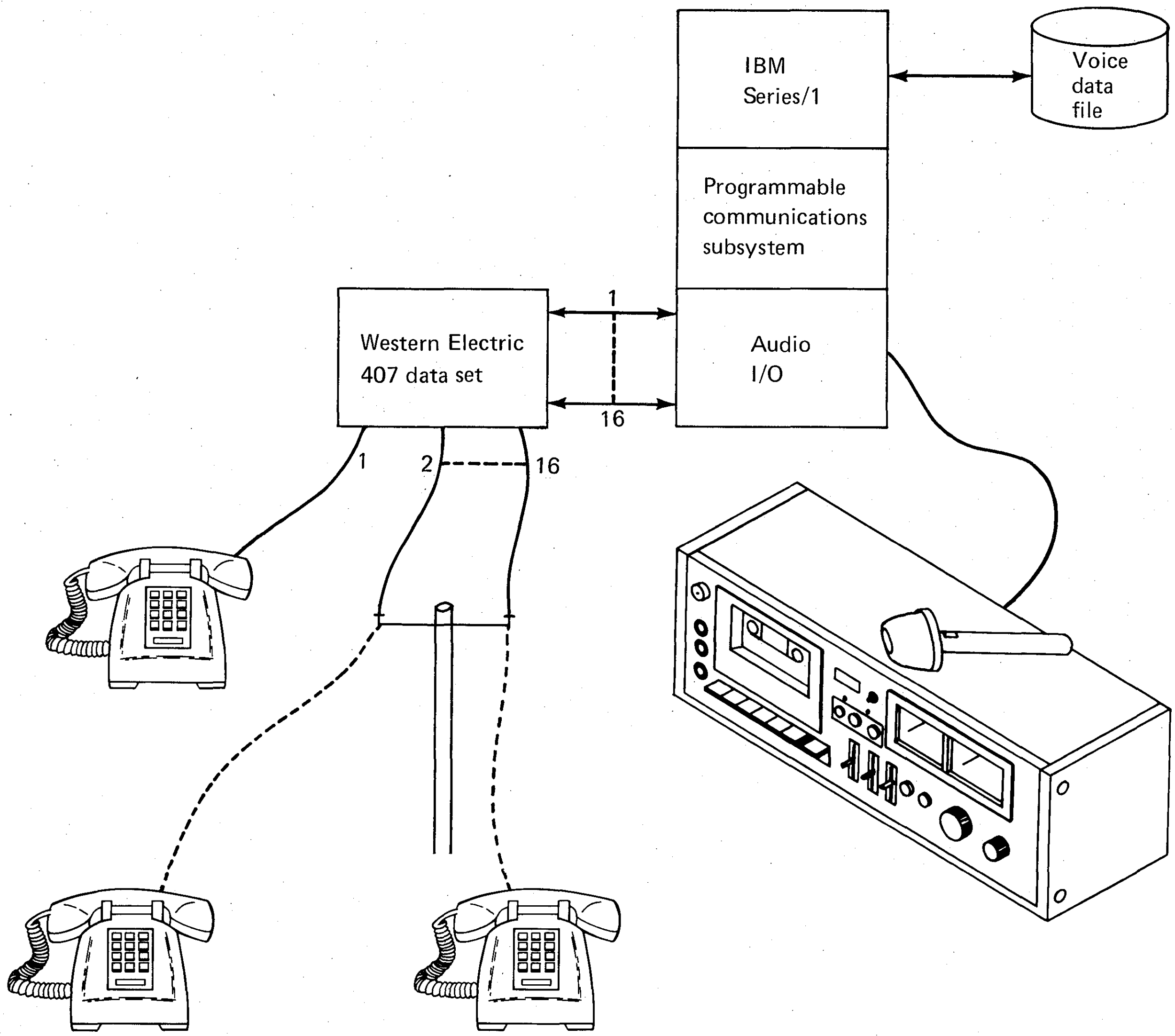


Figure 1-1. Audio input/output system

Chapter 2. Operation

The Audio Input/Output Feature is used to:

- Interface with a 407A, B, or C data set with parallel options (or equivalent)
- Provide audio encoding and decoding

If only the 407 data set interface (Touch-Tone* function) is used, this feature may be mixed with up to a maximum of 15 other features in the IBM 4987 Programmable Communications Subsystem with a single scanner. If the audio function is used, a maximum of eight Audio Input/Output Feature cards may be used with each 4987 subsystem scanner. Other features should not be intermixed when the audio function is used. Up to 16 Audio Input/Output Feature cards may be contained in one 4987 with two scanners.

The audio portion of the card operates at data rates of from 10.24K baud to 19.20K baud, which are selectable by switches on the card. The higher data rates produce better audio quality; however, at higher data rates, more storage is required and fewer lines may be actively receiving and transmitting audio simultaneously, as shown in Table 1.

Typical applications require audio for a small portion of a transaction. Because the audio lines operate independently of the data set lines, an audio line can be opened, a message can be transmitted, and then the line can be closed. Thus if the amount of audio required is low, a large number of telephone lines can be serviced, and the number in the second column of Table 1 can be exceeded. This maximum number is also affected by differences in coding the function strings.

Data overruns may occur if these maximums are exceeded. An overrun recovery procedure to transmit the remainder of the data block can greatly reduce the effect of the overrun.

The buffers that hold the audio data should be capable of holding a spoken word or about one-half second of data. Two buffers should be assigned for each audio line that will be operating simultaneously—one buffer is being used for transmitting while the other is being updated by the disk.

Audio data rate (kilobaud)	Maximum no. of lines operating simultaneously	Scan rate (minimum entries in scan table) Transmit	Receive	Storage for one sec. of speech (kilobytes)
19.20	1	4	8	2.40
17.07	2	4	8	2.13
15.36	2	4	8	1.92
13.96	3	4	4	1.75
12.80	3	4	4	1.60
11.82	3	2	2	1.48
10.97	3	2	2	1.37
10.24	3	2	2	1.28

Table 1. Data rate

* Trademark of American Telephone & Telegraph Co.

Programming Considerations—Controls

The Audio Input/Output Feature is controlled by two device control registers. Some bits in these registers can be set by the controller in the 4987 Programmable Communications Subsystem. The meanings of the bits in the registers varies, depending on whether the registers are being read from or written into. You cannot read what you have written into the registers except in wrap-back test. When a Write Device Control (WRDC) order is executed, the register is in write mode. When a Branch on Device Control (BDVC) order is executed, the register is in read mode. Both modes allow the controller to specify when certain control lines on the interface become active.

Each Audio Input/Output Feature card is assigned two device addresses: one odd and one even. The even address is assigned to the 407 data set functions. The odd address is assigned to the audio functions. See Figure 2-1.

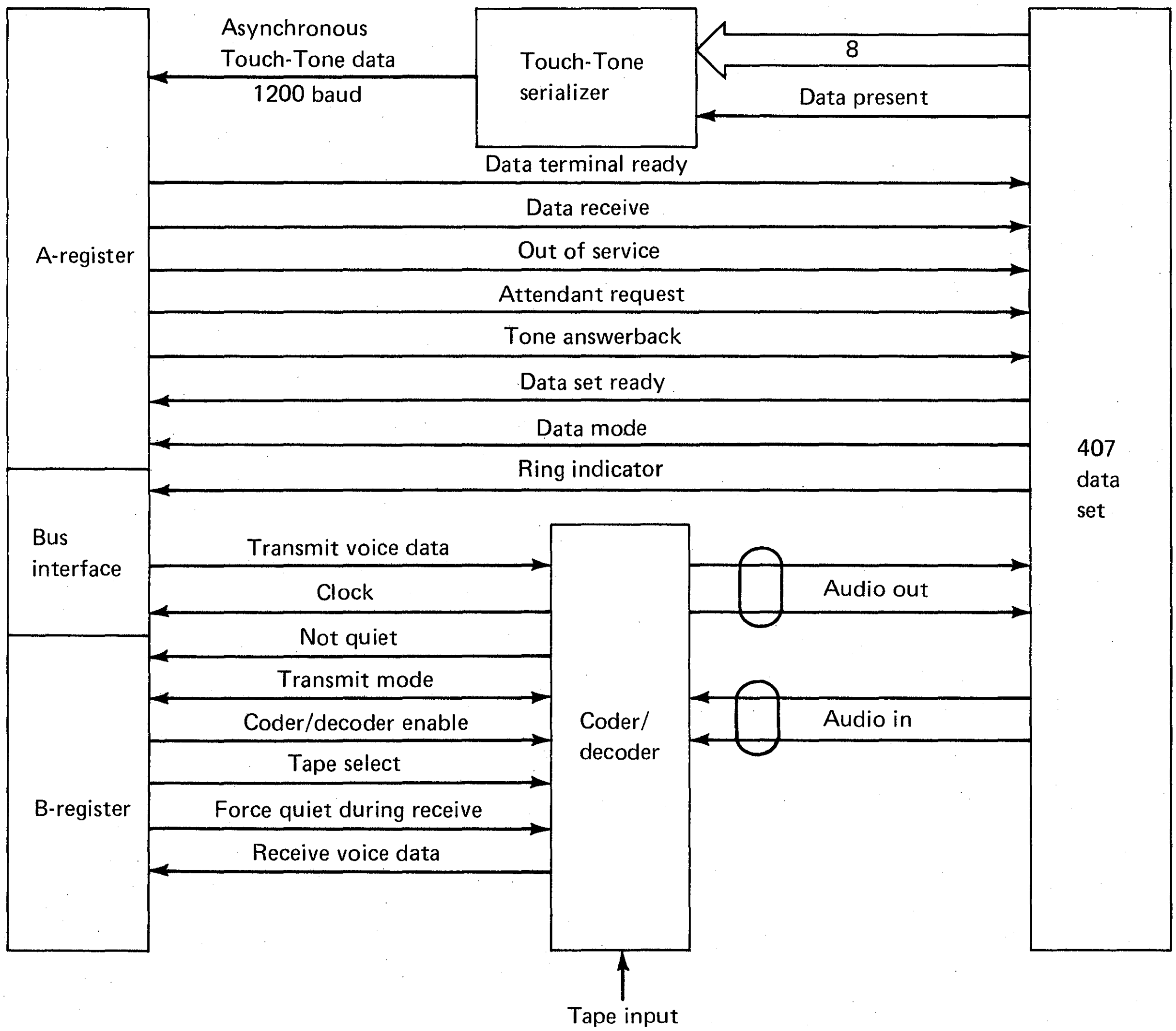


Figure 2-1. Audio Input/Output Feature interface lines

The control bus assignments are:

Control register A (even address)—407 data set

Bit	Write mode	Read mode
0	Data terminal ready	Data terminal ready (wrap-back)
1	Transmit mode	Transmit mode (wrap-back)
2	Request to send	Request to send (wrap-back)
3	Data receive	Touch-Tone data
4	Out of service	Data present*
5	Attendant request	Data set ready
6	Tone answer back	Data mode
7	Ring interrupt enable	Ring indicate (gated)

* Negative active

Control register B (odd address)—audio data

Bit	Write mode	Read mode
0	Data terminal ready (PCS required wrap)	Data terminal ready (wrap back)
1	Transmit mode	Transmit mode (wrap-back)
2	Request to send	Request to send (wrap-back)
3	Not assigned	Voice data
4	Not assigned	Not quiet
5	Tape select	Not assigned (tied active)
6	Coder/decoder enable	Not assigned (tied active)
7	Force quiet during receive	Not assigned (tied in active)

Tape select—when active, audio from tape input is selected for digitizing; when inactive, audio from the telephone input is selected for digitizing.

Coder/decoder enable—when active audio data is encoded or decoded; when inactive, in receive mode, an alternating 1,0 pattern is produced ('AA' or '55'). When inactive in transmit mode, quiet is produced on the audio output.

Force quiet during receive—should be inactive for normal operation.

Data Set to Feature Interface

Ring indicator—indicates that a ringing signal is presently being received.

Data set ready—indicates that the data set is either in the data mode and ready to receive data, to transmit answer-back signals, or both. The loss of 'data set ready' results in either an attention interrupt (if it occurs between data commands) or a modem interface error (if it occurs during data transfer).

Data mode—indicates that the data set is enabled to receive Touch-Tone data, and occurs three seconds after 'data receive' is active. If this bit is turned off during receive, a modem interface error results.

Data present—required by the feature card, and indicates that the data set is receiving a valid Touch-Tone signal.

Feature to Data Set Interface

Data terminal ready—prepares the data set to be connected to the communication line and maintains the connection once it is established.

*Data receive**—connects the Touch-Tone receiver in the data set to the line.

*Tone answerback**—generates an answer tone of 2025 Hz.

*Attendant request**—signals the associated equipment that operator intervention is required.

*Out of service**—makes the data set appear busy to incoming calls.

* These bits are controlled by the WRDC order.

A more detailed explanation of these controls is available in *Bell System Technical Reference Publication* 41408 (407A or B) and in Bell System Technical Reference Publication 41409 (407C).

Audio Input/Output

The audio device is a coder/decoder that can be configured and controlled by bits in the B-register to digitize audio data from the tape recorder/voice receive inputs or to generate audio signals from a digital data stream.

Audio signals presented at the tape input, when 'tape select' (B-register, write bit 5) is active, are digitized at the selected clock rate and presented synchronously to the scanner, in serial fashion, where they are loaded eight bits at a time and shifted to the right. Before the start of coding, the line should be placed in receive mode by either an ERM or SRT Order. 'Coder/decoder enable' (B-register, write bit 6) must be inactive for a minimum of *one full character time* to allow synchronization to occur on hexadecimal 'AA' or '55.' To code audio data, 'coder/decoder enable' must be made active after synchronization has occurred.

To select voice receive input from the telephone line, 'tape select' must be inactive.

When audio is not present or too low to be useful, the 'not quiet' bit (B-register, read bit 4) is inactive, forcing alternate 1's and 0's patterns in the data stream. This data appears as hexadecimal 'AA' or '55.' To skip over short, quiet periods, as occurs between syllables, a time-out of 'not quiet' is provided. Thus, when no audio is present, 'not quiet' remains active for approximately 200 milliseconds after the last useful audio input. With the resumption of useful audio input, 'not quiet' becomes immediately active or it remains active if a 200-millisecond time-out has not occurred.

The 'force quiet during receive' bit (B-register write bit 7), when made active during receive mode, immediately forces 'not quiet' to the inactive state and places alternate 1's and 0's patterns on 'voice data.' "Force quiet during receive" bit is inactive for normal operation.

Decoding the digital data (a reverse process) to produce the original audio signal requires that:

- Transmit mode is active—B-register, write bit 1 (SXMT Order)
- 'Coder/decoder enable' is active—B-register, write bit 6 (WRDC Order)

407 Data Set Control

Touch-Tone data is detected by the 407 data set and presented to the feature card in a two-out-of-eight code, which is serialized and passed to the scanner when 'data present' becomes inactive. Data is presented asynchronously to the scanner at a 1200-baud rate, high group first and then low group, in the following sequence: B4, B3, B2, B1, A4, A3, A2, and A1. These eight data bits are preceded by a start bit and followed by at least one stop bit. The frequency matrix used in the Touch-Tone phone is as follows:

High group

		B1	B2	B3	B4 (see Note)
Low group	A1	1	2	3	a
	A2	4	5	6	b
	A3	7	8	9	c
	A4	*	0	#	d

Note: B4 column not usually present.

Touch-Tone dial

For example, pressing button number 4 results in bits B1 and A2 being active; the eight data bits would be 00010010 when stripped of the start and stop bits, as seen by the controller. The resulting hexadecimal data in the 4987 subsystem is as shown in the following chart:

Touch-Tone key	**PCS data shift**	
	Right	**Left**
0	14	28
1	88	11
2	84	21
3	82	41
4	48	12
5	44	22
6	42	42
7	28	14
8	24	24
9	22	44
*	18	18
#	12	48
a	81	81
b	41	82
c	21	84
3	11	88

The data set is capable of receiving 10 characters per second, which is the limiting factor in data rate transfer.

To receive Touch-Tone data (enable the 407 to do so), 'data receive' (A-register write bit 3) must be active.

Vocabulary Generation Tips

Recording Equipment

A high-quality tape recorder and microphone should be used. The recorder must be capable of driving a 0-dBm signal level into an 8-k ohm load. The 6.1-m (20-ft) audio input cable attaches to the recorder with a standard phono plug provided with RPQ D02337.

Voice Recording

The recording should be made in a reverberant-free room, with the microphone held close to the mouth to minimize background noise.

During the voice tape recording, allow 2 or 3 seconds between messages; this allows manual start and stop using keyboard entry.

Speak in a slow, clear manner, and maintain the volume level meter peaks at 0 dBm on the tape recorder. If the recorded vocabulary is produced by a professional experienced in voice tape recording, the quality of the digitized vocabulary will be enhanced.

Editing Audio Data

In general, the only editing of the audio record that should be done is the truncation of the end of the phrase. After the phrase is recorded, as described earlier, the sectors of quiet at the end of the file should be removed.

The user might consider a weighted censoring routine (if a sector is 90% quiet, it should be deleted to compensate for possible background noise). Because the audio record is highly bit sensitive, do not remove any data, including quiet (intraphase), from the middle of phrases; inflection and impact is changed if a pause is shortened.

Vocabulary Dividing Points

If messages are long enough to require chaining of data buffers, higher quality audio can be produced if the transitions between buffers are made at a quiet part of the speech. These transitions can be determined experimentally, or the following algorithm may be used to determine the relative loudness of the audio.

Loudness approximation algorithm:

1. Starting at a quiet portion of the speech, examine the digitized audio data bit by bit. Add 1 to a counter each time a sequence of three 1's or three 0's occurs. Example:

Data	0011010000010111110001
Counter	0- - - - - - - 123- - - - 45- - 6-

2. Every 42 bytes, divide the counter by 2 (shift right 1). The counter now contains an approximation of the loudness of the audio.

Vocabulary Storage

To digitize audio information from either tape or phone, the user must first allocate a data set on disk. Then, using the facilities of chained tables and program-controlled interrupt (PCI), a Read command is issued to the Programmable Communications Subsystem (PCS) specifying the odd-address line of the pair, and referencing the appropriate function string on the PCS. When there is no sound presented to the audio unit, a string of 1's and 0's (101010) is generated; thus, a series of bytes containing "A's" or "5's" indicates silence. Typical function strings should not begin transferring information to the Series/1 until a byte of non-quiet information is encountered. Once data transfer starts, the data is streamed out of the disk until either a byte count is satisifed or an activity timer truncates the operation. The function string could use either a buffer count (how many buffers are transferred) or a timer to terminate the operation.

Sample Function Strings

The controller directs the transfer of data between the Series/1 processor storage and the remote communication facility by a series of orders called function strings. Differences in coding the function strings may affect the order timing.

All audio lines should be opened first; then the 407 data set lines should be opened. This allows the audio lines to occupy the high priority positions of the scan table. The scan table directs the scanner when and how often in the scan cycle each particular device address is serviced. After all lines are open, the audio lines may be closed and opened as desired.

Examples of audio I/O function strings are as follows:

```
*-----------------------------------------------------------------------*
*     4 0 7   D A T A S E T   O P E N   D A T A   B L O C K             *
*-----------------------------------------------------------------------*
*
          1    LINE DESCRIP.|  SCAN RATE        :    1701
          2    BIT RATE     |  FAT ENTRY        :    0800
          3    CCT ENTRY    |  DATA LINK TYPE   :    0040
          4    BCC TYPE     |                   :    8000
          5          CRC POLYNOMIAL             :    0000
          6           CRC CONSTANT              :    0000
          7    SYNC CHAR.   |  ACU ADDRESS      :    0000
          8    TIMER 0      |  TIMER 1          :    0014
          9    TIMER 2      |  TIMER 3          :    9E3C
         10    TIMER 4      |  TIMER 5          :    2802
         11    POLL CHAR. 1 |  POLL CHAR. 2     :    0000
         12    ADDR CHAR. 1 |  ADDR CHAR. 2     :    0000
         13    POLL-ADDR CNT|  RESERVED         :    0000
*
*
*
*-----------------------------------------------------------------------*
*     A U D I O   O P E N   D A T A   B L O C K                         *
*-----------------------------------------------------------------------*
*
*
          1    LINE DESCRIP.|  SCAN RATE        :    0708
          2    BIT RATE     |  FAT ENTRY        :    0000
          3    CCT ENTRY    |  DATA LINK TYPE   :    0010
          4    BCC TYPE     |  PARITY MASK      :    8000
          5           CRC POLYNOMIAL            :    0000
          6            CRC CONSTANT             :    0000
          7    SYNC CHAR.   |  ACU ADDRESS      :    5500
          8    TIMER 0      |  TIMER 1          :    0014
          9    TIMER 2      |  TIMER 3          :    9E3C
         10    TIMER 4      |  TIMER 5          :    2802
         11    POLL CHAR. 1 |  POLL CHAR. 2     :    0000
         12    ADDR CHAR. 1 |  ADDR CHAR. 2     :    0000
         13    POLL-ADDR CNT|  RESERVED         :    0000
```

```
*---------------------------------------------------------------------*
*                                                                     *
*             S E R I E S / 1   A U D I O   I / O                     *
*          S A M P L E   F U N C T I O N   S T R I N G                *
*                                                                     *
*---------------------------------------------------------------------*
        ORG   MAIN+X'9200'
AUDIO   DC    128X'0'             RESERVED AREA
        DC    A(FIT)              FUNCTION INDEX TABLE ADDR
        DC    A(CIT)              CONTROL CHAR TABLE INDEX
*
FIT     DC    A(1)                COUNT=1
        DC    A(FAT)              ADDR OF FUNCTION ADDR TABLE
*
CIT     DC    A(1)                COUNT=1
        DC    A(CCT)              ADDR OF CONTROL CHAR TABLE
*
*---------------------------------------------------------------------*
*          F U N C T I O N   A D D R E S S   T A B L E                *
*---------------------------------------------------------------------*
FAT     DC    A(9)                9 FUNCTION STRINGS
        DC    A(ANSWER)           ID=0     WAIT FOR RING
        DC    A(READTONE)         ID=1     READ TOUCH-TONE DATA
        DC    A(WRITEVC)          ID=2     TRANSMIT VOICE
        DC    A(TAPEI)            ID=3     READ TAPE INITIAL
        DC    A(PHONEI)           ID=4     READ PHONE INITIAL
        DC    A(READC)            ID=5     READ CONTINUE
        DC    A(BEEP)             ID=6     TRANSMIT TONE
        DC    A(HANGUP)           ID=7     DISCONNECT
        DC    A(DSTONE)           ID=8     DATASET TONE
*
*---------------------------------------------------------------------*
*          C O N T R O L   C H A R A C T E R   T A B L E              *
*---------------------------------------------------------------------*
CCT     DC    AL1(6)              6 ENTRIES
        DC    X'18'               0 = *  (END OF TOUCH-TONE INPUT)
        DC    X'12'               1 = #  (ERROR/RETRY SIGNAL)
        DC    X'55'               2 = SYNC CHARACTER
        DC    X'AF'               TONE SEQ CHARACTER 1
        DC    X'F5'               TONE SEQ CHARACTER 2
        DC    X'00'               TONE SEQ CHARACTER 3
*
*---------------------------------------------------------------------*
*                    E Q U A T E S                                    *
*---------------------------------------------------------------------*
STAR    EQU   0                   *DISPLACEMENT INTO CCT
BOX     EQU   1                   #DISPLACEMENT INTO CCT
SYNC    EQU   2                   X'55' DISPLACEMENT INTO CCT
BEEP1   EQU   3                   TONE
BEEP2   EQU   4                      SEQUENCE
BEEP3   EQU   5                             CHARACTERS
*
TIMER0  EQU   0                   RING TIMER     (0)             F'0'
TIMER1  EQU   1                   TONE TIMER     (1 SEC)         F'20'
TIMER2  EQU   2                   ACTIVITY TIMER      (30 SEC)   F'600'
TIMER3  EQU   3                   XMIT/RECEIVE TIMER  (3 SEC)    F'60'
TIMER4  EQU   4                   TONE TIMER  (2 SEC)            F'40'
TIMER5  EQU   5                   DISCONNECT TIMER  (100 MSEC)   F'2'
TIMER6  EQU   6                       -NOT USED-
TIMER7  EQU   7                       -NOT USED-
```

```
*---------------------------------------------------------------------*
*              W A I T   F O R   R I N G     ( 0 )                    *
*                                                                     *
*   THIS FUNCTION STRING ENABLES THE AUTO ANSWER MECHANISM AND        *
*   WAITS INDEFINITELY (TIMER0 = 0) FOR AN INCOMING CALL. WHEN        *
*   THE RING INTERRUPT OCCURS, THE PROGRAM ANSWERS THE PHONE          *
*   AFTER TWO RINGS, DELAYS FOR ONE SECOND WHILE THE DATA SET         *
*   SENDS AN ANSWER TONE, AND THEN POSTS THE SERIES/1 PROGRAM.        *
*                                                                     *
*---------------------------------------------------------------------*
ANSWER   EQU   *
         ERI   1,TIMER0             ENABLE RING (WAIT FOREVER)
         ERM   ,                    GO TO RECEIVE MODE
         DLY   TIMER1               ALLOW DATA SET TO SEND TONE
         WRDC  X'10'                TURN ON 407 DATA RECEIVE
         LDCBI 0                    POST DEV END (OR CHAIN)
*---------------------------------------------------------------------*
*             R E A D   T O U C H T O N E    ( 1 )                    *
*                                                                     *
*   THIS FUNCTION STRING READS TOUCH-TONE DIGITAL INPUT UNTIL A       *
*   TERMINATOR CHARACTER (*) IS RECEIVED OR THE BYTE COUNT IN         *
*   DCB GOES TO ZERO.  IF A DELETE CHARACTER (#) IS ENTERED, THE      *
*   BUFFER ADDRESS IS RESET TO THE BEGINNING AND THE CALLER CAN       *
*   BEGIN REENTERING DIGITS IMMEDIATELY.  AN ACTIVITY TIMER IS        *
*   USED TO LIMIT THE TIME BEFORE THE FIRST DIGIT IS ENTERED AND      *
*   BETWEEN SUBSEQUENT DIGITS TO 30 SECONDS -- IF A TIMEOUT OCCURS,   *
*   AN EXCEPTION IS POSTED TO THE SERIES/1 PROGRAM.                   *
*                                                                     *
*---------------------------------------------------------------------*
READTONE EQU   *
         SRT   TIMER3               GO TO RECEIVE MODE
         SAT   TIMER2               START ACTIVITY TIMER
*                                   (30 SECONDS BETWEEN DIGITS)
         MST   2,DIGITS             MONITOR INPUT FOR * OR #
         RDTXT ,                    READ TOUCH-TONE INPUT
         PXEI  1                    TOO MANY CHARACTERS
*
END      EQU   *                    * RECEIVED --
         STCHR NOBCC                STORE * IN BUFFER
         PDEI  0                    AND POST DEVICE END
*
DEL      EQU   *                    #RECEIVED --
         RIDBF ,                    RESET BUFFER ADDRESS AND COUNT
         RDTXT ,                    AND RESTART READ
         PXEI  1                    TOO MANY CHARACTERS
*
*---------------------------------------------------------------------*
*          S T A T E   B R A N C H   T A B L E                        *
*---------------------------------------------------------------------*
DIGITS   DC    A(STAR)              IF * RECEIVED,
         DC    A(END)               GO TO END
         DC    A(BOX)               IF # RECEIVED,
         DC    A(DEL)               GO TO DEL
*
*
*---------------------------------------------------------------------*
*             T R A N S M I T   V O I C E    ( 2 )                    *
*                                                                     *
*   THIS FUNCTION STRING PERFORMS A VOICE OUTPUT OPERATION.  DATA     *
*   WILL BE CONTINUOUSLY TRANSMITTED AS LONG AS THE CHAIN BIT IS      *
*   ON IN THE USER'S DCB.                                             *
*                                                                     *
*   NOTE: THE DEVICE CONTROL REGISTER IS RESET BETWEEN DCB CHAINS     *
*         TO PREVENT THE AUDIO OUTPUT FROM TRANSMITTING NOISE (FF'S), *
*         AND THE LINE IS PLACED IN RECEIVE MODE BEFORE ISSUING       *
*         LDCBI TO PRIORITIZE THE CHAIN OPERATION.                    *
*                                                                     *
*---------------------------------------------------------------------*
WRITEVC  EQU   *                    WRITE VOICE
         SXMT  TIMER3,TIMER0        ENTER TRANSMIT MODE
         XMI   1,SYNC,NOBCC         SEND QUIET
         WRDC  X'02'                ENABLE CODER/DECODER
         XMTXT ,                    SEND MESSAGE
         ERM   ,                    GO TO RECEIVE MODE
         WRDC  X'00'                DISABLE CODER/DECODER WHILE CHAINING
         LDCBI 0                    CHAIN TO NEXT DCB
```

```
*---------------------------------------------------------------------*
*               R E A D   F R O M   T A P E ( 3 )                     *
*                                                                     *
*   THIS FUNCTION STRING READS VOICE FROM TAPE INPUT.  THE 1ST        *
*   DCB MUST SPECIFY FUNCTION ID 3, AND CHAINED DCB'S MUST SPECIFY    *
*   FUNCTION ID 5 (READ CONTINUE).                                    *
*                                                                     *
*---------------------------------------------------------------------*
TAPEI    EQU   *                     READ FROM TAPE
         ERM   ,                     ENTER RECEIVE MODE
         WRDC  X'00'                 RESET DEVICE CONTROL REGISTER
         RDSYN 1                     SYNC UP
         WRDC  X'06'                 ENABLE CODER/DECODER, SELECT TAPE
         JMP   ERM                   CONTINUE
*
*---------------------------------------------------------------------*
*               R E A D   F R O M   P H O N E ( 4 )                   *
*                                                                     *
*   THIS FUNCTION STRING PROVIDES THE ONLINE VOICE RECORDING          *
*   FUNCTION.  THE 1ST DCB MUST SPECIFY FUNCTION ID 4, AND            *
*   CHAINED DCB'S MUST SPECIFY FUNCTION ID 5 (READ CONTINUE).         *
*                                                                     *
*---------------------------------------------------------------------*
PHONEI   EQU   *                     READ FROM PHONE
         ERM   ,                     ENTER RECEIVE MODE
         WRDC  X'00'                 RESET DEVICE CONTROL REGISTER
         RDSYN 1                     SYNC UP
         WRDC  X'02'                 SET DEV CTL REG
*
*---------------------------------------------------------------------*
*                                                                     *
*   THE FOLLOWING CODE INITIATES A READ OPERATION (TAPE OR PHONE      *
*   INPUT).                                                           *
*                                                                     *
*   ONCE SYNC IS ESTABLISHED, DATA IS READ AND DISCARDED A CHARAC-    *
*   TER AT A TIME UNTIL 'NON-QUIET' IS DETECTED.  THIS CODE SHOULD    *
*   BE DELETED TO ACHIEVE MAXIMUM THROUGHPUT.                         *
*                                                                     *
*---------------------------------------------------------------------*
ERM      EQU   *
SEARCH   EQU   *                     DISCARD LEADING QUIET
         RDCHR ,                     CONTINUE READING 55'S
         CIBE  SYNC,SEARCH           UNTIL VOICE DETECTED
*
*
*
*
*---------------------------------------------------------------------*
*               R E A D   C O N T I N U E   ( 5 )                     *
*                                                                     *
*   READ VOICE DATA.  VOICE INPUT IS READ INTO THE CURRENT            *
*   BUFFER UNTIL IT IS FILLED; THEN THE NEXT DCB IS LOADED.  IT IS    *
*   ASSUMED THAT THE BUFFER SIZE IS 1280 BYTES.                       *
*                                                                     *
*---------------------------------------------------------------------*
READC    EQU   *                     CONTINUE RECORD OPERATION
         RDREC 1280                  READ FULL INPUT BUFFER
         LDCBI 0                     CHAIN TO NEXT DCB
*
*
*
*
*
```

```
*---------------------------------------------------------------------*
*                   T R A N S M I T   T O N E                         *
*                     U S I N G   A U D I O    ( 6 )                 *
*                                                                     *
*   THIS FUNCTION STRING SENDS A SHORT TONE USING AUDIO OUTPUT.       *
*   THE TIME DURATION OF THE TONE CAN BE ADJUSTED BY MODIFYING THE    *
*   TIMER4 VALUE. (TIMER4 IS USED EXCLUSIVELY FOR THIS PURPOSE.)      *
*                                                                     *
*   NOTE:  LDCBI IS USED TO TERMINATE IN ORDER TO ALLOW CHAINING.     *
*                                                                     *
*---------------------------------------------------------------------*
BEEP     EQU   *
         SXMT  TIMER3,TIMER0        GO TO TRANSMIT MODE
         XMI   1,SYNC,NOBCC         SEND '55'  ?
         SIT   TIMER4,STOP          START TONE TIMER
         WRDC  X'02'                ENABLE CODER/DECODER
*
BEEPC    EQU   *                    CONTINUE SENDING TONE SEQUENCE
         XMCS  3,TONESEQ             UNTIL
         BCH   BEEPC                  TIMER EXPIRES
*
STOP     EQU   *
         WRDC  X'00'                DISABLE CODER/DECODER
         ERM   ,                    ENTER RECEIVE MODE
         LDCBI 0                    CHAIN
*
TONESEQ  DC    AL1(BEEP1)           CONTROL
         DC    AL1(BEEP2)            SEQUENCE
         DC    AL1(BEEP3)             LIST
         DC    AL1(0)                  EXTRA BYTE FOR ALIGNMENT
*
*
*
*
*
*
*---------------------------------------------------------------------*
*                    H A N G   U P            ( 7 )                   *
*                                                                     *
*  DISCONNECT LINE. (CONTROLLER WAITS FOR TIMEOUT BETWEEN ACTIVA-     *
*  TING DTR AND CHECKING FOR DSR.)  IF DSR STILL ACTIVE WHEN TIMER    *
*  EXPIRES, AN EXCEPTION INTERRUPT IS POSTED WITH ISB BIT 0 = 1.      *
*                                                                     *
*  NOTE:    LDCBI IS USED TO TERMINATE IN ORDER TO ALLOW CHAINING     *
*              TO THE 'ANSWER' FUNCTION IF DESIRED.                   *
*                                                                     *
*---------------------------------------------------------------------*
HANGUP   EQU   *
         DISC  TIMER5               ALLOW 100 MS FOR DISCONNECT
         LDCBI 0                    POST DEVICE END OR CHAIN
*---------------------------------------------------------------------*
*                 T R A N S M I T   T O N E                           *
*                   F R O M   D A T A S E T    ( 8 )                 *
*                                                                     *
*  THIS FUNCTION STRING CAUSES THE 407 DATASET TO TRANSMIT A TONE     *
*  FOR THE TIME SPECIFIED BY TIMER4.                                  *
*                                                                     *
*---------------------------------------------------------------------*
DSTONE   EQU   *
         WRDC  X'12'                TURN ON TONE ANSWER BACK
         DLY   TIMER4               TONE DURATION
         WRDC  X'10'                TURN OFF TONE ANSWER BACK
         LDCBI 0
         END
```

Chapter 3. Maintenance Information

The Audio Input/Output Feature card plugs into the IBM Series/1 4987 Programmable Communications Subsystem, which consists of the scanner and the backplane, as shown in Figure 3-1.

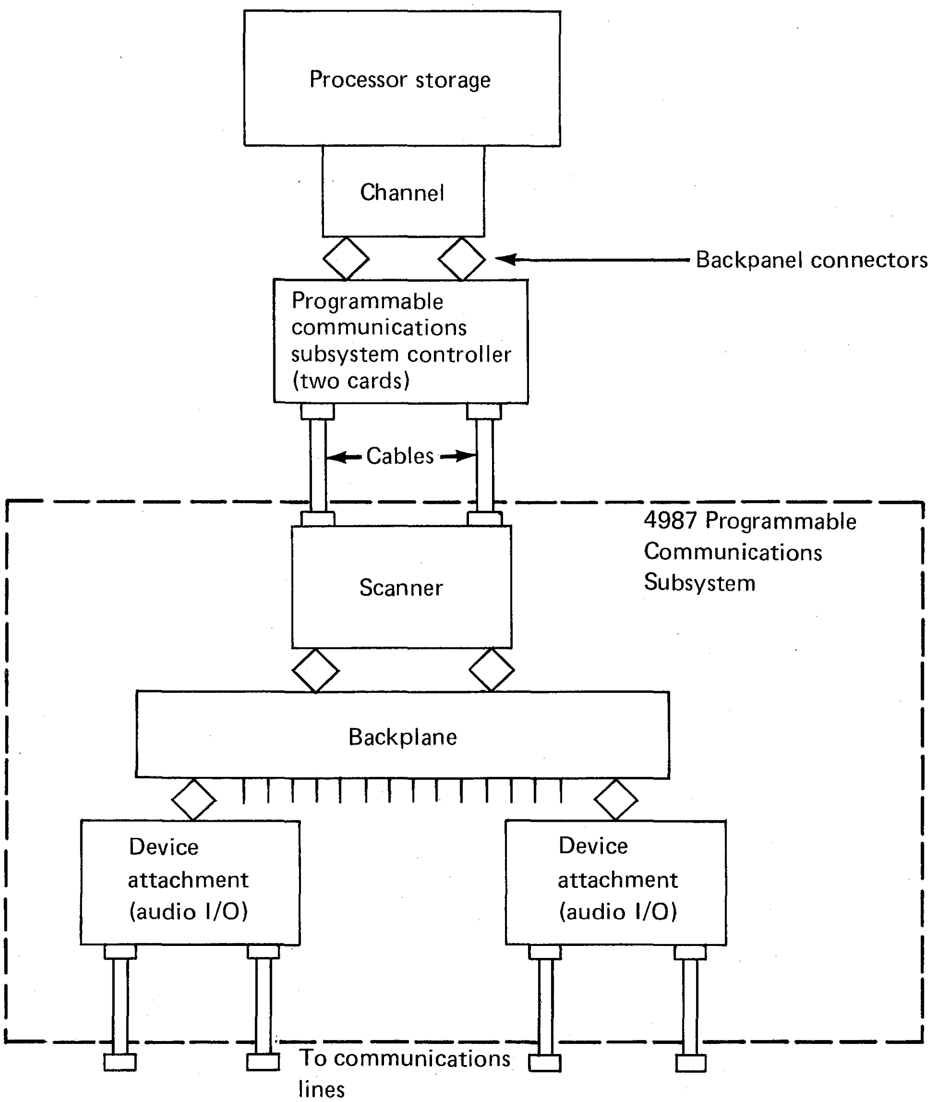


Figure 3-1. Data communications system block diagram

Scanner

The scanner performs the multiplexing and serializing/deserializing functions for the subsystem. The scanner contains storage and logic functions that allow it to be loaded with control parameters. Using these parameters, the scanner addresses each active line, in turn, and assembles characters (on receive) or sends bits (on transmit). The scanner also signals the controller when a character is needed for transmission or when a character has been assembled and is available for storing. Up to two scanners can be housed in the subsystem unit; the second scanner is called an Expansion Scanner feature. (If two scanners are used, a second controller must be used for the second scanner.)

Subsystem Unit

The subsystem unit houses up to two scanners and up to 16 device attachment feature cards. It also contains the power supply for these components. The scanners are plugged into the subsystem unit backplane from the front; the device attachments plug into the backplane from the rear. Device pin assignments are:

- All data and control signals on the Audio Input/Output Feature card are minus (−) active.
- Auxiliary bus bits are not used by this feature.
- Control bus bits are bidirectional signals that convey basic command and response information for each device. The control bus assignments are defined in Chapter 2.
- 'Device select A' and 'device select B' control registers A and B, respectively.
- 'Card select' is a decoded address signal that is distributed by card file backplane wiring.
- 'Group select' effectively divides the subsystem backplane into two halves: connectors 1–8 are group select low; connectors 9–16 are group select high. When only one scanner is installed, the signals to group select low are routed through a mid-panel jumper card. When two scanners are installed, the jumper card is removed, thereby connecting scanner A to connectors 1–8 and scanner B to connectors 9–16.

Pin B-row	Signal name
1	Ground
2	Spare
3	Aux bus bit 0
4	Aux bus bit 2
5	Aux bus bit 4
6	Aux bus bit 6
7	Aus bus bit 5
8	Time pulse (153.6 kHz)
9	1200-Hz clock
10	4800-Hz clock
11	Spare
12	Spare
13	+5V PWR
14	Control bus bit 0
15	Control bus bit 2
16	Control bus bit 4
17	Control bus bit 6
18	Send data (space)
19	Device reset
20	Control read
21	−5V PWR
22	−8.5V PWR
23	+8.5V PWR
24	+5V PWR
25	Control write
26	Device select A
27	Ground
28	Card select

Pin A-row	Signal name (component side)
1	Ground
2	Spare
3	Aux bus bit 1
4	Aux bus bit 3
5	50-ms time P
6	Aux bus bit 7
7	300-Hz clock
8	600-Hz clock
9	2400-Hz clock
10	9600-Hz clock
11	Spare
12	Spare
13	+5V PWR
14	Control bus bit 1
15	Control bus bit 3
16	Control bus bit 5
17	Control bus bit 7
18	Send strobe
19	Clock in
20	Aux read
21	−5V PWR
22	−8.5V PWR
23	+8.5V PWR
24	+5V PWR
25	Aux write
26	Device select B
27	Ground
28	Group select

Functional Units

The Audio Input/Output Feature accepts Touch-Tone or voice input and responds with voice output. Figure 3-2 is a block diagram of the feature card. The arrows show the direction of the signals; the numbers indicate the number of lines on the bus or cable.

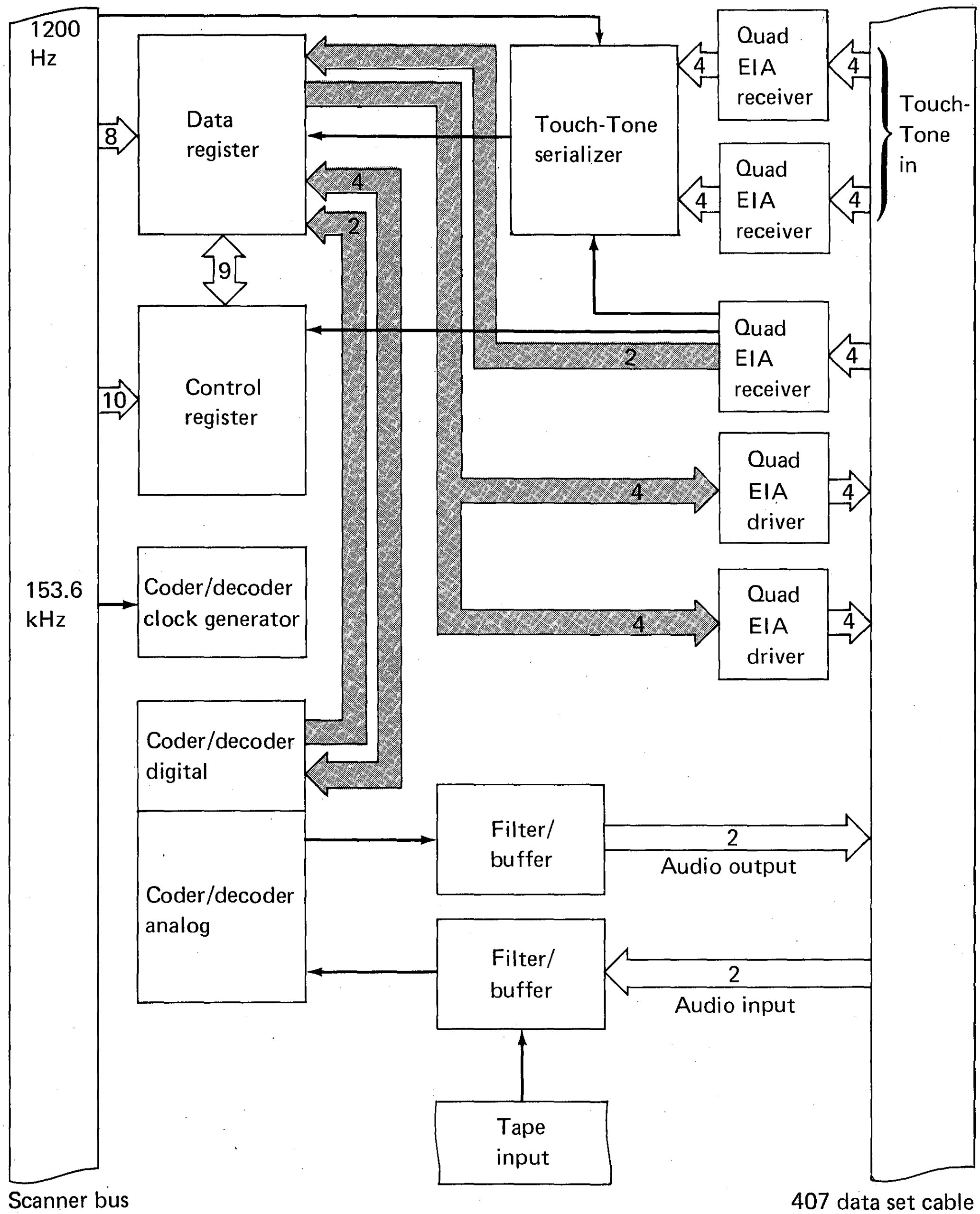


Figure 3-2. Functional block diagram

Coder/Decoder Description

Data Rate

The quality of the audio signal presented to the data set is directly proportional to the selected data sampling rate. The sampling rate is selected manually by a switch on the feature card, as shown in Figure 3-3 and Table 2.

Data sampling rate (kHz)	Switch settings *SA*	*SB*	*SC*
19.20	ON	ON	ON
17.07	OFF	ON	ON
15.36	ON	OFF	ON
13.96	OFF	OFF	ON
12.80	ON	ON	OFF
11.82	OFF	ON	OFF
10.97	ON	OFF	OFF
10.24	OFF	OFF	OFF

Table 2. Data rate switch

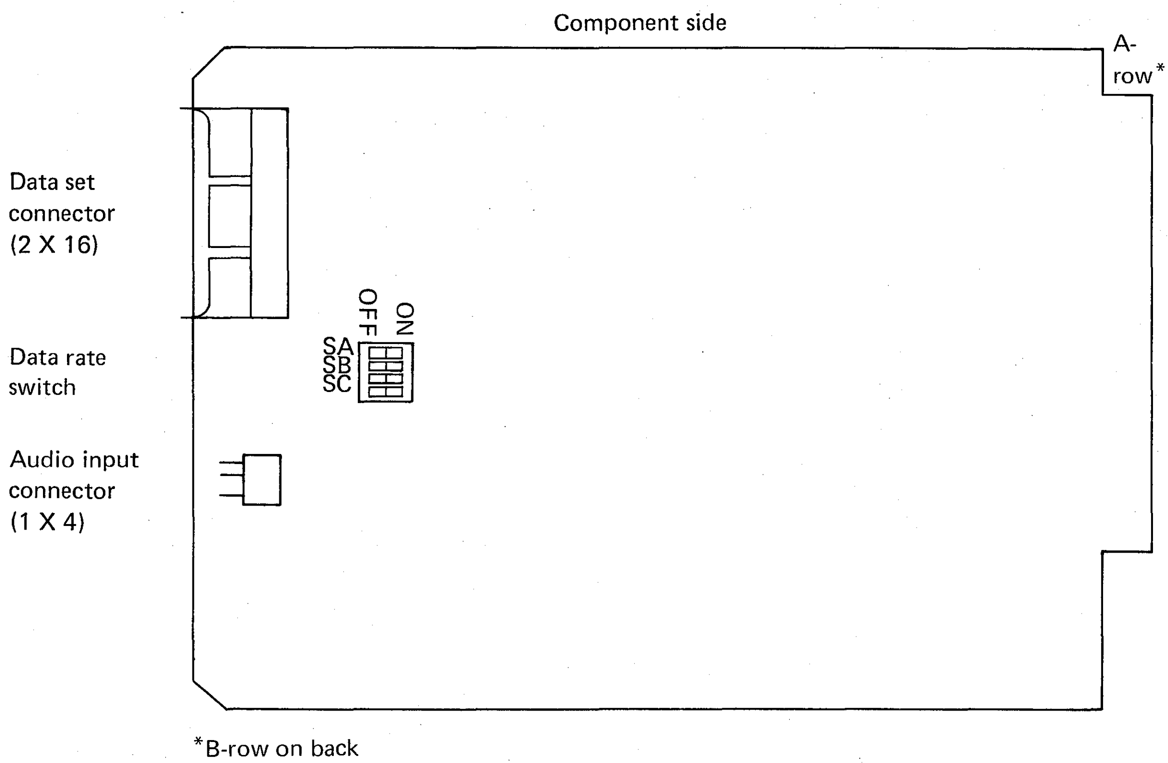
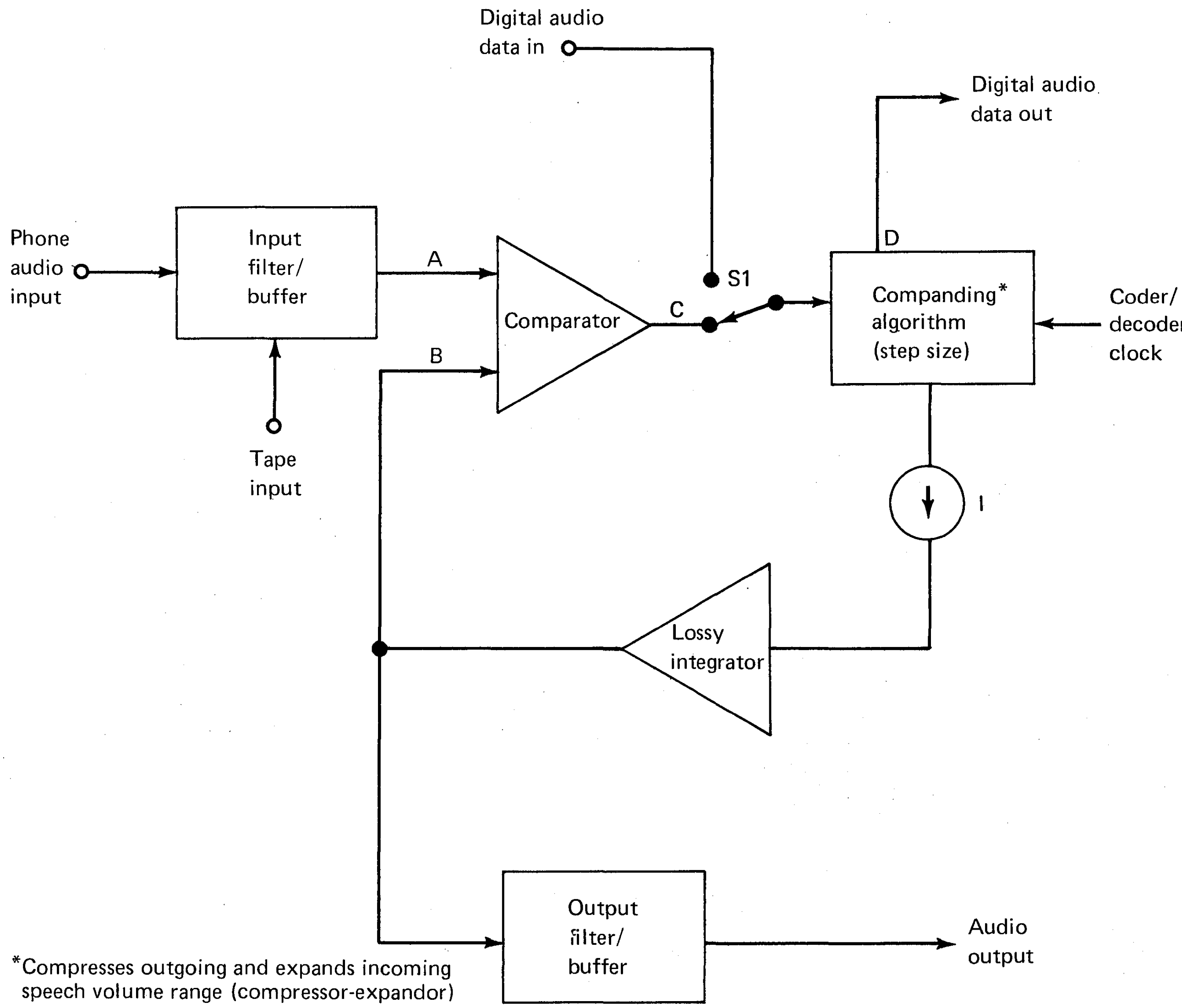


Figure 3-3. Audio I/O Feature card

Coder/Decoder Circuit

The Audio Input/Output coder/decoder is a device of mixed analog and digital circuitry capable of quantizing in time (coding) audio voice signals. The resulting output is a digitized stream of 1's and 0's that contains all the coded information necessary to recreate (decode) the original audio signal. Decoding is accomplished by using the same circuitry used in the coding process. The coder/decoder can be represented as shown by the block diagram in Figure 3-4.



*Compresses outgoing and expands incoming speech volume range (compressor-expandor)

Figure 3-4. Coder/decoder

Audio input (analog) is passed through a buffer network that:

- Terminates the input signal
- Adjusts the level by amplification
- Filters out low frequencies (under 300 Hz) that are not reproducible by telephone equipment
- Filters out high frequencies (above 3.3 kHz) that the audio I/O card cannot code

The resulting signal (A) is then connected to the positive input of the comparator. There is a continuous feedback loop in the coder/decoder, and input (B) of the comparator is tracking input (A) as a starting point.

With signals (A) and (B) present, the comparator output (C) is either a 1 (up level), when (A) is greater than (B), or a 0 (down level), when (A) is less than (B). Output (C) is sampled by the companding algorithm block at the selected clock rate. The resulting bit stream (D) is the digital stream of coded 1's and 0's. Each bit determines the polarity of the current (I) input to the lossy integrator. The 1's produce positive current; 0's produce negative current. Thus, when comparator input (A) is more positive than input (B), a stream of 1's is created; positive current is supplied to the integrator, forcing input (B) more positive until it matches the value at input (A). This is a continuous process of (B) "racing" to catch (A). In a constant-slope system, the value of current (I) remains fixed; this is called a continuous slope delta modulator. Decisions occurring at the comparator force signal (B) to approach signal (A).

The audio I/O card uses a more advanced technique, whereby the step-size current (I) is adjusted continuously. This improves the ability of input (B) to track the rapidly changing (A) inputs. This is accomplished by monitoring the past history of the bit stream and detecting coincidental bit sequences. In the audio I/O design, a coincidence of three 1's or 0's in sequence is detected and passed to a syllabic integrator, which forms part of the companding algorithm. The syllabic integrator then determines the magnitude of the current (I).

At high sampling rates, smaller time lapses occur between adjustments to (B), and signal (B) is adjusted more often. Thus, audio quality (fidelity) improves with higher sampling rates.

Decoding is accomplished by presenting the digital audio data to the companding algorithm logic via logic function switch S1 (shown in transmit mode). Since the original digital data was created using the same circuitry, the audio signal is identical to the one present during the coding process. The resulting audio output, after filtering, is a good reproduction of the original audio input.

Appendix A. Reference Information

A 25-pin wrap connector that allows testing all drivers and receivers as well as the Touch-Tone and audio I/O functions, is supplied in the RPQ D02337 ship group. The audio I/O wrap connector is illustrated in Figure A-1.

Figure A-2 illustrates the card connectors.

Connector pin	Line
3	A1 low group
5	A3 low group
9	B1 high group
11	B3 high group
25	Out of service
4	A2 low group
6	A4 low group
10	B2 high group
12	B4 high group
13	Spare
8	Voice receive
20	Tone answerback
14	Ring indicator
15	Attendant request
16	Data present
21	Data receive
17	Voice answerback
19	Data mode
22	Data terminal ready
23	Data set ready

Figure A-1. Audio I/O wrap connector

Parts

Audio cable – 6839411
Wrap connector – 6839412
Data set cable – 6839410

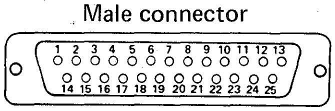
Signal connections for modem

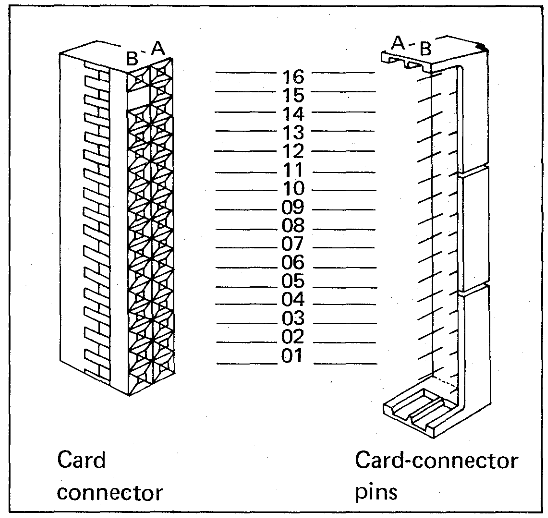
1	No connection (407 frame ground)	A01
2	Voice receive	A02
3	A1 low group*	A03
4	A2 low group*	A04
5	A3 low group*	A05
6	A4 low group*	A06
7	No connection	A07
8	Voice receive	A08
9	B1 high group*	A09
10	B2 high group*	A10
11	B3 high group*	A11
12	B4 high group*	A12
13	Spare	A13
14	Ring indicator**	A14
15	Attendant request**	A15
16	Data present*	A16
17	Voice answerback	B01
18	Voice answerback	B02
19	Data mode**	B03
20	Tone answerback**	B04
21	Data receive**	B05
22	Data terminal ready**	B06
23	Data set ready**	B07
24	Signal ground	B08
25	Out of service**	B09

*Negative active

**Postive active

Male connector





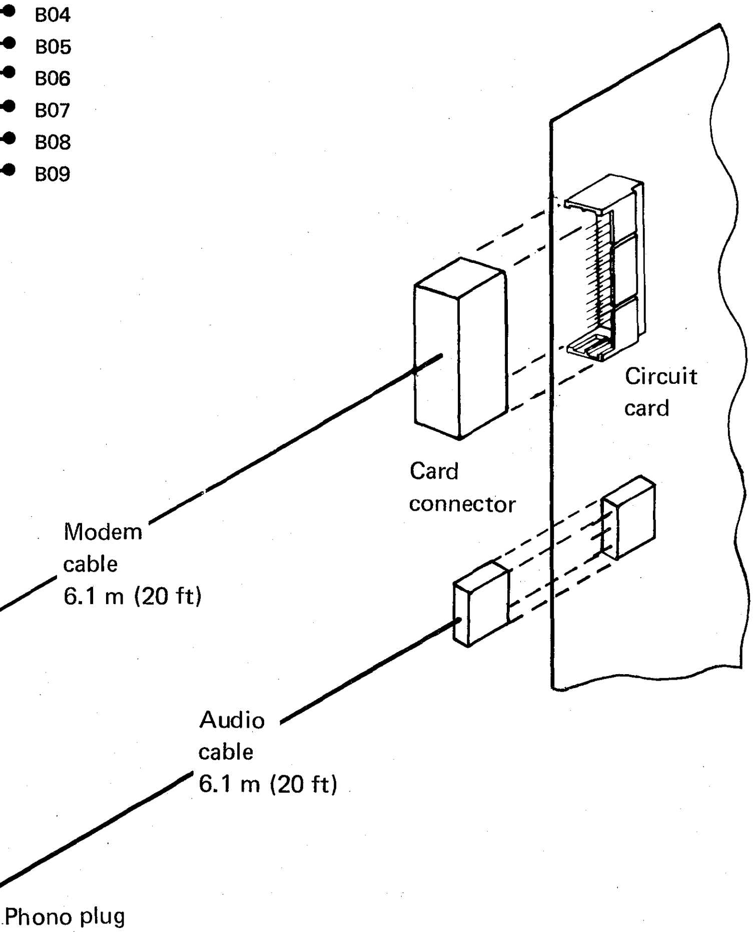


Figure A-2. Card connectors

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Custom Feature

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