Operation

OMNI SI®

Purpose

This Technical Practice has been prepared for FGBS Employees who operate and maintain the equipment manufactured and sold by Fujitsu GTE Business Systems, Inc. The information in this Practice is subject to change and may not be suitable in all situations. Fujitsu GTE Business Systems, Inc. acknowledges that a customer's special requirements or practices may take precedence over those supplied in this Practice if a conflict develops during installation or ongoing operation. Fujitsu GTE Business Systems, Inc. hereby disclaims any responsibility or liability for any consequential or inconsequential damages that may result from the use of this practice.

OMNI SI® SVR 5210 Technical Practices

This FGBS Practice is part of a series of Practices for the FGBS OMNI SI, System Version Release 5.2.1.0. The series includes the following:

TL-130000-1001	System Description/Features
TL-130100-1001	Operation
TL-130200-1001	Maintenance
TL-130300-1001	Installation
TL-130400-1001	Data Base
TL-130500-1001	System Configuration
TL-130600-I 001	5120 to 5210 Upgrade
TL-130700-1001	Index
TL-100000-1001	ADMP User's Guide

USER RESPONSE/REQUEST FORM

Send To: Fujitsu GTE Business Systems
Publications Manager, Dept. 443
2411 West 14th Street
Tempe, Arizona 85281-9904

Date Submitted: ________________

- Document Number: Section TL-130100-1001
- Document Title: OMNI SI Operation
- issue Number: 1 Date: August 1987

☐ General Comments. Enter Comments concerning overall organization, presentation, or content of this document in the space below.

☐ Suggested Revisions. Enter specific additions deletions or changes to this document in the space below, or provide a marked up copy of the original document.

Paragraph, Figure, or Table Number Comment

- Revision Priority ☐ 1: Service Affecting Problem
  ☐ 2: Potentially Service Affecting Problem
  ☐ 3: Administrative, Documentation, and Other Maintenance Problems
- Please attach a copy of your marked up pages.
- May we contact you concerning your comments? ☐ YES ☐ NO
- May we send you information on other available publications? ☐ YES CI NO
- Please enter your:

Name: ____________________
Title: ____________________
Company/Department: ____________________
Address: ____________________
Area Code/Tel. No./Ext.: ____________________

Thank you for your cooperation and comments.

FOLD OR TEAR HERE CAREFULLY. DOCUMENT WILL BE SCANNED BY OCR EQUIPMENT

FOLD HERE CAREFULLY. DOCUMENT WILL BE SCANNED BY OCR EQUIPMENT

NO POSTAGE NECESSARY IF MAILED IN THE UNITED STATES

BUSINESS REPLY MAIL

FIRST CLASS PERMIT NO. 216 TEMPE, AZ

POSTAGE WILL BE PAID BY ADDRESSEE

FUJITSU GTE BUSINESS SYSTEMS
PUBLICATIONS DEPARTMENT
2411 WEST 14TH STREET
TEMPE AZ 85281-9904

FOLD HERE CAREFULLY. DOCUMENT WILL BE SCANNED BY OCR EQUIPMENT

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

THIS PAGE INTENTIONALLY LEFT BLANK.

1.0 GENERAL

This section provides the station feature operating procedures for the Fujitsu GTE Business Systems' OMNI SI Digital Private Automatic Branch Exchange (PABX) (System Version Release (SVR) 5.2.1 .O).

THIS PAGE INTENTIONALLY LEFT BLANK.

2.0 STATION EQUIPMENT

The operations described in this part involve the use of standard telephone station instruments, equipped with a ringer (or equivalent) and an appropriate signaling device (dial pulse or dual tone multi-frequency). Agent instruments and Attendant Consoles are equipped with an electronic telephone which incorporates a keypad and an audible signaling device; therefore, the described operations apply to these units as well. However, control sequences will be somewhat different and certain system limitations will be set. Separate subparts describe the operations of each of these units.

Telephones without dials and/or ringers can also be used to perform certain functions. For instance, an instrument without a dial/keypad could be used for terminating calls (answer only). Another instrument without a ringer could be assigned hot-line (automatic ringdown) duty where an off-hook would initiate a call to a specified termination. In either case, a fully equipped telephone could also be used.

All operating procedures described assume that the appropriate class of service is provided to allow the function. In a step where the operation results in a feature access acknowledgment, only the mention of the feature confirmation tone (busy tone at 60 IPM or slow busy) is made, and the feature denial tone (busy tone at 120 IPM or fast busy) can occur indicating the feature access would have to be reinitiated. For simplicity, only the terms “Hookswitch Flash” or “Flash” are used to describe the typical station operation (depressing the hookswitch, holding it down for approximately one second, and then releasing).

2.1 Tones

Tones are used in normal operation and in various features. The system is capable of supplying the following tones:

(a) Dial tone, 350/440 Hz:

1. Dial tone uninterrupted
2. Tick tone, short burst of tone every 2 seconds
3. Break-in tone, burst of dial tone for a period of 1 second

(b) Busy tone, 480/620 Hz:

1. Line-busy tone interrupted at 60 IPM
2. Trunk-busy tone interrupted at 120 IPM
3. Camp-on tone, burst of trunk busy tone for 1 second
4. Feature-confirmation tone, same as line slow busy tone at 60 IPM
5. Feature-denial tone, same as trunk-busy tone

(c) Ringback tone, 440/480 Hz:

1. Ringback tone, interrupted at 1 second on, 3 seconds off

2. Distinctive dial tone, continuous

(d) Feature confirmation tone, 350/440 Hz

2.2 Station Equipment Procedures

2.2.1 Listed Directory Number Calls (Manual Incoming Calls)

All incoming calls to the Listed Directory Number (LDN) from dial-to-manual or manual-to-manual tie trunks are routed to an attendant. The system recognizes when an LDN trunk Central Office (CO) or when a ringdown tie trunk is seized, and switches the call to an idle loop of an idle attendant (Attendant Console).

2.2.2 Station-to-Station Calling (Internal Call)

The station user, operating a standard two-wire Dial Pulse (DP) or Dual Tone Multi-Frequency (DTMF) telephone, can place a call to another station within the same system by dialing a three- or four-digit number. If Direct Inward Dialing (DID) is used, the internal station number consists of the last three or four digits of the seven-digit number. To place an internal call, the following steps are to be performed.

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard
2	Dial desired number.	Ringback or busy tone heard.

2.23 Attendant Call or Recall

To call the attendant or operator from a station, perform the following:

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial attendant access code.	Ringback or busy tone heard.

To recall the attendant or operator from a station with an outside party or station on the line, perform the following:

STEP	OPERATION	RESULT
1	Flash hookswitch.	Distinctive dial tone heard.
2	Dial attendant access code.	Ringback tone heard. When attendant answers, a two- or three-way call is established (depending on data base programming).

NOTE: If busy tone is heard, reflash hookswitch to re-establish the original connection.

2.2.4 Direct Outward Dialing (DOD)

A station user can gain access to the Central Office network or to tie trunks without the attendant assistance. The user has direct access by dialing a one-, two-, or three-digit access code, depending on the numbering plan used and subject to system restrictions. To place a DOD call, perform the following:

STEP	OPERATION	RESULT
1	Go off-hook.	Internal dial tone heard.
2	Dial outside access code.	One of two cases will occur: Case 1: An FRL is insufficiently high to permit completing the call. Line busy signal heard. Ask attendant to dial the number or dial directly via MERS. Case 2: An FRL is high enough to permit completing the call. External dial tone heard.
3	Dial desired number.	Ringback or busy tone heard.

NOTE: Tick tone is an additional indicator showing that the system will accept the digits (data base option).

2.2.5 Trunk Queuing

Trunk Queuing allows a station user to enter into an on-hook or off-hook queue when the outside trunk line is busy. Note that Trunk Queuing and MERS Trunk Queuing are mutually exclusive.

(a) On-Hook Queuing

STEP	OPERATION	RESULT
	Begin from an attempt to make an outside call and all trunks are busy	
1	Flash hookswitch.	Distinctive dial tone heard.
2	Dial on-hook call queuing access code.	Queuing indicated by receiving 60-IPM confirmation tone.
3	Go on-hook (see note 1).	
4	Station rings (see note 2), indicating that trunk is idle and the system is calling the station.	
5	Go off-hook.	Central Office dial tone heard.
6	Dial number of desired destination (see note 3).	

NOTES:

1. Stations can originate or receive other calls while on queue.
2. When a trunk becomes available, the station rings for a period of time (defined in data base). If not answered, the next station(s) in the queue is serviced until a second time period has elapsed (data base specified). When a trunk becomes available, the station rings for the specified time, and if not answered before the end of that period, station is dropped from trunk queue.
3. Do not redial the outside access code. For a MERS call, no other numbers need to be dialed.

(b) Off-Hook Queuing

STEP	OPERATION	RESULT
	Begin from an attempt to make an outside call and all trunks are busy.	
1	Stay off-hook.	(a) 60-IPM confirmation tone heard.

STEP	OPERATION	RESULT
		(b) The station is placed on hold. (c) Central Office dial tone is heard when trunk is available.
2	After receiving Central Office dial tone dial the desired destination (see note).	

NOTE: Do not redial the outside access code.

2.2.6 Most Economical Route Selection (MERS) Access

NOTE: Trunk Queuing and MERS Trunk Queuing are mutally exclusive.

(a) MERS Access - Off-Hook Queue

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone received.
2	Dial MERS access code.	Quiet connection heard.
3	Dial 7- or lo-digit number. or IDDD number.	One of the following two cases will occur: Case 1: An FRL is insufficiently high to permit completing the call. Distinctive dial tone heard. Enter your Authorization Code. Quiet connection heard. Case 2:An FRL is high enough to permit completing the call. Quiet connection heard. After either case: (a) If MERS route not busy, ringback tone heard. (b) If MERS route busy, busy tone heard, go to Step 4.

NOTES:

1. The “1 " digit preceding a DOD call is not required and for long distance calls with the local NPA (area) code number, these three digits are also not required.

STEP	OPERATION	RESULT
	2. IDDD number format is 011 + country code + national number.	
4	Remain off-hook after busy tone heard.	(a) Busy tone ceases, confirmation tone heard. (b) Station is automatically entered into ueue. (c) Call is placed when MERS route becomes available. (d) Ringback tone heard.
5	Continue as with a normal line-to-line call.	

(b) MERS Access - Trunk Call Queue

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial MERS access code.	Quiet connection heard.
3	Dial seven- or ten-digit number.	(a) If MERS route not busy, ringback tone heard. (b) If MERS route busy, busy tone heard, go to Step 4.
NOTE:	The "1" digit preceding a DDD call is not required and for long distance calls with the local NPA (area) code number, these three digits are also not required.	
4	Flash hookswitch.	Distinctive dial tone heard.
5	Dial trunk call queue access code.	(a) Confirmation tone heard. (b) Station automatically entered in MERS queue.
6	Go on-hook.	When MERS route becomes available, system rings phone.
7	Go off-hook.	(a) System automatically dials previously entered numbers. (b) Ringback tone heard.

2.2.7 Facility Restriction Level (FRL)/Authorization Code/Traveling Class Mark (TCM)

STEP	OPERATION	RESULT
1	A station attempts to access an outside facility, (usually trunk for external call), however, station does NOT possess high enough FRL (Facility Restriction Level) to access facility.	Distinctive dial tone heard when accessing trunk via MERS. Otherwise, user will hear busy signal. The user must either ask attendant to dial the number or user can dial the number via MERS.
2	After hearing distinctive dial tone, the MERS dialer can either: (a) Dial the Authorization Code (string of digits up to 7 in length). Each Authorization Code is engineered or assigned an FRL (see note). If the FRL for the Authorization Code is equal to or higher than the facility's FRL, access is granted.	Quiet connection heard.
	or	
	(b) Wait for the Trunk Call Queuing feature (On- or Off-Hook Queuing - see section 2.2.5 for operations and details). After waiting on the trunk call queues, the FRL may eventually be "bumped" to a high enough level to pass the FRL restriction level (see note).	Quiet connection heard.
3	The FRL that eventually passes the FRL restriction level is carried on to the adjacent PABX as a TCM (Traveling Class Mark). This TCM is used like a station FRL for access privileges to its facilities.	

NOTE: FRL "Bumping" and Authorization Codes are system-engineerable and are an option available to your PABX system/network.

2.2.8 Call Transfer and Station Consultation

The station user, while connected to another station or trunk, can transfer the calling party to another station or trunk, or have a consultation-hold-type call and automatically return to the calling party when the station of the consulted party is placed on-hook. The station user can also establish a 3-way conference connection during a consultation-hold-type call by momentarily flashing the hookswitch.

(a) Call Transfer

STEP	OPERATION	RESULT
	Begin with an outside or internal call on the line.	
1	Flash hookswitch.	(a) Distinctive dial tone heard. (b) Calling party placed on hold.
2	Dial desired station number.	(a) Ringback tone heard. (b) Called station rings.
3	When the called station answers, announce call and go on-hook.	Holding party and called station connected.

NOTES:

1. In case of misdial, busy or no answer, flash hookswitch to return to outside party.
2. The forwarding station may also go on-hook before called station answers.

(b) Call Transfer to IVMS

Any station user may transfer an internal or external call to IVMS.

STEP	OPERATION	RESULT
1	When an outside party or another station is on the line, flash hookswitch.	(a) Distinctive dial tone is heard. (b) Calling party placed on hold.
2	Dial access code.	(a) Ringback or busy tone heard. (b) IVMS is being called.
NOTE:	In case of misdial, busy, or no answer, flash hookswitch to return to the first party. When IVMS responds, the first party and IVMS are connected.	
3	Place station on-hook.	

(c) Station Consultation

STEP	OPERATION	RESULT
	Begin with an outside or internal call on the line.	
1	Flash hookswitch.	(a) Distinctive dial tone heard. (b) Calling party placed on hold.
2	Dial the desired station. If consulting an outside party, dial outside access code, wait for second dial tone, then dial desired directory number.	(a) Ringback tone heard. (b) Called station rings.
3	When called station answers, announce conference.	Transmission established between two parties.
NOTE:	In case of misdial, busy, or no answer on a station call, flash hookswitch to re-establish the original connection; on an outside call, flash hookswitch twice.	
4	To return to the original party, flash hookswitch or wait until the consulted party hangs up to be automatically connected to the original party. If the call is to an outside number, flash hookswitch twice.	Original connection re-established.

(d) Add-On Conference (Three-Way)

Add-On Conference allows a station user to establish a three-way conference call while connected to a station or trunk. The third party can be either a station or a trunk.

STEP	OPERATION	RESULT
	Begin with an outside or internal call on the line.	
1	Flash hookswitch.	(a) Distinctive dial tone heard. (b) Original party placed on hold.
2	Dial desired station. If an outside party, dial outside code, wait for a second tone, and then dial desired directory number.	(a) Ringback tone heard. (b) Called station rings.
3	When called station (third party) answers, announce conference.	Transmission established between two parties.

STEP	OPERATION	RESULT

MOTE: If busy tone or no answer is received, flash the hookswitch to re-establish original connection.

4	Flash hookswitch.	Three-way connection established.

NOTE: Another hookswitch flash drops the third party from the connection.

2.2.9 Call Hold

Hold allows a station user to put another station or trunk on hold. The called station does not have to remain off-hook while the calling party is on hold. The holding party can establish and receive new calls, and continue to use the system features as required. To return to the held party, dial the call hold answer access code. A call placed on hold can only be picked up by the station placing the call on hold.

STEP	OPERATION	RESULT
1	Inform caller of intent to place on hold and flash hookswitch.	(a) Distinctive dial tone heard. (b) Calling party is placed on hold.
2	Dial call-hold access code.	Confirmation tone heard.
3	If no system function is desired, place station on-hook. (Station is now free to originate or receive calls.)	Call-Hold party status does not change.
4	Return to party on hold by going off-hook and dialing access code.	Calling party reconnected to station.

NOTES:

1. The party on hold can be placed in a three-way conversation by following the three-way calling procedure. However, the call-on-hold access code is dialed rather than the party's station number.

2. The calling party placed on hold is automatically returned to the called station within a predetermined time. If the recalled station does not answer, the call is diverted to the attendant.

2.2.10 Call Park

Call Park allows a station user to place a call (station or trunk) in a call-park queue. Once in the queue, the call can be answered by dialing the call park answer code. A call not answered within a certain period (data base specified) is diverted back to the attendant. A call in call park can automatically ring the station dialed first, then direct to the attendant, or just divert to the attendant (data base option). When conversing with another party, and the other party wishes to speak to another party at another station, perform the following:

STEP	OPERATION	RESULT
1	Flash hookswitch.	Distinctive dial tone heard.
2	Dial call-park access code followed by station number desired.	(a) Confirmation tone heard. (b) Calling party parked on station.
NOTE:	The station that the call is parked on can still originate or receive other calls.	
3	Phone or page, and inform called party of party in call park. If called party cannot be reached, notify calling party.	Party called answers by dialing the call-park access code and the station number

NOTES:

1. Calls may be parked to any station number by dialing the call-park access code and the desired number. Calls may also be parked to an unequipped station provided the line is parked maintenance busy.
2. Calls will automatically ring the parked-on station with a predetermined time (up to four minutes) and if not answered, diverts to the attendant or station.
3. The call-park procedure does not affect operation of the station on which a call is parked or from which a call is parked.
4. A parked call may be retrieved by any user from any phone. The retrieving user does not have to be the party that parked the call nor the station to which the call was parked.
5. The park feature is independent of line appearance arrangements (i.e., the parked-to line need not appear on the retrieving user's set).
6. Multiple calls may be parked on a single number and retrieved on a First-In First-Out (FIFO) basis by repeatedly dialing the access code.

2.2.11 Call Waiting, Originating

Call Waiting, Originating allows a station user to camp on to a station that has been called and from which a busy indication is received. Refer to the call waiting, terminating procedure for answer details and camp-on restrictions.

STEP	OPERATION	RESULT
	Begin with a call to a busy station.	
1	Flash hookswitch.	Distinctive dial tone heard.
2	Dial call waiting, originating access code.	(a) Camp-on indicated by ringback tone. (b) Camped-on station hears burst of tone.
3	Remain off-hook until camped-on station answers.	

2.2.12 Call Waiting, Terminating

Call Waiting, Terminating allows a station user to have a calling party receiving a busy signal to be camped on automatically. The calling party receives a ringback tone and the called party receives a camp-on tone. The called party is automatically rung by the calling party when the station goes on-hook. Alternatively, the station user can depress the hookswitch and dial the call-waiting answer access code. This action reverses the status of the waiting call by placing the current call on hold while connecting the called party to the waiting call. The call on hold is re-established by either of the preceding methods. This method of operation allows the answered call to be transferred (if required) before returning to the waiting caller.

NOTES:

1. Call Waiting may only be established on line-to-line or line-to-trunk stations.. It may not occur when a station is ringing, dialing, consulting, or in transferring modes, or when a call having data line security is marked on either party's station. Call Waiting, Terminating is not re-established if the calling party is part of a three-way consultation-transfer call.

2. The maximum number of calls allowed waiting or camped on is determined by the system data base.

3. Two classmarks are provided for Call Waiting, Terminating, and a station may have both. One classmark allows for DID calls only. The other classmark allows for non-DID calls.

STEP	OPERATION	RESULT
1	Terminate conversation on receiving a camp-on tone and place station on-hook.	Station will ring.
2	Inform caller of a call waiting and intent to place on hold upon receiving a camp-on line and flash hookswitch.	Calling party placed on hold.
3	Transfer the camp-on caller to another station if desired.	
4	Return to caller on hold by dialing call-waiting answer code.	Calling party is reconnected.

NOTE: It is possible to transfer the camped-on party to some other station before returning to the original party.

2.2.13 Call Waiting, Toggling

Call Waiting, Toggling is a variety of terminating call waiting which operates like the call waiting feature available from central offices in the public telephone network today. It allows a user to have a calling party camped on automatically. The calling party receives a ringback tone and the called party receives a camp-on tone. The called party is automatically rung by the calling party when the station goes on-hook. Alternatively, the station user can depress the hookswitch. This action reverses the status of the waiting call by placing the current call on hold while connecting the called party to the waiting call. The call on hold is re-established by either of the preceding methods.

NOTES:

1. Call Waiting may only be established on line-to-line or line-to-trunk stations. It may not occur when a station is ringing, dialing, consulting, or in transferring modes, or when a call having data line security is marked on either party's station. Call Waiting, Terminating is not re-established if the calling party is part of a three-way consultation-transfer call.

2. The maximum number of calls allowed waiting or camped on is determined by the system data base.

3. Two classmarks are provided for Call Waiting, Terminating, and a station may have both. One classmark allows for DID calls only. The other classmark allows for non-DID calls.

4. If a system is engineered for Call Waiting, Toggling, station users having one or more waiting calls will not be able to consult, transfer or establish three-way connections with other parties. Only toggling between the present call and the waiting call is allowed.

5. Call Waiting, Toggling via flash is not to be available to integrated featurephone users.

6. Unlike the Call Waiting feature available in the public telephone network, Call Waiting, Toggling cannot be disabled by dialing an access code on the calling station.

STEP	OPERATION	RESULT
1	Terminate conversation after receiving a camp-on tone and place station on-hook or go to Step 2.	Station will ring.
2	Inform caller of a call waiting intent to place current caller on hold upon receiving a camp-on tone.	Calling party placed on hold.
3	Flash hookswitch.	Waiting call answered.
4	Return to caller on hold by flashing hookswitch.	(a) Original party is reconnected. (b) Current caller placed on hold.

2.2.14 Executive Override

Executive Override allows a properly classmarked user to break into a two-party conversation. After receiving a busy tone, the calling party depresses the hookswitch, receives distinctive dial tone by dialing an access code, and the override breaks into the conversation. Break-in tone is applied to the connection before break-in.

If the called station is busy, perform the following:

STEP	OPERATION	RESULT
1	Flash hookswitch.	Distinctive dial tone heard.
2	Dial executive-override access code.	(a) Conversing parties receive break-in tone. (b) Three-way connection established.

NOTES:

1. Break-in can only be established on station-to-station or station-to-trunk calls. It cannot occur when a station is in ringing, dialing, consulting, or transferring mode, or when a call having data line security is marked on either party's station.

2. Executive Override is not enabled to a station that has terminating call waiting.

2.2.15 Station Camp-On, Callback, and Cancel

Station Camp-On allows the user with a properly classmarked station either to camp on to one station or to have multiple stations camped on at any separate time without any indication to that station.

If the camp-on function cannot be recalled due to classmark restriction or if the camp-on list is exhausted, a reorder tone is heard.

NOTES:

1. The station camp-on; attendant camp-on; call waiting, originating; and call waiting, terminating features use the same camp-on mechanism. The activation of any of these features constitutes a camp-on condition for the stations involved.

2. A maximum of 20 station camp-ons with callback can be in effect at one time.

(a) Station Camp-On with Callback

STEP	OPERATION	RESULT
	Begin with a call to a busy station.	
1	Flash hookswitch.	Distinctive dial tone heard.
2	Dial camp-on with callback access code.	Camp-on indicated by slow busy tone (60 IPM).
3	Go on-hook.	When camped-on station is placed on-hook, station that is camped on will ring.
4	Terminate call by not answering within six rings.	
5	Go off-hook.	(a) Ringback tone heard. (b) Four seconds after station is taken off-hook, camped-on station will ring.

STEP	OPERATION	RESULT
6	Cancel call in four seconds by placing station on-hook.	

(b) Cancel Station Camp-On with Callback

STEP	OPERATION	RESULT
1	Dial cancel camp-on with callback code.	Cancellation of camp-on indicated by slow busy tone (60 IPM).
2	Go on-hook.	

22.16 Dial Call Pickup

Dial Call Pickup allows a station user in **a** dial call pickup group to be connected to a station in the ringing mode within the same pickup group by dialing a special access code. Each pickup group has stations associated with it programmed in the data base. The maximum size of a call pickup group is all lines in the system. The maximum number of pickup groups is 127 on a system basis.

STEP	OPERATION	RESULT
	Begin with an unattended station with the group ringing.	
1	Go off-hook.	Dial tone heard.
2	Dial call pickup access code.	Caller connected to ringing station.

2.2.17 Directed Dial Call Pickup

Directed Dial Call Pickup allows a station user to transfer a ringing call to another station by dialing a special access code plus the station number of the ringing line. Any properly classmarked station can use this feature.

STEP	OPERATION	RESULT
	Begin with an unattended station in another group ringing.	

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial directed call pickup access code.	
3	Dial number of ringing station.	The call is diverted to station dialing the access code, and call can be answered.

2.2.18 Extended Dial Call Pickup

Extended Dial Call Pickup allows selected stations to pick up calls of more than one group by being associated with one of 30 extended pickup tables. Any number of stations can be associated with an extended pickup table. Each extended pickup table identifies up to five, including the station user's primary group, out of the 127 pickup groups that can be picked up by these stations. When the selected station dials the extended dial pickup access code, the station picks up stations in the first group identified in the extended pickup table.

If the selected station dials the extended dial call pickup access code, it can pick up stations in the primary group or any one of the other four groups identified in the extended pickup table.

NOTES:

1. Calls can only be picked up by idle stations, or stations that have placed their current calling party on hold or park before requesting the call pickup.
2. The attendant cannot use this procedure from the Attendant Console.
3. A station being recalled via the station camp-on and callback procedure cannot be picked up via the dial call pickup or directed call pickup procedure.

STEP	OPERATION	RESULT
	Begin with an unattended station in extended group ringing.	
1	Go off-hook.	Dial tone heard.
2	Dial extended call pickup access code, and answer call code.	Call diverted to station dialing call pickup access code.

2.2.19 IVMS Direct Access

STEP	OPERATION	RESULT
1	Take station off-hook.	Dial tone heard.
2	Dial IVMS directory number.	A recorded message or busy tone is heard.
3	Follow spoken, English language commands given by IVMS to manipulate your personal mailbox.	As specified by the IVMS User's Guide.

2.2.20 Call Forwarding

Call Forwarding allows incoming calls to a station to be automatically forwarded to an alternate station, attendant, a recorded announcement, or hunt group pilot number. Calls can only be forwarded to destinations internal to the system. The conditions under which a call is forwarded and the destination of the forwarded call can be fixed or variable.

Restrictions to the call-forwarding procedures are as follows:

(a) This procedure does not modify any of the originating features of the station.

(b) Call forwarding is subject to the terminating procedures of the destination station.

(c) Call forwarding, under the does-not-answer condition encountering a busy condition at the destination, reverts to the called station and continues ringing.

(d) Call forwarding, under the busy condition encountering a busy condition at the destination, reverts to the original station and receives a busy tone. The calling party may then use other operations on the called station (e.g., Busy Override).

(e) Call forwarding, under the all-calls condition encountering a busy condition at the destination, gives a busy tone to the caller.

(f) The call-forwarding and station-hunting operations share a common data base allowing the member of a hunt group to use the call-forwarding fixed feature to be removed from the hunting sequence. Use of call forwarding variable within a station-hunting group should be applied only to the final station in a terminal hunt group. If this rule is not followed, the hunt group is disrupted.

(g) Call forwarding cannot be used to change both the forward condition and the destination if the destination is a recorded announcement.

Call Forwarding features are as follows:

(a) Variable Call Forwarding

Variable Call Forwarding allows the user to forward incoming calls automatically to the attendant, to another station, or to the pilot number of a hunt group.

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial variable call-forwarding access code.	
3	Dial appropriate new condition code. New condition codes are as follows: (a) For forwarding all calls if the station is busy, dial 1. (b) For forwarding all calls if the station does not answer, dial 2. (c) For forwarding all calls if the station is busy or does not answer, dial 3. (d) For forwarding all calls, dial 4.	
4	Dial the directory number of the station to which the calls are to be diverted. Directory numbers may represent a station, attendant, recorded announcement, IVMS, individual speed call list entry, or tone.	If call forwarding is successful, the station receives a slow busy tone (60 IPM). If call forwarding is unsuccessful, the station receives a fast busy tone (120 IPM).
5	Go on-hook.	

(b) Cancel Variable Call Forwarding

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial variable call-forwarding access code.	

STEP	OPERATION	RESULT
3	Dial new condition code 0 (forward no calls).	(a) Canceled if slow busy heard. (b) Not canceled if fast busy heard.
4	Go on-hook.	

(c) Fixed Call Forwarding

Fixed Call Forwarding allows station users to change the conditions under which calls are to be forwarded, but not the destination.

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial fixed call-forwarding access code.	
3	Dial appropriate new condition code. New condition codes are as follows: (a) For forwarding all calls if the station is busy, dial 1. (b) For forwarding all calls if the station does not answer, dial 2. (c) For forwarding all calls if the station is busy or does not answer, dial 3. (d) For forwarding all calls, dial 4.	If call forwarding successful, station receives slow busy tone (60 IPM). If, unsuccessful, the station receives a fast busy tone (120 IPM).
4	Go on-hook.	Calls will be forwarded to predetermined destination.

(d) Cancel Fixed Call Forwarding

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial fixed call-forwarding access code.	

STEP	OPERATION	RESULT
3	Dial new condition code 0 (forward no calls).	
4	Go on-hook.	(a) Canceled if slow busy heard. (b) Not canceled if fast busy heard.

2.2.21 Conferences

(a) Attendant/Station Conference

The station user, with attendant assistance, may establish up to seven conference connections (a total of eight including the attendant) on each of the two conference bridges.

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial attendant access code.	Ringback tone heard.
3	When attendant answers, provide conference information.	

(b) Meet-Me Conference

Meet-Me Conference enables the station users to set up a conference connection by dialing a special meet-me-conference access code, plus the conference bridge number. The maximum capacity of the conference is seven stations or trunks, plus one attendant. Trunks must be inserted by the attendant, or incoming tie trunks can have direct access if appropriately classmarked. The system has a maximum of two conference bridges, the same two bridges used for attendant and progressive conferences.

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial meet-me conference access code number plus digit (0 or 1) to select appropriate conference bridge number.	Break-in tone heard by calling party and by all conferences. Calling party added to conference.

STEP	OPERATION	RESULT
3	Other stations performing Steps 1 and 2 will meet in the conference bridge.	

(c) Station Controlled Progressive Conference

Station Controlled Progressive Conference allows the addition of non-preselected stations, as well as trunks, to a station-initiated conference. The system has two conference bridges which are used for the attendant and meet-me conference. The maximum capacity of a conference bridge is seven, stations or trunks, plus one attendant. The conference access code for progressive conference is different from the meet-me conference code. The conference originator or attendant dialing the access code is the only station that can add parties to the conference. Trunks without disconnect supervision remain bridged until all other connections have been released. When the controlling station disconnects, only the progressive feature is disabled. The conference is still in effect.

STEP	OPERATION	RESULT
1	Any equipped station dials access code assigned to progressive conference feature and conference bridge (refer to data base program).	
2	Flash hookswitch.	Distinctive dial tone heard.
3	Dial number of station to be admitted to conference.	(a) Called station receives ringing tone. (b) Calling station receives ringback tone.
4	Take called station off-hook.	(a) Ringback tone stops. (b) Ringing stops. (c) Transmission established between the stations.
5	Flash hookswitch of the calling station.	Called station is added to conference circuit.
6	Repeat Steps 2 through 5 to add stations and trunks to conference circuit.	Results are same as in Steps 2 through 5.
7	Perform Steps 1 through 5 for conference bridge 2.	Results are the same as in Steps 1 through 5.

NOTE: When the station has a conference circuit in the progressive mode, the conference circuit is made busy to any other station. The station that initiates the conference can call the attendant without disconnecting from the conference circuit.

2.2.22 Speed Calling and Update

(a) Individual Speed Calling

Individual Speed Calling allows the station user to have a unique set of up to eight speed-calling entries programmed into an individual speed-calling list in the system data base. This individual list allows the station user to place local and/or toll calls to frequently called numbers by dialing the individual speed-calling list access code plus a speed-calling list entry number (1-8). Each entry may contain up to 15 digits. The ability to bypass toll restrictions is determined by the toll restriction status for the station. A particular station can have access to both an individual speed-calling list and the group speed-calling list.

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial speed-calling access code.	
3	Dial speed-calling entry number (l-8) of the desired outside party.	Ringback or busy tone heard.
4	Continue as with a normal line-to-line call.	

(b) Individual Speed-Calling Update

Individual Speed-Calling Update allows a station user who has access to an individual speed-calling list to update the list. The station user can add, delete, or change any of the eight telephone numbers stored in the system data base for an individual list.

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial access code for updating individual speed-calling data.	Dial tone stops.
3	Dial speed-calling entry number (l-8) to be modified.	Dial tone heard again indicating that system is prepared to accept speed-calling data.

STEP	OPERATION	RESULT
4	If the list entry is to be deleted, dial a # and procedure is complete; otherwise, go to Step 5.	
5	Dial speed-calling information which can consist of 1 to 15 digits.	Dial tone stops.
NOTE:	If a pause or a special character entry is required, dial an *, then dial a 1 for a short pause or a 2 for a long pause (durations determined in the data base) or the * or # character, then continue dialing digits 0 through 9.	
6	Dial # to indicate procedure is complete.	Confirmation tone heard.
7	Go on-hook.	

(c) Group Speed Calling

Group Speed Calling allows station users and attendants to place outgoing local and/or toll calls to frequently called numbers by dialing an access code plus a speed-calling list entry number. The speed-calling list is defined in the system data base and has 1000 speed-calling entries. Each entry can contain up to 15 digits. Within a speed-calling list, each entry can be optionally marked to bypass all toll and trunk group restriction checks. Each station can be assigned to one of the 48 group speed-calling classes. Each class allows or denies access to 250 subgroups (in any combination) and each subgroup identifies four entries of automatic speed-calling sequences. Some stations could have access to all 1000 entries; some stations could access only the first eight and the last four entries, etc.

NOTE: A station which does not have MERS access cannot make use of group speed call list entries which contain the MERS access code.

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial speed-calling access code.	
3	Dial speed-calling entry number of desired outside party (000-999).	Ringback or busy tone heard.
4	Continue as with a normal line-to-line call.	

(d) Group Speed-Calling Update

This update allows a station user to add, delete, or change numbers that are stored in the group speed-calling list.

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial appropriate code for updating the group speed-calling data.	Dial tone stops.
3	Dial group speed-calling list entry number to be modified (000-999).	Dial tone heard again, indicating that system is prepared to accept group speed-calling data.
4	If group speed-calling list entry is to be deleted, dial # and procedure is complete; otherwise, go to Step 5.	
5	Dial a digit for the bypass trunk group access restriction (0 = restrict, 1 = bypass).	Dial tone stops.
6	Dial a value for the bypass toll restriction (0 = restrict, 1 = bypass).	
7	Dial group speed-calling information which can consist of l-l 5 digits.	
NOTE:	If a pause or a special character entry is required, dial an *, then dial a 1 for a short pause or a 2 for a long pause (durations determined in the data base) or the * or # character, then continue dialing digits 0 through 9.	
8	Dial # to indicate procedure is complete.	Confirmation tone heard.
9	Go on-hook.	

2.2.23 Dictation Equipment Access

Dictation Equipment Access allows station and attendant users access and control of customer-owned dictation recording equipment.

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.

STEP	OPERATION	RESULT
2	Dial dictation-access code.	Ready tone heard.
3	Dial code to start the dictation machine, then dictate.	Dictation machine begins recording.
4	Dial code to stop dictation machine.	Dictation machine stops.
5	Dial code to rewind tape a short distance.	Dictation machine rewinds a short distance.
6	Dial code to play back recording.	Dictation machine rewinds and plays back recording.
7	Dial code to release dictation equipment.	Dictation machine released.
a	Go on-hook.	

2.2.24 Paging Equipment Access

Paging Equipment Access allows the station user access to customer-owned voice-paging equipment.

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial paging access code.	
3	Dial zone digit (data base option).	
4	Page desired party.	Voice is transmitted through paging equipment.
5	Go on-hook and wait for paged party to answer.	

2.2.25 Code Calling

Code Calling enables the station user to access the paging circuits and have the system perform the code-calling function (no additional code-calling equipment is required).

STEP	OPERATION	RESULT
1	Take station off-hook.	Dial tone heard.
2	Dial code-calling access code.	
3	Dial zone digit (data base option).	
4	Dial code number of desired party.	Tones are transmitted over paging equipment.
5	Keep station off-hook until called party answers using code call-answer access code.	

2.2.26 Universal Night Answer (UNA) and Predetermined Night Answer (PNA)

Calls to zones designated UNA are handled by the station user as follows (see note for PNA).

STEP	OPERATION	RESULT
1	An audible or visible signal is received in universal-night-answer zone.	
2	Go off-hook.	Dial tone heard.
3	Dial universal-night-answer access code.	
4	Dial appropriate universal-night-answer zone digit (l-4, or 0 for all zones).	Call diverted to station user.

NOTE: Any station designated as a PNA station can receive incoming calls or place outside calls.

2.2.27 Remote Station Access to System Features

Remote Access allows an outside station, not associated with the immediate system, access to system features.

STEP	OPERATION	RESULT
1	From an outside touch-calling telephone, dial number assigned to remote access feature.	One of the following occurs: (a) If an authorization code is required, distinctive dial tone is returned to the station. (b) If an authorization code is not required, regular dial tone is returned to station, indicating access is allowed to system services.
2	If required, dial authorization code.	Regular dial tone returned to station, indicating access allowed to system services.
3	Dial access code for desired service.	System processes call in normal manner.
4	Go on-hook.	

22.28 Timed Reminder

Timed Reminder allows a station user to dial an access code and a five-digit time code so that the station will be automatically called at the specified time.

(a) Initialization of Timed Reminder

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial timed-reminder access code.	
3	Dial five-digit time code (see note).	One of the following occurs: (a) If successful, confirmation tone heard. (b) If unsuccessful, reorder tone is heard.
4	Go on-hook.	
NOTE:	The first digit of the time code is either 2 (or A for AM) or 7 (or P for PM), and the remaining four digits are the hour and minute.	

(b) Cancellation of Timed Reminder

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial timed-reminder cancel code.	(a) If successful, confirmation tone heard. (b) If unsuccessful, reorder tone heard.
3	Go on-hook.	

NOTE: From'an administrative telephone position, the operator will have to dial the station number after the access code or the cancel access code. An administrative telephone position occurs when a station, such as a dial call pick-up unit, functions in an attendant mode.

2.2.29 Message Waiting (Administrative Telephone Position)

Message Waiting allows the user to cause a telephone-associated lamp to flash as an indication to a station that a message is waiting.

(a) Initialization of Message Waiting

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial message-waiting access code.	
3	Dial station number.	(a) If successful, confirmation tone is heard. (b) If unsuccessful, reorder tone is heard. (c) The associated telephone lamp flashes.
4	Go on-hook.	

(b) Cancellation of Message Waiting

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Dial message-waiting cancel code.	
3	Dial station number.	(a) If successful, confirmation tone heard. (b) If unsuccessful, reorder tone heard.

STEP	OPERATION	RESULT
4	Go on-hook.	

2.2.30 Secretarial Services (Screening)

Screening allows a station functioning in a secretarial mode or display set to override a station in the forward or divert all calls condition and return the call to that station (if the originally called station is not off-hook). For different call types, the display telephone set exhibits the following:

(a) Originally called number for station diverted to station calls

(b) Calling number for station to station calls

(c) Calling number for station to hunt group number calls

(d) Blank display for trunk to station calls

STEP	OPERATION	RESULT
1	Answer forwarded call.	Transmission established.
2	Flash hookswitch.	Distinctive dial tone heard.
3	Dial station number of forwarded call.	Ringback heard if station is idle.
4	Go on-hook.	Call routes back to originally called station.

NOTE: Any station or Attendant Console not operating in a night answer mode can be assigned for secretarial services.

2.2.31 Direct Trunk Access (Maintenance Station)

By dialing a one-, two-, or three-digit access code plus a three-digit trunk number from a station designated a maintenance class of service in the data base, the user can access a specific trunk.

2.2.32 Specialized Common Carrier (SCC)

SCCs may require more than 30 digits to be entered in order to establish a long distance connection. A user must first access the correct trunk group (l-3 digits), dial the SCC directory number (7-10 digits), dial the SCC authorization code (l-10 digits), and finally dial the desired DDD number (7-11 digits).

The SCC access feature performs segments of this sequence with a minimum number of digit entries. Data base programming provides five SCC access sequences. Any one can provide a fully automatic connection to the SCC DDD network or can require the manual entry of an authorization code before connecting to the SCC DDD network. Speed calling can also be used with the SCC access sequence offering further system enhancement.

(a) SCC Access - Manual Authorization Code Entry

STEP	OPERATION	RESULT
1	Dial SCC access code.	(a) Restriction indicator test performed for: 1. Bypass trunk group access checks 2. Bypass toll access checks 3. Bypass both checks 4. Bypass no checks (b) Correct trunk group accessed. (c) SCC directory number automatically dialed. (d) SCC equipment dial tone heard.
2	Dial the required SCC authorization code in full.	SCC DDD network dial tone is heard.
NOTE:	If the SCC network does not return dial tone, pause for 2 seconds before proceeding.	
3	Dial the DDD number or dial an "*" and the appropriate speed-calling list access code and list number (individual or group speed).	Ringback is heard.

(b) SCC Access - Automatic Authorization Code Entry

STEP	OPERATION	RESULT
1	Dial the SCC access code.	(a) The restriction indicator test is performed for: 1. Bypass trunk group access checks

STEP	OPERATION	RESULT
		2. Bypass toll access checks 3. Bypass both checks 4. Bypass no checks (b) Correct trunk group is accessed. (c) SCC directory number is automatically dialed. (d) SCC equipment dial tone heard.
2	Dial an "*".	Required SCC authorization code automatically dialed.
3	Dial DDD number or dial an "*" and appropriate speed-calling list access code and list number (individual or group speed).	Ringback heard.

2.2.33 Change/Restore

This feature allows PABX station users to change the primary destination of a special directory number to a preselected alternate destination and also allows the primary destination to be restored. The possible destinations that can be used are as follows: line, attentdant, hunt group, agent group, recorder announcer and intercept code.

STEP	OPERATION	RESULT
1	Dial the change/restore access code.	Confirmation tone heard. The special directory number will terminate to the new destination.

NOTE: The change/restore access code is class restricted; therefore, any stations with the proper class have access to this feature.

3.0 INTEGRATED FEATUREPHONES (FeatureComm III/IV [Analog] and V/VI [Digital])

There are two types of Featurephones: the Analog and the Digital. The Featurephone shown in Figure 3.1, is a microprocessor-controlled desk telephone which offers enhanced capabilities through the use of an advanced digital data link. The Digital Featurephone operation is similar to the Analog Featurephone except that it utilizes a single card interface versus the line card and Control Interface to Periphery (CIP) card used with the Analog Featurephone.

3.1 General

Certain basic differences exist between the Featurephone and the standard telephone instrument. For instance, an electronic transducer is used to reproduce electronically generated tones to indicate “ringing”. Three distinctive ringer cadences are provided to indicate regular inside calls, outside calls, and inside intercom calls.

Action confirmation tones are also provided. These 1/1 0 of a second bursts of tone are referred to as “beeps” for simplicity. For instance, repetitious beeps (at about l-second intervals) indicate an error in programming and also act as the audible alerting signal for reminder and duration alarm features. A triple beep is used to indicate an invalid entry during programming. The double and single beeps are used for various alerting functions and will be indicated in the procedures. Visual signals will also be noted in the procedures. For example, the fluttering LED (480 IPM), along with the repetitious beeps, indicates an error in programming.

The Featurephone feature buttons perform more than one function. A depression of approximately 1/2 second (hereinafter referred to as “depress”) provides sufficient on-time to activate a preprogrammed feature, while simultaneously requiring enough on-time to prevent accidental activation. A depression of 2-3 seconds (hereinafter referred to as “depress and hold”) initiates entry into, or exit from, programming.

The Featurephone display provides prompts for programming and operating conditions in addition to information messages and system displays. Each specific message or number display will be enclosed in quotes ("") in the operating procedures that immediately follow a button depression or other action.

The Featurephone can seize the line by several methods other than taking the handset off-hook. Depressing the monitor, new call, or speakerphone (optional) feature button will also effect seizure. In the latter cases system tones and voice are heard via the internal loudspeaker. Depression of many other feature buttons initiates automatic seizure which activates the loudspeaker, acknowledges dial tone, and dials stored digits.

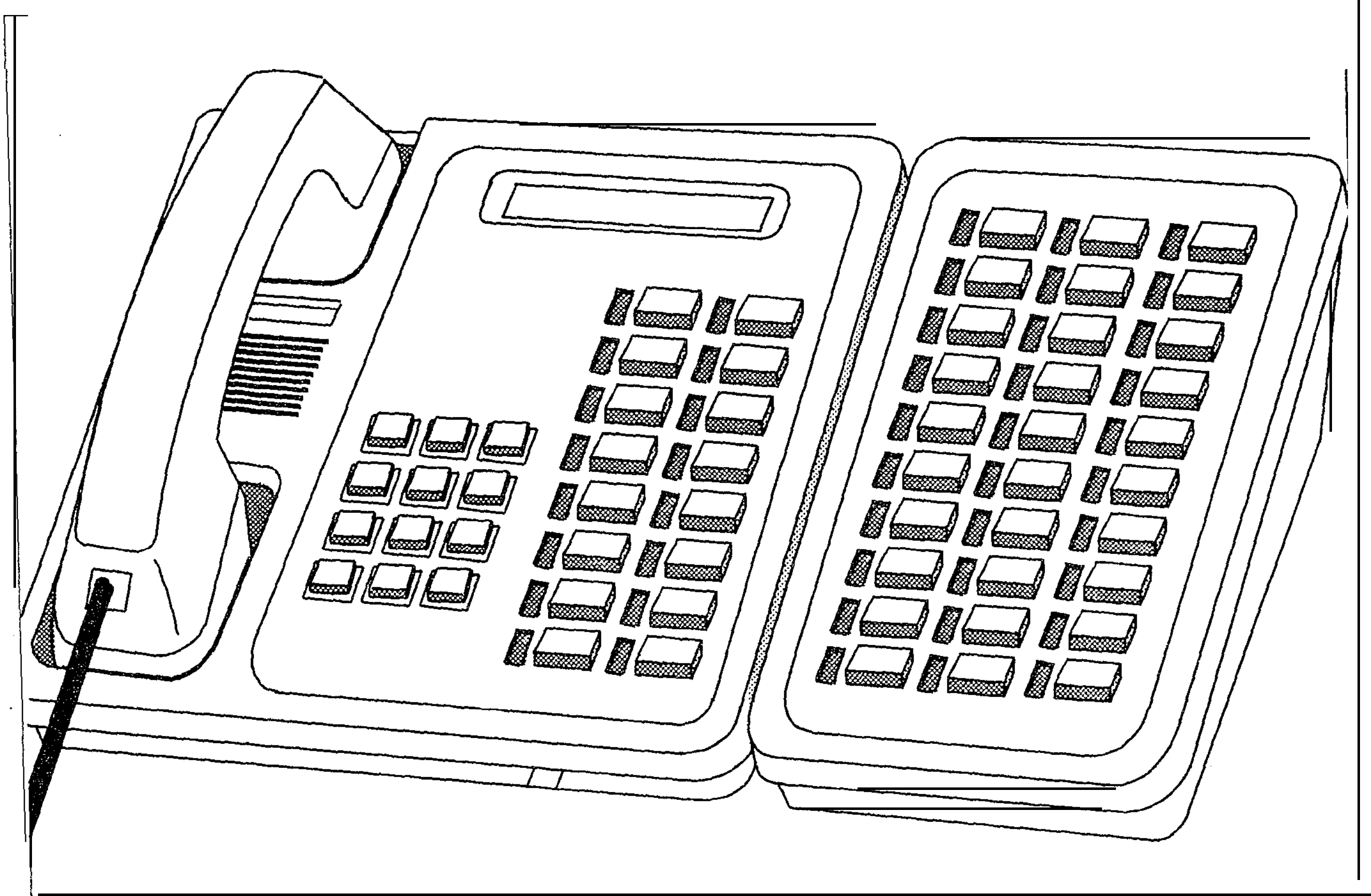


Figure 3.1 Featurephone With 30-Button Add-On Module

The Featurephone shares some identical operational traits with the standard telephone instrument. The standard "hookswitch flash" operation can be performed in the same manner as with a standard telephone, although one feature button can have the "flash" feature programmed into it. The touch keypad not only operates as in a standard telephone, but also provides digital signals to the system.

The information provided in this section covers programming and utilization of features unique to the Featurephone. Although the Featurephone can perform all the functions associated with standard DTMF station equipment, these will not be discussed here, except where enhancements are provided by the Featurephone.

3.2 Controls, Visual and Audible Indicators, Displays, and Messages

Two versions of the Featurephone are available, an 8-button unit and a 16-button unit. Options for both units include a speakerphone (16-button unit), and 30-button add-on module. The 8-button unit provides a 16-character LCD display located across the top-front of the instrument, while the 16-button unit has a 24-character display. This version has the increased unit width required for the second row of buttons. LEDs associated with each button are located near the left of the respective button and are used to indicate operation confirmation or visual signaling. All buttons are non-locking and a standard DTMF keypad is provided. A slider-type volume control located at the front edge of the case is provided for adjusting the sound level for monitor, voice page, and speakerphone. The volume and pitch of ringing and signaling tones are digitally programmed. Figure 3.1 shows a 16-button version of the IFP, with a 30-button add-on unit.

3.2.1 Dual-Function Buttons

Where a feature button has been programmed for the SHIFT function, depressing SHIFT changes the function of the next feature button depressed to its alternate function. Depressing the SHIFT button also causes the SHIFT LED to light and remain lit until the depression of another feature button or a second depression of the SHIFT button.

3.2.2 Visual Signals

The top LED adjacent to each feature button, when lit, indicates the status of the feature option as shown in Table 3.1. The bottom LED, when lit, indicates that the SHIFT key has been depressed and the feature button is in its alternate mode.

3.2.3 Audible Signals

Programming tones and their meanings are listed in Table 3.2, and distinctive ringing tones are listed in Table 3.3. Note that all tones normally associated with standard station equipment (Paragraph 2.1) can be accommodated by the Featurephone.

Table 3.1 Visual Signals

LED	STATUS
Steady LED	Feature button function is in operation
Winking LED (60 IPM)	Processing call while feature button is in use (1/2 sec. on, 1/2 sec. off)
Flashing LED (120 IPM)	Prompts for input (1/4 sec. on, 1/4 sec. off)
Fluttering LED (480 IPM)	Error in programming (1/16 sec. on, 1/16 sec. off)
Holding LED	Shows phone line on Hold (1/10 sec. on, 9/10 sec. off)
Exclusive Hold LED	Shows phone line on Exclusive Hold (1/10 sec. on. 1/10 sec. off, 1/10 sec. on, 7/10 sec. off)
Unit LED	Feature button is not in operation
Status LED*	Upper LED on optional Line buttons (Multi-line sets only)
I-Use LED*	Lower LED on optional Line buttons (Multi-line sets only)

*See paragraph 3.7.24, Line Buttons

Table 3.2 Audible Signals

PROGRAMMING TONES	MEANING
Single Beep	Announces programming mode to input (e.g., storing a number to Station Speed Calling).
Double Beep	Announces programming mode to change a button's function (e.g., from Release to Station Lock). Prompts for second access code when programming a Dual Access.
Triple Beep	Alerts you to an invalid entry during programming (such as entering 13 for the Month field in Time and Date). Also heard when switch denies action of phone.
Continuous Beep	Alerts you to an error in programming, announces a Reminder or Duration Alarm.
Ringing Tone	Announces incoming calls.
Buzzer	If equipped, a buzzer tone indicates calls transferred from your secretary.

Table 3.3 Distinctive Ringing

PROGRAMMING TONES	MEANING
Internal Calls	A repetitive ringing cycle of 1 second on, 3 seconds off.
External Calls	A repetitive ringing cycle of 3/4 second on, 1/2 second off, 3/4 second on, 3 seconds off.
Intercom	A repetitive ringing cycle of one second on, one second off.
Manual Signal	A 1/4 second beep tone that is muted if your phone is busy. Also indicated by lighting the BUZZ LED for 1/2 second.
Call Waiting	A 1/4 second beep tone, accompanied by a display showing the current call that is waiting.

3.2.4 Displays and Messages

Numerous displays and messages appear during programming and routine operation of the Featurephone. These are divided into three categories: automatic displays, source displays, and feature button related displays. Automatic displays are those which occur automatically in the course of normal operation, e.g., "PLEASE DIAL", and are shown in Table 3.4. Source displays indicate the source placing the call. Inside and outside call source display formats are shown in Tables 3.5 and 3.6. Feature **button related** programming displays result from the programming sequence of any of the feature buttons and contain various prompts and messages.

3.3 Preparing the Featurephone for Operation

Both the Digital Featurephone and the Analog Featurephone require downloading from the system data base prior to operation. Downloading the Digital Featurephone is similar to the downloading of the Analog Featurephone except:

(a) Adding a new Voice Control Interface to Periphery (VCIP) Card or Data Voice Control interface to Periphery (DVCIP) card physically to the system will cause a download to occur to the card.

(b) Adding a new Digital Featurephone to the system will cause a download to occur to the Digital Featurephone.

Table 3.4 Automatic Displays

CONDITION	DISPLAY
Dial Tone Heard	"PLEASE DIAL"
Permanent Time Out Occurs	"PLEASE HANG UP"
Partial Dial (Dial Time Out)	"PLEASE REDIAL"
Off-Hook, No Line Selected	"WHAT LINE"
Manual Dialing in Progress	"XXXX ---" Shown as dialing progresses
Automatic Dialing (Repertory, Last Number)	"JON J JONES" if Alpha Stored "XPXX ---" if Number Stored
Vacant Code or Unused Number	"CANNOT COMPLETE"
Incorrect Button Depression	"INVALID REQUEST"

NOTE: Actual messages may vary depending on conditions causing the display.

Table 3.5 Inside Call Source Display Format

There are two basic types of inside call displays. They are shown below using the letters X, A, C, and Y for the different messages that are possible. The other messages are given below. The call displays are:

C# =	XXXX AAAAAAAA
	OR
C# =	XXXX CCC YYYY
XXXX =	4-Digit Directory Number (DN) of calling party The calling number will be displayed for a minimum of 60 seconds after going off-hook.
AAAAAAAA =	NORMAL INTERCOM PRIORITY (Executive Priority) WAIT ORG (Call Wait Originate) CALLBACK CONF RCL (Conference Recall) REMINDER HOLD RET (Hold Retrieve) PARK RET (Park Retrieve) WAIT RET (Wait Retrieve) OVERRIDE (Busy Override) RCL PARK (Recall Park) BARGE IN TRANSFER (Call Transfer)
CCC =	FWA (Forward All) FWN (Forward No Answer) FWB (Forward Busy) PI (Call Pickup) YYYY= Forward from/Pickup from 4-digit DN

Table 3.6 Outside Call Source Display Format

There are two basic types of outside call displays. They are shown below using the letters X, B, C, and Y for the different messages that are possible. The other messages are given below. The call displays are:

TTT	XXXBBBBBBBB
	OR
TTT	xxx ccc YYYY
TTT = (Trunk Type)	COT (Central Office Trunk) FXT (Foreign Exchange Type) TIE (Tie Trunk) WTS (WATS Trunk) CLR (Toil Operator Trunk) DIC (Dictation Access Trunk) PAG (Paging Access Trunk) REC (Recorded Announcement) CAS (Centralized Attendant)
XXX =	3-digit trunk number
BBBBBBBB =	NORMAL CALLBACK (Hold Retrieve) PARK RET (Park Retrieve) WAIT RET (Wait Retrieve) RCL PARK (Recall Park) TRANSFER (Call Transfer)
ccc =	FWA (Forward All) FWN (Forward No Answer) FWB (Forward Busy) PI (Call Pickup)
YYYY =	Forward from/Pickup from 4-digit DN

3.3.1 Downloading

When the system is initially brought into operation, or a new Featurephone is connected to the system, the first action initiated by the Featurephone is to make a download request to the system. Download may also be initiated as a maintenance request, activating a sequence which sends information from the system's memory to the Featurephone memory.

Upon request to the switch for download to begin, the Featurephone display shows “WTG FOR LOAD XXXX”, where XXXX is the date the ROM portion of the Featurephone program was created. While the actual download is taking place (6-7 minutes), the Featurephone will display “LOADING XXXX AA”, where XXXX is the date the download portion of the program was created, and AA is the block in memory being loaded. Download, when complete, provides the Featurephone with all necessary system operating parameters, including specific site-dependent data base program information. The display shows flashing time and date when download is complete.

A single-line Featurephone now has the top four buttons adjacent to the keypad preprogrammed for specific functions. Starting at the top, these functions are MONITOR, DO NOT DISTURB, FLASH, and NEW CALL. A multi-line set will have two additional buttons programmed. The fifth button is programmed for HOLD and the eighth button for LINE. All other buttons are programmed as REPERTORY DIAL in downloading and must be reprogrammed by the user for other desired features.

Certain control functions are also programmed in the download operation. These are RING VOLUME, RING PITCH, CLOCK DISPLAY, CLOCK TYPE, PAUSE DURATION, CALL FORWARD NO ANSWER, and REFERENCE.

The feature buttons and control functions programmed in download are user reprogrammable, thus allowing the user to tailor the Featurephone to best suit the situations.

All programming functions associated with the Featurephone are accomplished in a simple sequential manner. Confirmation or alerting tones accompany each operation as do the flashing, fluttering, or steady LED indications, and the displaying of pertinent information in a menu-type presentation, and prompts the user throughout any programming sequence. Changes to any programmed instruction can be accomplished at any time the Featurephone is not in use. Control function reprogramming is covered in Paragraph 3.4 and feature programming/reprogramming are listed alphabetically in Paragraph 3.5.

3.3.2 Operational Verification (User Diagnostics)

The following diagnostic tests enable the user to perform a check on the instrument’s LEDs, feature buttons, keypad buttons, display, and audible signals.

User diagnostics are accessed by depressing #. The test codes shown in Table 3.7 are assigned to operate the tests which follow.

Table 3.7 Test Codes

TEST CODES	TEST
2	Audible Signals
3	LED, Feature Buttons, Keypad
4	Display

Perform the following steps to verify operation of LEDs, feature buttons, keypad, audible signals, and display. Diagnostics are accessed by depressing #.

(a) Audible Signals Test (Test Code 2)

STEP	OPERATION	RESULT
1	Depress and hold #.	(a) Beep heard. (b) "DIAGNOSTICS"
2	Release #.	"WHICH TEST = ?"
3	Depress 2.	(a) "AUDIBLE TEST" (b) Audible signals heard once each. (c) Display clears.

(b) LEDs, Feature Buttons, Keypad Test (Test Code 3)

STEP	OPERATION	RESULT
1	Depress and hold #.	(a) Beep heard. (b) "DIAGNOSTICS"
2	Release #.	"WHICH TEST = ?"
3	Depress 3.	(a) "LED TEST" (b) All LEDs on for 2 seconds.
4	Starting with key 1, proceed left to right, top to bottom depressing each key in turn.	(a) Display fills with characters depressed. (b) 2-3 seconds after # released, "BUTTON TEST".

NOTE: In the following tests, the sequences described must be followed, or else a programming error will result.

STEP	OPERATION	RESULT
5	Starting at the top left, depress feature button.	The specific feature programmed is displayed.

STEP	OPERATION	RESULT
6	Release feature button.	Display remains.
7	Continue down the button row(s) performing Steps 5 and 6 above.	Same as Steps 5 and 6 above.
8	2-3 seconds after last button is released.	(1) Double beep heard. (2) Display clears.

(c) Display Test (Test Code 4)

STEP	OPERATION	RESULT
1	Depress & hold #.	(a) Beep heard. (b) "DIAGNOSTICS"
2	Release #.	"WHICH TEST =?"
3	Depress 4.	(a) The following each appear for 2-3 seconds in sequence. (1) "DISPLAY TEST" (2) All LCDs active. (3) Blank display (4) Alpha 1: "ABCDEFGHIJKLM" (5) Alpha 2: "NOPQRSTUVWXYZ" (6) Numeric: "1234567890" (7) Special Characters: "*# + /:?&." (b) Display clears.

NOTE: Depress and hold designated feature button firmly until the required number of beeps are received.

3.4 Programming Control Parameters

All control functions are programmed to default values in the initial downloading operation (except time and date). The FeatureComm III/IV has preprogrammed levels of pitch and volume for the station ringer, timing for a Pause, Call Forwarding, and display format for Time and Date (Clock). The controls are programmable and may be changed by utilizing the following Control Codes (see Table 3.8 and the following examples). Control programming is accessed by depressing *; administration (set time and date) by 0.

Table 3.8 Control Codes

CONTROL CODES	CODES
2	Ring Pitch
3	Ring Volume
5	Clock Display
6	Clock Type
7	Pause Duration
8	Call Forward On No Answer Duration
9	Preference (Multi-Line Sets Only)

3.4.1 Call Forward on No Answer Duration (Control Code 8)

STEP	OPERATION	RESULT
1	Depress and hold *.	(a) Beep heard. (b) "CONTROL PROGRAM"
2	Release *.	"WHICH CONTROL = ?"
3	Depress 8 (Control code for CALL FORWARD).	"FWD NO ANS = 10SEC"
NOTE:	Any of these timing choices may be selected. If number 1 is selected, calls will be forwarded after 10 seconds; number 2 after 20 seconds; and number 3 after 30 seconds.	
4	Enter the appropriate number.	(a) "FWD NO ANS = 10SEC" if 1 is entered. (b) Beep heard. (c) "NEW ENTRY STORED"
5	Depress and hold *.	
6	Release *.	Display clears.

3.4.2 Clock Display (Control Code 5)

STEP	OPERATION	RESULT
1	Depress and hold *.	(a) Beep heard. (b) "CONTROL PROGRAM"
2	Release *.	"WHICH CONTROL = ?"
3	Depress 5.	"1 = DISPLAY; 0 = NONE"
NOTE:	Choose one of the following: 1 - To display time and date whenever the station is idle (feature button not required). 0 - For no display of time and date except via feature button operation (reprogram a feature button to operate Time and Date).	
4	Depress 1.	"DISPLAY ON IDLE"
5	Depress 0.	"NO IDLE DISPLAY"
6	Depress and hold *.	(a) Beep heard. (b) "NEW ENTRY STORED"
7	Release *.	Display clears.

3.4.3 Clock Type (Control Code 6)

STEP	OPERATION	RESULT
1	Depress and hold *.	(a) Beep heard. (b) "CONTROL PROGRAM"
2	Release *.	"WHICH CONTROL = ?"
3	Depress 6 (Control Code for Clock Type).	"1 = 12HR; 2 = 24HR"
4	Depress 1.	"CLOCK = 12 HOUR"
	OR	
5	Depress 2.	"CLOCK = 24 HOUR"

STEP	OPERATION	RESULT
NOTE:	Choose the desired clock type and depress the appropriate key. If 12-hour time is chosen, the HH:MM is followed by A or P for AM or PM, respectively. If 24-hour time is chosen, the HH:MM display will not be followed by an A or P, and HH will range from 00 to 23.	
6	Depress and hold *.	(a) Beep heard. (b) "NEW ENTRY STORED"
7	Release *.	Display clears.

3.4.4 Pause Duration (Control Code 7)

STEP	OPERATION	RESULT
1	Depress and hold *.	(a) Beep heard. (b) "CONTROL PROGRAM"
2	Release *	"WHICH CONTROL = ?"
3	Depress 7 (Control Code for Pause Duration).	"PAUSE = ?"
NOTE:	Choose one of the following: 2, to program a 2.0 second Pause Duration 3, to program a 2.5 second Pause Duration 4, to program a 3.0 second Pause Duration	
4	Depress and hold *.	(a) Beep heard. (b) "NEW ENTRY STORED"
5	Release *.	Display clears.
NOTE:	If an invalid entry is made, a triple beep will sound indicating an error. The selection process may continue at this point.	

3.4.5 Preference (Multi-Line Sets Only; Control Code 9)

STEP	OPERATION	RESULT
1	Depress and hold *.	(a) Beep heard. (b) "CONTROL PROGRAM"

STEP	OPERATION	RESULT
2	Release *.	"WHICH CONTROL = ?"
3	Depress 3 (control code for preference).	"PREF = 10100"
4	Depress * to see preference types.	"PREF = R I P L I"
5	Enter 0 for OFF or 1 for ON.	
6	Depress # to go to next field. Continue for 5 fields.	0s and 1s as entered. "XXXXX"
7	Depress and hold *.	"NEW ENTRY STORED"
8	Release *.	Display clears.

Preference Types

The five different types of preference include Incoming and Outgoing.

R = Incoming Ringing: The line with an incoming call, with audible alerting, will automatically be seized when going off-hook.

I = Incoming, Non-Ringing: The line with an incoming call, with visual alerting only, will automatically be seized when going off-hook.

P = Prime Line: The line that is the prime control line will automatically be seized when going off-hook to originate a call. The secondary LED associated with this line button will remain on steady to indicate this prime line preference.

L = Last Line Used: The line that was last used will automatically be seized when going off-hook to originate a call. The secondary LED associated with this line button will remain on steady to indicate this last line used preference.

I = Idle Line: Any one idle line on the instrument will automatically be seized when going off-hook to originate a call. The secondary LED associated with this line button will remain lit steady to indicate this idle line preference.

For any combinations of the different types of preference, incoming and ringing has the highest priority down to idle line, which has the lowest priority. For example, if preference has been programmed for incoming, ringing, and prime line, the I-Use LED for the prime line will always be on when on-hook; but if an incoming call is received, and the handset lifted, the ringing line will automatically be seized.

Preference selection is intended to be dynamic and constantly changing as the lines are made busy and go idle. The choosing of a line by a preference feature is intended to mean its “predesignation for use” if and when the instrument user goes off-hook. This should not imply either a reservation of a line or seizure of the line.

3.4.6 Ring Pitch (Control Code 2)

STEP	OPERATION	RESULT
1	Depress and hold *.	(a) Beep heard. (b) “CONTROL PROGRAM”
2	Release *.	“WHICH CONTROL = ?"
3	Depress 2. (Control program for pitch.)	“TONES = 1 through 8”
4	Depress desired tone (l-8).	(a) “PITCH = 1" (b) Ringing heard for about 1 second.
5	Adjust pitch as desired by depressing the appropriate number	(a) Display reflects number. (b) Ringing tone higher.
6	If no entry is made, the last tone sounded is the programmed pitch.	(a) Beep heard. (b) “NEW ENTRY STORED”
7	Release *.	Display clears.

3.4.7 Ring Volume (Control Code 3)

STEP	OPERATION	RESULT
1	Depress and hold *.	(a) Beep heard. (b) “CONTROL PROGRAM”
2	Release *.	“WHICH CONTROL = ?"
3	Depress 3.	“VOLUME = 1 through 8”
4	Depress desired volume (l-8)	(a) “VOLUME = 1" (b) Ringing heard for about 1 second.
5	Adjust volume as desired by entering the appropriate number.	(a) Display reflects number. (b) Ringing tone higher.

STEP	OPERATION	RESULT
6	Depress and hold *.	(a) Beep heard. (b) "NEW ENTRY STORED"
7	If no entry is made, the last level sounded will be the programmed level.	
8	Release *.	Display clears.
NOTE:	If you have a FeatureComm III/IV instrument, the preprogrammed Ring Volume is 1. If you have a FeatureComm V/VI, it is 4.	

3.4.8 Setting the Time and Date

STEP	OPERATION	RESULT
1	Depress and hold 0 (Access Code for Administration).	(a) Beep heard. (b) "ADMINISTRATION"
2	Release 0.	"00:00A DAY 00/00"
3	Enter hours (01-12 or 00-23), #, minutes (00-59), #.	"XX:XXA DAY 00/00 *"
4	Enter AM/PM (A or P in 12 hr mode, no entry 24 hr), #.	"XX:XXA DAY 00/00 *"
5	Enter day, #.	"XX:XXA MON 00/00" if Monday entered.
6	Enter month (01-12), #, date (01-31).	"XXXXAMON XX/XX"
7	Depress and hold 0.	"XXXXAMON XX/XX" time and date as entered.
8	Release 0.	Display clears.
NOTE:	* appears in the display indicating that other information is available.	

3.5 Function Codes, Feature Button Assignment and Data Entry

Feature buttons are programmed/reprogrammed as outlined in the following paragraphs and charts. Codes required for the various functions are shown in alphabetical and numerical order in Tables 3.9 and 3.10, respectively.

Table 3.9 Function Codes Required for Programming/Reprogramming the Feature Buttons, Listed Alphabetically

FUNCTION	CODE	FUNCTION	CODE
Account Code	40	Lines (Multi-Line Sets Only)	42
Automatic Call Announcing	27	Manual Signal	39
Automatic Callback	37	Message Leaving	25
Automatic intercom	23	Message Waiting	24
Bad Line	45	Monitor	04
Busy Override	34	New Call	05
Call Announce	26	Privacy Release	46
Call Forward	19	Release	17
Call Hold	21	Reminder	13
Call Park	32	Report Tone	44
Call Waiting	30	Repertory/Feature Access	09
Conference	38	Ringer Cutoff	01
Data Option	41	Save Number Redial	07
Data Protection	33	Shift	43
Dial Display	16	Speakerphone	18
Direct Station Selection (DSS)	11	Special Call Forward	20
Dual Access	16	Special Call Wait	31
Executive Priority Call	35	Station Lock	02
Extended Call Pickup	29	Station Speed Calling	08
Extended Group Select	12	System Speed Calling	47
Flash	03	Timer/Elapsed Time	14
Forced Busy	36	Time and Date (Clock)	15
Group/Directed Call Pickup	28		
Intercom	22		
Last Number Redial	06		

Table 3.10 Function Codes Required for Programming/Reprogramming the Feature Buttons, Listed by Code

CODE	FUNCTION	CODE	FUNCTION
00	Unassigned	24	Message Waiting
01	Ringer Cutoff	25	Message Leaving
02	Station Lock	26	Call Announcing
03	Flash	27	Automatic Call Announcing
04	Monitor	28	Group/Directed Call Pickup
05	New Call	29	Extended Call Pickup
06	Last Number Redial	30	Call Waiting
07	Save Number Redial	31	Special Call Wait
08	Station Speed Calling	32	Call Park
09	Repertory/Feature Access	33	Data Protection
10	Dual Access	34	Busy Override*
11	Direct Station Selection (DSS)	35	Executive Priority Call*
12	Extended Group Select	36	Forced Busy
13	Reminder	37	Automatic Callback
14	Timer/Elapsed Time	38	Conference
15	Time & Date (Clock)	39	Manual Signal
16	Dial Display	40	Account Code*
17	Release	41	Unassigned Data Option
18	Speakerphone (Optional)	42	Line (Multi-Line Sets Only)
19	Call Forward	43	Shift
20	Special Call Forward	44	Remote Tone
21	Call Hold	45	Bad Line
22	Intercom*	46	Privacy Release
23	Automatic Intercom*	47	System Speed Calling*

NOTE: Any number of buttons can be programmed to perform:

- Save Number Redial
- Repertory/Feature Access
- Dual Access
- Secretarial Answering Service (DSS)
- Automatic Intercom
- Automatic Call Announcing
- Manual Signal
- Lines (Multi-Line Sets Only)

* Requires special programming.

3.51 Programming/Reprogramming a Feature on a Button

STEP	OPERATION	RESULT
	Remain on-hook.	
1	Depress and hold feature button to be assigned change.	(a) Single beep heard. (b) "FEATURE NAME" of existing feature displayed.
2	Release the feature button.	(a) LED flashes throughout programming. (b) "CONTENTS STORED" or "FEATURE NAME" displayed.
3	Depress and hold 2.	(a) Double beep heard. (b) "CHANGE FUNCTION"
4	Release 2.	(a) LED flashes. (b) "FUNCTION CODE = __."

NOTE: The * indicates that additional information is available. Repeated depressions of the * will display the features and function codes in a scrolling action. Tables 3.9 and 3.10 show the code numbers to be entered to select a function for a feature button.

STEP	OPERATION	RESULT
5	Entered desired function code via keypad number (2 digits).	(a) "CODE NUMBER = XX" (b) "FEATURE NAME" displayed.
6	Depress # where XX represents the code number digits.	

NOTE: If # is not entered in Step 6, three beeps will be heard, the phone will EXIT programming and the display will clear or revert to time and date.

STEP	OPERATION	RESULT
7	Depress and hold the feature button.	(a) Single beep heard. (b) "NEW ENTRY STORED"
8	Release feature button.	(a) LED off. (b) Display clears or reverts to time and date.
9	Change the feature button label to reflect the new feature selected.	

To Verify

OPERATION	RESULT
1. Depress and hold the button changed. 2. Release button and press again. Display clears or reverts to time and date.	(a) Beep heard. (b) "ABCD--" name of feature programmed.

NOTES:

1. If, at any point in programming a feature, no action is taken for 60 seconds, a triple beep will be heard, the LED will go dark, and the display will revert to whatever was displayed prior to the beginning of programming. All attributes that were assigned to that button up to that point will be disregarded.
2. If error beep is heard when entering a function code, it may be because that feature button is already programmed, or the entry attempted is invalid for that instrument (such as a line button on a single-line set).
3. When programming a shifted button (second feature), the bottom (alternate) LED of that button will flash during programming.

3.5.2 Entering Data

Data is entered using the keypad at the appropriate step in programming or when prompted by the display. The entered data is then displayed until entered into memory by the next button depression.

3.5.3 Cancellation of Programming

Programming can be canceled at any point by taking the handset off-hook or by a rapid depress/release of the feature button being programmed. Programming must then be re-initiated rather than be resumed at the point of cancellation to avoid the repetitious beeping caused by a programming error. There is no change in the previously stored data when cancellation occurs.

3.5.4 Special Data Programming/Entry

Special data programming/entry is covered in the following paragraphs and figures. These include pause, flash, and blank entries; alpha programming, next field and list requests.

PAUSE • used in storing frequently called numbers to insert proper timing between digits (which normally occurs when you manually dial an outside line and wait to receive a second dial tone.) To enter a Pause:

STEP	OPERATION	RESULT
1	Depress and hold #.	Beep heard.
2	Release #.	

FLASH - used in storing a feature access code (such as in the dual access and repertory dialing features). To enter a Flash:

STEP	OPERATION	RESULT
1	Depress and hold *.	Beep heard.
2	Release *.	

BLANK - used to space out characters on the display. To enter a Blank:

STEP	OPERATION	RESULT
1	Depress and hold 0.	Beep heard.
2	Release 0.	

ALPHA - Alpha entries may be made in the first field of station speed calling and reminder features. Alpha entries are made using the alphabetic designations on the keypad and the *, 0, and # to designate the left (*), center (0), or right (#) letter on the button. (See Figure 3.2.) To make a numeric entry in the alpha field, key in the desired digit twice.

NOTE: If you have a FeatureComm III/IV instrument, you may also make an alphabetic entry into the first field when programming Repertory Dial and Message Waiting.

EXAMPLE: ENTER "ROD"

STEP	OPERATION	RESULT
1	Depress 7, 0.	(a) Beep heard. (b) "R"
2	Depress 6, #.	(c) Beep heard. (d) "RO"
3	Depress 3, *.	(e) Beep heard. (f) " ROD"

TO ENTER ROD:

1. DEPRESS R KEY (7 KEY).
2. DEPRESS 0 TO INDICATE THAT R IS THE CENTER LETTER ON THE 7 KEY.
3. DEPRESS 0 (6 KEY).
4. DEPRESS # TO INDICATE THAT 0 IS THE RIGHTMOST LETTER ON THE 6 KEY.
5. DEPRESS D (3 KEY).
6. DEPRESS • TO INDICATE THAT D IS THE LEFTMOST LETTER ON THE 3 KEY.

NOTE: A beep will be heard after each entry.

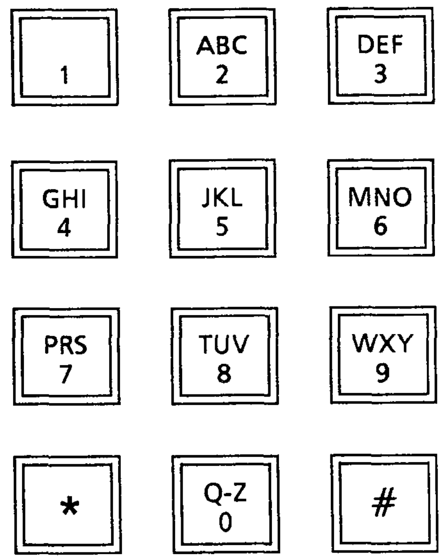


Figure 3.2 Entering Alpha Information in Integrated Featurephone

NEXT FIELD REQUEST - Station Speed Calling, Reminder, Duration Alarm, Message Leaving, Direct Station Sèlection (DSS) line button, and time and date programming contain fields for entering alphabetic identifiers, phone numbers, hours, minutes, day, month, and date. Alter entering data in a field or to skip by the field without changing its contents, enter a # to go to the next field.

Depress # (Cursor moves to first item in the next field)

The next field command (#) allows you to "edit" a previous entry; i.e., just enter programming and depress # until the display shows the field which you wish to change (such as the time of a reminder), enter the new data, and exit programming by depressing and holding the feature button.

LIST REQUEST - Lists contain items stored to the reminder and station speed calling feature buttons. A third index lists the function codes of the 48 programmable features. During programming/usage, a "*" will appear on the display to indicate that a list is available. Depressing the * key on the keypad allows scrolling through the list. Each item within an index is listed with a code number as shown in Table 3.11.

To Request List:

Display shows "*" at the end of the character line.

Depress *.

(Index items appear with each * entered or a "help" message appears on the display. Enter your selected item and the display will advance to the next prompt.)

Table 3.11 List Types

LIST TYPE	NUMBER OF ITEMS	CODE NUMBERS
Function Code	48 Items	00 through 47 (except 11 and 12)
Reminder	8 Items	1 through 8
Speed Call	20 Items	10 through 29

After a list has been requested and items are displayed, you may dial # to see the next field of a particular item (such as the phone number associated with a Reminder item). Depress * again to see the next item in the list.

3.6 Alphabetic Featurephone Button Programming

Some feature buttons require additional programming after assignment to facilitate/enhance their operation. Instructions are provided in the following paragraphs and are arranged alphabetically followed by feature number in parentheses.

Feature buttons not requiring additional programming are listed in Table 3.12.

3.6.1 Automatic Call Announcing (27)

STEP	OPERATION	RESULT
1	Depress and hold CALL ANN	(a) AUTO CALL ANN LED on. (b) Beep heard. (c) "AUTO CALL ANNOUNCE"
2	Release.	(a) CALL ANN LED flashes. (b) "DN = _"
3	Enter a new 3- or 4-digit number.	"DN - XXXX"
4	Depress #.	"VALIDATING DN"
	If valid,	"STORE ENTRY"
	If invalid,	(a) Triple beep heard. (b) "DN = XXXX"
5	Another number may be tried.	
6	To store, depress and hold CALL ANN.	(a) CALL ANN LED on steady. (b) Beep heard. (c) "NEW ENTRY STORED"

Table 3.12 Features Not Requiring Additional Programming

FUNCTION	CODE	FUNCTION	CODE
Account Code	40	Forced Busy	36
Automatic Callback	37	Group/Directed Call Pickup	28
Bad Line	45	Intercom	22
Busy Override	34	Last Number Redial	06
Call Announcing	26	Message Leaving	25
Call Hold	21	Monitor	04
Call Park	32	New Call	05
Call Waiting	30	Privacy Release	46
Conference	38	Reminder	13
Data Protection	33	Remote Tone	44
Dial Display	16	Save Number Redial	07
Dual Access	10	Shift	43
Executive Priority Call	35	Speakerphone	18
Extended Call Pickup	29	System Speed Calling	42
Extended Group Select	12	Time and Date	15
Flash	03	Timer	14

STEP	OPERATION	RESULT
7	Release.	(a) CALL ANN LED off. (b) Display clears or reverts to time and date.

NOTE: Call Announcing may be programmed only within the intercom group associated with the instrument.

3.6.2 Automatic Intercom (23)

STEP	OPERATION	RESULT
1	Depress and hold AUTO ICOM.	(a) AUTO ICOM LED on. (b) Beep heard. (c) "AUTO INTERCOM"
2	Release.	(a) AUTO ICOM LED flashes. (b) "NUMBER = ---"
3	Enter desired number.	"DN = XXXX"
4	Depress #.	"VALIDATING DN"
	If valid,	"STORE ENTRY"
	If invalid,	(a) Three beeps heard. (b) "NUMBER = XXXX"
5	Another number may be tried.	
6	To store, depress and hold "AUTO ICOM".	(a) AUTO ICOM LED on steady (b) Beep heard. (c) "NEW ENTRY STORED"
7	Release.	(a) A ICOM LED off. (b) Display clears or reverts to time and date.

3.6.3 Call Forward (19)

STEP	OPERATION	RESULT
1	Depress and hold FORWARD button.	(a) Beep heard. (b) CALL FORWARD LED on. (c) "CALL FORWARD" displayed.
2	Release FORWARD button.	(a) CALL FORWARD LED flashes. (b) "TYPE = ?"
3	Enter number of type of forwarding desired (l-4). 1 = Forward on busy 2 = Forward on no answer	"TYPE = X"

STEP	OPERATION	RESULT
	3 = Forward on busy and no answer	
	4 = Forward all calls.	
4	Depress # to change fields.	"NUMBER = _"
5	Enter 3- or 4-digit number to which calls will be forwarded.	(a) "NUMBER = 4321" (b) "VALIDATING DN", then "STORE ENTRY"
6	To store, depress and hold FORWARD.	(a) Beep heard. (b) "NEW ENTRY STORED"
7	Release.	(a) FWD LED off. (b) Display clears or reverts to time and date.

3.6.4 Direct Station Selection (DSS) (11)

STEP	OPERATION	RESULT
1	Depress and hold DSS.	(a) DSS LED on. (b) Beep heard. (c) "DSS"
2	Release.	(a) DSS LED flashes. (b) "XXXX R G" (see note 1)
3	Enter desired number.	"XXXX" (see note 2)
4	Depress #.	(a) VALIDATING DN" (b) "XXXX R G" with the cursor under the alerting type field.
5	Enter alert type. 1 = Immediate visual and audible 2 = Immediate visual and no audible 3 = Immediate visual and delayed audible 4 = Delayed visual and delayed audible	"XXXX 4 G"
6	Depress #.	"XXXX 4 G"

STEP	OPERATION	RESULT
7	Enter group type (see note 3). 1 = Full DSS 2 = Alternate DSS 3 = Call Only DSS	"XXXX 4 3"
8	To store, depress and hold DSS.	(a) DSS LED on steady. (b) Beep heard. (c) "NEW ENTRY STORED"
9	Release.	(a) DSS LED off. (b) Display clears or reverts to time and date.

NOTES:

1. If the number is invalid, the display shows: XXXX R G with the cursor under the number field and three error beeps sound. You can then re-enter the directory number.

2. This display indicates the following: XXXX is the directory number currently assigned "-" - it is blank if there is none; R is the alerting type; and G is the DSS group type.

3. If the DSS TYPE (l-3) or the type of ALERTING (l-4) is invalid, an error beep will be received. The DSS types are:

   1 = Full DSS: Answering button for group normally serviced.

   2 = Alternate DSS: Answering button for the alternate (extended) group serviced.

   3 = Call Only DSS: Always functions as a repertory dialing button.

3.6.5 Dual Access (10)

Dual Access allows the operation of two access codes with one button.

STEP	OPERATION	RESULT
1	Depress and hold DUAL ACCESS.	(a) DUAL ACCESS LED on. (b) Beep heard. (c) "DUAL ACCESS"
2	Release.	(a) DUAL ACCESS LED flashes. (b) "1 + " (see note)

STEP	OPERATION	RESULT
3	Enter desired access code.	"1 = X XXXXXXX" (phone number)
4	To store, depress and hold DUAL ACCESS.	(a) DUAL ACCESS LED on steady. (b) Double beep heard. (c) "NEXT ENTRY"
5	Release.	(a) DUAL ACCESS LED flashes. (b) "2 = "*
6	Enter desired number	"2 = X XXXXXXX" (phone number)
7	To store, depress and hold DUAL ACCESS.	(a) DUAL ACCESS LED on steady. (b) Beep heard. (c) "NEW ENTRY STORED"
8	Release.	(a) DUAL ACCESS LED off. (b) Display clears or reverts to time and date.

NOTE: If the Dual Access button was previously programmed, the repertory dial number previously programmed will appear in the display.

3.6.6 Line (42)

Line may be programmed on multi-line sets only.

STEP	OPERATION	RESULT
1	Depress and hold line.	(a) LINE LED on. (b) Beep heard. (c) "LINE"
2	Release.	(a) LINE LED flashes. (b) "XXXX R" (see note 1)
3	Enter desired number.	"XXXX" as entered (see note 2).
4	Depress #.	"VALIDATING DN" followed by "XXXX R".
5	Enter alert type (see note 3).	
6	To store, depress and hold LINE.	(a) LINE LED on steady. (b) Single beep heard. (c) "NEW ENTRY STORED"

STEP	OPERATION	RESULT
7	Release.	(a) LINE LED off. (b) Display clears or reverts to the time and date.

NOTES:

1. This display indicates the following: XXXX is the directory number currently assigned - it is blank if there is none, and R is the alerting type.

2. If the number is invalid, the display shows: XXXX R G with the cursor under the number field and three error beeps sound. Another directory number can then be entered.

3. If the type of ALERTING (l-4) is invalid, an error tone will be received. The ALERTING types are:

   1 = Immediate Visual and Audible
   2 = Immediate Visual and No Audible
   3 = Immediate Visual and Delayed Audible
   4 = Delayed Visual and Delayed Audible

3.6.7 Manual Signal "BUZZ" (39)

STEP	OPERATION	RESULT
1	Depress and hold BUZZ.	(a) BUZZ LED on. (b) Beep heard. (c) "MANUAL SIGNAL"
2	Release.	(a) BUZZ LED flashes. (b) "DN = ?"
3	Enter desired number.	(a) "DN = XXXX" as entered, followed by (b) "VALIDATING DN" (see note)
4	To store, depress and hold BUZZ.	(a) BUZZ LED on steady. (b) Beep heard. (c) "NEW ENTRY STORED"
5	Release.	(a) BUZZ LED off. (b) Display clears or reverts to the time and date.

NOTE: If requested number is not verified, error tone is heard, the LED is extinguished, and the entry may be made again.

3.6.8 Message Waiting (24)

STEP	OPERATION	RESULT
1	Depress and hold WAIT.	(a) WAIT LED on. (b) Beep heard. (c)
2	Release.	(a) WAIT LED flashes (b) Display cleared.
3	Enter Message Center number.	"XXXX"
4	To store, depress and hold WAIT.	(a) WAIT LED on steady. (b) Beep heard. (c) "NEW ENTRY STORED"
5	Release.	Display cleared. (a) WAIT LED flashes. (b) "MESSAGE WAITING"

NOTE: If you have a FeatureComm III/IV instrument, key in the desired Alpha Entry (Message Center Attendant's name, etc.) and then a # to move the cursor to the next field.

3.6.9 Repertory Dial (09)

STEP	OPERATION	RESULT
1	Depress and hold REP DIAL.	(a) REP DIAL LED on. (b) Beep heard. (c) "REP DIAL"
2	Release.	(a) REP DIAL LED flashes. (b) Display cleared.
3	Enter desired number, with pauses, etc.	Phone number or access code dialed appears.
4	To store, depress and hold REP DIAL.	(a) REP DIAL LED on steady. (b) Beep heard. (c) "NEW ENTRY STORED"

STEP	OPERATION	RESULT
5	Release.	(a) REP DIAL LED off. (b) Display clears or reverts to the time and date.

NOTES:

1. See Special Data Programming /Entry for pause programming information.
2. If you have a FeatureComm III/IV instrument, key in the desired Alpha Entry and then a # to move the cursor to the next field.

3.6.10 Special Call Forward (20)

STEP	OPERATION	RESULT
1	Depress and hold SP FWD.	(a) SP FWD LED on. (b) Beep heard. (c) "SPEC CALL FWD"
2	Release.	(a) SP FWD LED flashes. (b) "TYPE = _"
3	Enter desired type (e.g., 2) 1 = Forward Specific Calls 2 = Forward Except	"TYPE = 2"
4	Depress # to change fields.	"NUMBER = _"
5	Enter desired number.	"NUMBER = XXXX"
6	Depress #. Enter up to three more numbers as in Steps 5 and 6 above, or depress # to skip field.	"NUMBER = _"
7	After last number of field.	"STORE ENTRY"
8	To store, depress and hold SP FWD	(a) SP FWD LED on steady. (b) Beep heard. (c) "NEW ENTRY STORED"

STEP	OPERATION	RESULT
9	Release.	(a) SP FWD LED off. (b) Display clears or reverts to the time and date

3.6.11 Special Call Wait (31)

STEP	OPERATION	RESULT
1	Depress and hold SP WAIT.	(a) SP WAIT LED on. (b) Beep heard. (c) "SPECIAL WAIT"
2	Release.	(a) SP WAIT LED flashes. (b) "TYPE = _?"
3	Enter type (e.g., 1). 1 = Wait Only 2 = Wait Except	"TYPE = 1"
4	Depress #.	"NUMBER = _"
5	Enter desired number.	"NUMBER = XXXX"
6	Depress #.	"NUMBER = _"
7	Enter up to three or more numbers as in Steps 5 and 6 above or depress # to skip field.	
8	After last number or field.	"STORE ENTRY"
9	To store, depress and hold SP WAIT.	(a) SP WAIT LED on steady. (b) Beep heard. (c) "NEW ENTRY STORED"
10	Release.	(a) SP WAIT LED off. (b) Display clears or reverts to the time and date.

3.6.12 Station Lock (02)

STEP	OPERATION	RESULT
1	Depress and hold LOCK.	(a) LOCK LED on. (b) Beep heard. (c) "STATION LOCK"
2	Release.	(a) LOCK LED flashes. (b) "LOCK CODE ="
3	Enter code.	
	If incorrect,	(b) Beep heard. (b) LOCK LED flashes. (c) "NEW CODE = ?"
	if correct,	"NEW CODE = ?"
	(Preprogrammed code is 000)	
4	Enter new code.	"CODE VERIFY = ?"
5	Enter new code again.	
	If incorrect,	(a) Triple beep heard. (b) "NEW CODE = ?"
	If correct,	"STORE LOCK CODE"
6	To store, depress and hold LOCK.	(a) LOCK LED on steady. (b) Beep heard. (c) "NEW ENTRY STORED"
7	Release.	(a) LOCK LED off. (b) Display clears or reverts to the time and date.

3.6.13 Station Speed Calling (08)

STEP	OPERATION	RESULT
1	Depress and hold SPEED.	(a) SPEED LED on. (b) Beep heard. (c) "STATION SPEED"
2	Release.	(a) SPEED LED flashes. (b) "CODE =_"
3	Enter code (10 to 29) (e.g., 19).	"19 = 19"

STEP	OPERATION	RESULT
4	Depress #.	"19 = "
5	Enter alpha information, e.g., GIGI.	"19 = GIGI"
6	Depress #.	Display cleared.
7	Enter desired number.	"X XXXXXXX"
8	To store, depress and hold SPEED.	(a) SPEED LED on steady. (b) Beep heard. (c) "NEW ENTRY STORED"
9	Release.	(a) SPEED LED off. (b) Display clears or reverts to the time and date.

NOTES:

1. The above may be repeated for an additional 19 entries.
2. See Special Data Programming/Entry for pause programming information.

3.7 Alphabetical Operation of Features

3.7.1 Account Code Entry *'ACCOUNT CODE" (40)

The user may enter an account code any time during an incoming or outgoing external call. The number (1 to 8 digits, data base programmed) is associated with the accounting data for the call and is printed as part of the CALL record produced by the optional message detail recording subsystem.

STEP	OPERATION	RESULT
	Beginning at any time from off-hook receiving dial tone until just prior to the time when the first party in a call releases.	
1	Depress ACCT button.	(a) ACCT button flashes at 120 IPM. (b) "CODE = _"
2	Enter up to an 8-digit account code via the keypad.	(a) LED goes dark. (b) "CODE = X XXXXXXX"
3	Enter #.	Display clears or reverts to the time and date.

NOTES:

1. A new account code may be entered at any time during the call, but only the last number entered will be associated with the call.
2. To abort an entry, depress the ACCT button before the number is entered.
3. If the account code is more than eight digits, only the last eight digits entered will be stored.

3.7.2 Automatic Callback "CAMP ON" (37)

Automatic Callback provides access to either of two PABX features: Station Camp-On with Callback or On-Hook Trunk Call Queuing. AUTOMATIC CALLBACK may be initiated whenever station busy tone is encountered in making an internal call, or when trunk busy tone is encountered in making an outside call.

STEP	OPERATION	RESULT
1	A line or trunk has been dialed and is busy.	If line (a) Station busy tone heard. (b) Line busy. If trunk (a) Momentary confirmation tone. (b) "ALL TRUNKS BUSY"
2	Depress C BACK.	If successful (a) C BACK LED on. (b) Confirmation tone heard. (c) "CALLBACK"
(Camp-on disallowed due to lack of system resources or another camp-on already in effect)		If unsuccessful (d) C BACK LED unchanged. (e) Reorder tone heard. (f) "REQUEST DENIED"
3	Go on-hook.	
4	If (1) and (2) above, and the called line/trunk becomes idle, and the instrument is idle.	(a) C BACK LED on. (b) Distinctive ringing. (c) "XXXX CALLBACK" for callback to line or (d) "TRK CALLBACK" for callback to trunk. (e) C BACK LED off.
5	Answer callback.	(a) Ringback tone for call to line. (b) Dial tone for call to trunk.

STEP	OPERATION	RESULT
		(c) "PLEASE DIAL" for call to trunk. (d) C BACK LED off.
6	Trunk call unanswered, several attempts will be made. On last attempt,	(a) Same as Steps 4a and 4b. (b) Camp-on terminated. (c) C BACK LED off.
7	If first (and only) line callback is not answered.	(a) Same as Steps 4a, b, and c. (b) Same as Steps 6b and c.
8	If instrument busy and a trunk recall is attempted. If instrument is busy, a line recall will not be attempted until station is idle.	Same as **Step 6a.**

NOTES:

1. C BACK may be aborted at any time busy tone is not heard by depressing the C BACK button.

2. For calls placed to trunks, via MERS, the request for dialing (Steps 5b and c) will not occur.

3.7.3 Automatic Line Seizure

Automatic Line Seizure allows the user to originate a call on a selected line with the handset on-cradle by depressing any of the following automatic dialing feature buttons:

AUTOMATIC CALL ANNOUNCING "AUTO ANNOUNCE"
AUTOMATIC INTERCOM "AUTO ICOM"
DIAL DISPLAY "D DISP"
DUAL ACCESS "DUAL ACCESS"
LAST NUMBER DIAL "LAST #"
MESSAGE WAITING "MS CTR"
REPERTORY DIAL "REP DIAL"
SAVED NUMBER DIAL "SAVE #"
STATION SPEED CALL "SPEED CALL"

or by depressing one of the following non-dialing feature buttons:

CALL ANNOUNCING "VOICE"
CALL PARK "PARK"
CALL PICKUP "PICKUP"

Operating any of the above feature buttons shall cause the instrument to turn on the speakerphone (if equipped) or the monitor.

NOTE: In the event the user of a multi-line set has not preselected a line or a line has not been automatically selected by the instrument via the preference feature, the instrument will delay turning on the speakerphone/monitor amplifier until a line has been manually selected.

(a) **Automatic Line Seizure, Single-Line Set**

STEP	OPERATION	RESULT
1	Depress desired feature button.	(a) Associated feature LED on. (b) Speakerphone feature LED on. **OR** (c) Monitor feature LED on. (d) Dial tone is heard. (e) "PLEASE DIAL" (even if auto dialing feature button is depressed).
2	Continue as with Basic Calling (Internal/External).	

(b) Automatic Line **Seizure, Multi-Line Set**

STEP	OPERATION	RESULT
1	(a) Manually select an idle line by depressing a line button.	(a) Associated I-USE LED on. (b) Preference line I-USE LED extinguishes (if preference had selected a line). (c) Dial tone heard. (d) "PLEASE DIAL"
	OR	
	b) Choose to use the line selected by the preference feature, if active.	(a) Preference line LED on. (b) Dial tone heard. (c) "PLEASE DIAL"
2	Continue as in step 1 above.	

MOTES:

1. If a line has not been selected prior to Step 2, the user must now do so in order for the sequence to continue. The LED of the associated selected line is turned on.

2. If the line selected is a non-dial line, i.e., a personal CO line, the prompt is not displayed. Instead, CO dial tone wil be heard as confirmation of successful **CO** seizure.

See also: Line Preference
Line Seizure
Line Selection

3.7.4 Bad-Line Reporting "BAD LINE" (45)

Bad-Line Reporting allows the identification of a faulty line to the system by pressing the BAD-LINE feature button while on the connection.

STEP	OPERATION	RESULT
	Starting from a connected call state.	
1	Depress "BAD LINE".	(a) BAD LINE LED on for duration of connection. (b) A report is generated to the maintenance port.

NOTE: If BAD LINE LED is already lit, the display will show "INVALID REQUEST".

3.7.5 Basic Calling (Internal/External)

Originating a basic call is virtually identical to POTS operation.

STEP	OPERATION	RESULT
1	Seize a line.	(a) Dial tone heard. (b) "PLEASE DIAL"
2	Enter a number via keypad.	"XXXX--" as entered.
	Select a line if preference is inactive (multi-line set).	"WHICH LINE?" (if multi-line set)
3a	(Called Party Idle)	(a) Ringback tone heard. (b) "RINGING"
3b	(Called Party Busy)	(a) Busy tone heard. (b) "BUSY"
3c	(All Trunks Busy)	(a) Reorder tone heard. (b) "ALL TRUNKS BUSY"

STEP	OPERATION	RESULT
3d	(Incorrect, non-working, or restricted number)	(a) Reorder tone heard. (b) “CANNOT COMPLETE”
4	To end or abandon call, go ON-HOOK.	

integrated Featurephone call termination differs from the POTS phone termination because the Integrated Featurephone decides how the call will be treated. When the call is offered, the Integrated Featurephone may:

(a) ACCEPT the call. An ACCEPT does not constitute an answer but rather indicates that the phone will notify its user of the presence of the call.

Once the call has been accepted, the phone and/or user may.

(1) ANSWER the call. (Phone will automatically answer a CALL ANNOUNCING call. See CALL ANNOUNCING.)

(2) FORWARD the call, either due to a manual forward or no answer determined by the phone. (See CALL FORWARD.)

(b) REJECT the call, indicating that the phone/line is busy.

(c) FORWARD the call. The phone may forward the call for a forward call or forward-busy condition.

See also: Call Source Display
Distinctive Ringing
Line Selection
Line Seizure

3.7.6 Busy Override “OVRIDE" (34)

Busy Override allows a properly classmarked user, upon reaching a busy destination, to break into the existing conversation. The Busy Override will cause the break-in for the designated target instrument regardless of the line the called party is busy on (Data-protected, Three-Way, Conference, and Forced Busy excluded).

STEP	OPERATION	RESULT
	Beginning from off-hook, busy tone heard.	
1	Depress "OVRIDE" .	OVERRIDE LED flashes at 120 IPM.

STEP	OPERATION	RESULT
	If successful,	(a) A 3-party connection is made. (b) OVRIDE LED is on. (c) Broken-into stations hear busy override tone, and if integrated Featurephone "XXXX OVERRIDE" displayed.
	If unsuccessful,	(a) OVRIDE LED off. (b) Reorder tone heard. (c) "REQUEST DENIED"
2	Both parties in the overridden conversation will receive.	

3.7.7 Call-Announcing "VOICE" (26, 27)

Call Announcing provides the ability to make calls within a restricted group of phones, and receive priority-call handling and automatic termination (answer) on multi-line sets. The special call-announcing tone and automatic activation of monitor/speakerphone provides the called party with the knowledge that the call is distinct from a normal line-to-line call and comes from within a restricted group.

The Call Announcing differs from a normal call in several respects:

(a) The call is automatically answered and an alerting tone is sounded if called party is idle.

(b) Distinctive intercom ringing is supplied at the called phone if the called phone is busy.

(c) The call cannot be forwarded.

(d) The call cannot be diverted with call pickup.

(e) The call originating can be completed even though the called line is busy (pseudo call waiting).

(f) The call will automatically release privacy on a current line call.

Call Announcing "VOICE"

STEP	OPERATION	RESULT
1	Go off-hook or depress monitor or speakerphone.	(a) Dial tone heard. (b) "PLEASE DIAL"
2	Depress VOICE.	VOICE LED on.
3	Release VOICE.	(a) VOICE LED flashes at 120 IPM. (b) "NUMBER = _"
4	Depress a DSS or line button. If DSS or line button does not permit call announcing, the depression is treated as an error.	(a) VOICE LED on. (b) "NUMBER = XXXX"
5	a. If call is answered	1/2-second tone burst heard.
	b. If called party is busy and receiving call-waiting treatment	Ringback tone heard.
	c. If called party is busy and is not receiving call-waiting treatment	Busy tone heard.

NOTES:

1 If call announcing is denied, you will receive the message "REQUEST DENIED" on the display, all related LEDs will revert to their original state, and FLASH must be used to get back to the party that was to be placed on hold, if any. The call announcing sequence is abandoned.

2. Call announcing calls cannot be placed from CO lines.

Automatic Call Announcing (27)

STEP	OPERATION	RESULT
1	Go off-hook.	Dial tone heard.
2	Depress A VOICE.	(a) VOICE LED flashes (b) "NUMBER + XXXX" (c) VOICE LED on.
3	Continue as with a normal line-to-line call.	

NOTE: Auto announce calls cannot be placed from CO lines.

3.7.8 Call Forward "FWD" (19)

The Call Forward feature enables the user to have calls directed to his phone forwarded to another phone ‘within the switch. The features are activated and deactivated under user control at any time (with some restrictions). The feature options include:

(a) Calls may be forwarded automatically, manually, or not at all.

(1) Calls may be automatically forwarded by the phone when the feature is activated.

(2) Calls may be forwarded manually by the user on a call-by-call basis.

(3) Call forwarding can be deactivated (and activated) by the user at any time.

(b) Call-forward activation is of two types.

(1) Call forwarding activation permits fixed-forwarding to a preprogrammed default destination.

(2) Call forwarding activation permits variable forwarding, where the user may modify the forwarding type and destination at each activation.

(c) Calls may be forwarded conditionally or unconditionally. Conditions (or types) apply to both fixed-variable automatic forwarding. The conditions are:

(1) Forward on busy (1)

(2) Forward on no answer (2)

(3) Forward on busy and no answer (3)

(4) Forward all calls (4)

(d) Calls may be forwarded selectively (Special Forward, SP FWD) based on the call source.

(1) Based on the source of the incoming call, the call may be selectively forwarded (forward only = 1).

(2) Based on the source of the incoming call, the call may be selectively not forwarded (forward except = 2).

Call Forward and Special Call Forward, Operation

STEP	OPERATION	RESULT
1	To activate, depress FWD or SP FWD.	(a) Associated LED lights. (b) Feature activated.
2	To deactivate, depress FWD or SP FWD a second time.	(a) LED off. (b) Feature deactivated.

3.7.9 Call Hold "HOLD" (24)

Hold permits an existing call to be temporarily placed in a hold condition and to be retrieved by the station that placed the call on hold, or by another station with an appearance for that line. Exclusive Hold allows the user to prevent another station from retrieving a held call.

The HOLD button is used to place the call on hold. Retrieval is performed via the line button for multi-line sets, or the HOLD button for single-line sets.

For the single-line set user, while one call is being held, another call can be initiated. Once the call is established the call swap (or call hold flip-flop) feature can be used to alternate between the two calls.

With multi-line sets, only the station line used to initiate the call hold can retrieve the held call. When multi-set line appearances are present on other Featurephones, other users can also retrieve the call. If desired, the hold pushbutton can be depressed a second time which implements Exclusive Hold.

STEP	OPERATION	RESULT
	Begin from a normal two-way conversation.	
1	For normal hold, depress HOLD. OR For exclusive hold, depress HOLD again within 3 seconds.	
	If hold is successful,	(a) HOLD LED flashes at the appropriate rate depending on type of hold. For multi-line sets, the primary line LED will flash at the appropriate rate. If hold is not exclusive, all other appearances of the line will flash the primary line LED.

STEP	OPERATION	RESULT
		(b) "CALL HELD" See Visual Signals, Table 3.1.
	OR	
	If hold is unsuccessful,	(a) Hold LED will return to original status. (b) Reorder tone heard. (c) "REQUEST DENIED". Line LEDs unaffected.
	User may flash to return to party or re-attempt hold.	
2	To originate a new call, depress NEW CALL, FLASH, or select a new line.	See appropriate paragraph.

See also: Primary Line
Secondary Line

(3) The Call Park capabilities are subject to the switch class-of-service restrictions.

(4) The phone's prime-line directory number will be used as the park-to and park-retrieve default number. -Only the phone whose control line a call is parked to will be notified of the retrieval or abandon of that parked call. The parking phone will not be notified unless it is also the parked-to phone.

(5) The phone will distinguish between a call park and a park retrieve by whether the user is in a dialing state or not. If the phone has just received a collect digits signal from the switch, then the request will be to retrieve a parked call. This is true even if there is a call on soft hold. This arrangement allows the user to bring a parked call into a consult or three-way connection. If the user is currently engaged in a connection with another party, then depressing the PARK button indicates the user wants to park the call.

(6) When calls are parked to a control line that is not a prime control line, the STATUS message will be sent to the phone that controls the line. The phone will keep track of which line the call is parked to and automatically retrieve the call from the respective park number on default input.

(7) If the parked call is not retrieved before time out, the parked-to station is rung.

(8) If the parked party releases from the connection, the parked-to phone will be notified and the PARK LED will be extinguished if no other calls are parked.

Call Park

STEP	OPERATION	RESULT
	Beginning from a normal two-way conversation.	
1	Depress PARK.	(a) PARK LED flashes at 120 IPM. (b) "NUMBER = "
2	Enter the number to which the call is to be parked. Repertory dialing, or DSS buttons may be used. OR Enter #, or do nothing for 5 seconds and the user's directory number is used.	"NUMBER = XXXX"

STEP	OPERATION	RESULT
3	If park is successful,	(a) Confirmation tone. (b) PARK LED on steady on parked-to phone.
4	Release by Flash or New Call.	" "
5	If park is unsuccessful,	"REQUEST DENIED"
6	Flash to return to original connection.	PARK LED off.

NOTE: If parked-to station is a Featurephone and has a PARK LED, that LED will be on to indicate that a call is parked to that station number.

Call Park Retrieve

STEP	OPERATION	RESULT
	Begin from an idle, off-hook, receiving dial tone state.	
1	Depress the PARK button.	(a) PARK LED flashes at 120 IPM. (b) "NUMBER = "
2	Enter the number from which the call is to be retrieved. OR Enter # or wait for 5 seconds and the user's own directory number becomes the park retrieve number.	"NUMBER = XXXX"
3	If retrieval is successful,	(a) "C#XXXXRET" where XXXX is call which was parked OR "AA XXXX" if trunk group indicating type and number. (b) Transmission established.
	If unsuccessful,	(c) Reorder tone heard. (d) "REQUEST DENIED" (e) PARK LED will extinguish if no other calls are parked.

NOTE: When multiple calls are parked to a single number, they will be handled on an FIFO basis.

3.7.11 Call Pickup "PICKUP" (28, 29)

Call Pickup allows the user to answer another ringing station within a work group (Group Pickup) or from a larger expanded group (Extended Group Pickup). Both types provide for directed pickup so that the desired ringing station may be specified.

Group/Directed Call Pickup (28)

STEP	OPERATION	RESULT
	Begin from detection of a phone ringing within the pickup group.	
1	Depress PICKUP.	(a) PICKUP LED on.
	For directed pickup, enter the ringing number within 5 seconds. **OR** Do nothing for 5 seconds and group pickup is activated.	
		(b) "NUMBER = " indicating the call source, the number picked up from and PI to indicate a pickup. (c) Call is connected to user's phone.
2	Proceed with the call as normal.	
3	If there was no ringing call within the group.	(a) Reorder tone heard. (b) PICKUP LED off. (c) "NO CALL"
4	If pickup attempt was to a station restricted from pickup.	(a) Reorder tone heard. (b) PICKUP LED off. (c) "CANNOT COMPLETE"

NOTE: If there is more than one ringing phone in the group, PICKUP will connect to the station listed first in the call pickup group table, unless directed pickup is used.

Extended Group/Directed Call Pickup (29)

Same as Group/Directed Call Pickup except depress GRP PICKUP.

See also: Inside Call Source Display, Table 3.5.
Outside Call Source Display, Table 3.6.

3.7.12 Call Waiting "WAIT" (30)

Call Waiting permits an instrument user to cause incoming calls of certain types to wait for answering when the desired called line is busy with another call. To the calling line, it will appear as if the called line is idle. The called instrument provides special audible and visual signals to alert the user to the waiting call. The user may either release the current call and then answer the new call, or swap back and forth between calls. Up to two calls are allowed to wait on an Integrated Featurephone line.

Call Waiting, Initiation

STEP	OPERATION	RESULT
	Begin with a call to a station, busy tone returned.	
1	To activate a tone indicating to the called party that a call is waiting, depress CALL WAIT.	(a) CALL WAIT LED on. (b) Ringback tone (as if station was not busy) heard.
2	Remain off-hook (or depress MONITOR/SPEAKERPHONE) until party answers.	
3	Continue as with a normal line-to-line call.	

Call Waiting, Busy Station

STEP	OPERATION	RESULT
	Call waiting is active immediately following button programming.	
1	Incoming call arrives and user is busy with another call.	(a) CALL WAIT LED flashes at 60 IPM. (b) Beep tone heard. (c) "C# = XXXX WAITING"
2	To answer waiting call, depress CALL WAIT.	(a) CALL WAIT LED flashes at 60 IPM indicating that the original call is now waiting. (b) Call source, "WAIT ANSWER"

STEP	OPERATION	RESULT
3	Successive operations of CALL WAIT button swaps back and forth between calls.	Call source shows that the current call is waiting.
4	When finished with a call, go on-hook.	If only one call is waiting: (a) CALL WAIT LED off. (b) Distinctive ringing indicates call still waiting. (c) Call source "C# = XXXX NORMAL" if two calls waiting, as in Step 1 above.

NOTE: If the remote end of the active call disconnects, the user must depress CALL WAIT to be connected to the other call waiting or go on-hook as in Step 4.

Special Call Wait -- Activation/Deactivation (31)

STEP	OPERATION	RESULT
1	To activate, depress SP WAIT.	SP WAIT LED on.
2	To deactivate, depress SP WAIT again.	SP WAIT LED off.

3.7.13 Conference "CONF" (38)

Conference, Eight-Party

Conference provides the ability to form up to an eight-party conference through an add-on technique using a conference bridge. The conference is assembled by a single-control party rather than a meet-me or attendant arranged conference.

STEP	OPERATION	RESULT
	Beginning from an idle state, seize a line.	
1	Depress CONF button.	(a) CONF LED flashes at 120 IPM. (b) A conference bridge is selected by the switch. (c) "CONFERENCE" (d) CONF LED on (see note 1).

STEP	OPERATION	RESULT
2	Depress FLASH.	(a) Distinctive dial tone heard (see note 2). (b) CONF LED winks to indicate conference bridge is on hold. (c) "NUMBER = ?"
3	Enter the number (normally or by any of the features) of a conferee.	(a) "NUMBER = XXXX" (b) "RINGING"
4	If the conferee answers,	CONF LED goes on.
5	Depress CONF button. If conferee does not answer, is busy, or an incorrect party is reached, go on-hook.	Switch will call user back into conference bridge.

NOTES:

1. If the conference bridge is busy, the **CONF** LED will go dark, and the display will show: FACILITY BUSY.

2. All actions from this point on may be repeated up to seven more times to establish an 8-party conference call.

See also: Line Seizure

3.7.14 Data Protection "PROT" (33)

Data Protection allows the user to protect a call connection against inadvertent intrusion. The purpose of the feature is to provide protection to ensure the integrity of data connections using a modem. Data protection may be enabled by the user for any call and is only active for the duration of that call. Subsequent data calls will require reactivation.

Data Protection

STEP	OPERATION	RESULT
	Beginning with the phone off-hook in an idle state or with a call connected.	
1	Depress PROT button.	(a) PROT LED on. (b) Protection in effect.

NOTE: Data protection is automatically disabled when the call is released.

3.7.15 Dial Display "D DISP"

Dial Display allows the user to cause the phone automatically to dial the number associated with the message appearing on the phone's display. The phone will send the digits shown on the display, ignoring any alphanumeric text characters, and will provide DTMF tone feedback. The displayed number can result from any one of a number of instrument operations, e.g., retrieval of messages left via the message leaving feature, messages due to the Scheduled Reminder feature, the speed calling numbers that appear when viewing the Speed Calling list, etc.

Dial Display - On-Hook

STEP	OPERATION	RESULT
	Begin with the phone on-hook and a message with an associated number displayed.	
1	Depress D DISP.	(a) D DISP LED on. (b) A line is automatically selected (via preference or preselection) and seized via the speakerphone or monitor. (c) LED associated with Speakerphone or Monitor on. (d) "XXXX" digits displayed as sent. (e) DTMF tone heard. (f) D DISP LED off.
2	Continue as with a normal line-to-line call.	

Dial Display - Off-Hook

STEP	OPERATION	RESULTS
	Begin with the instrument off-hook and a message with an associated number being displayed, or the "PLEASE DIAL" message being displayed. For the latter case, if the "PLEASE DIAL" message has replaced a message containing the desired number, the number is retained in storage until it is replaced by a new number to be displayed.	
1	Depress D DISP button.	(a) D DISP LED on. (b) A line is automatically selected (via preference or preselection) and seized via the speakerphone or monitor. (c) LED associated with Speakerphone or Monitor on. (d) "XXXX" digits displayed as sent. (e) DTMF tone heard.

STEP	OPERATION	RESULT
		(f) D DISP LED off.
2	Continue as with Basic Calling (Internal/External).	

See Also: Call Source Display
Preference
Preselection

3.7.16 Direct Station Selection "DSS" (11)

DSS provides for both a secretarial answering service and a repertory dialing service. For repertory dialing, the DSS button is used to make intercom calls, call announcing calls, and station-to-station calls from the originating instrument to a prime or control line on another instrument associated with the DSS button. For answering service, the DSS button functions as a DIRECTED CALL PICKUP button when the associated prime or control line is ringing, i.e., it is used to divert an incoming call from the associated line to the answering instrument.

Depression of a DSS button causes the instrument to turn on the speakerphone or monitor feature (if the instrument is on-hook), and then automatically send out over the selected line the programmed digits associated with the DSS button. The switch then attempts to complete the call to the control line associated with the DSS button.

Direct Station Selection - Answering

DSS answering allows a secretary or an attendant to provide answering service for a large number of lines. This is accomplished by providing a DSS button per line to be serviced, either on the instrument itself and/or on the DSS Add-On Module.

STEP	OPERATION	RESULT
	Begin with the answering instrument on-hook and an indication of a ringing DSS line (LED associated with DSS button is flashing at 60 IPM).	
1	To "peek" (preview) at a call prior to answering, depress the ringing DSS button.	Call source display. See Tables 3.5 and 3.6.
2	To answer the call, select an idle line.	Secondary status LED on.
3	Go off-hook by any of the methods available.	(a) Associated LEDs on.

STEP	OPERATION	RESULT
		(b) Primary status LED associated with the seized line on. (c) Dial tone heard. (d) "PLEASE DIAL"
4	Depress the ringing DSS button.	(a) Display is replaced by call source pickup display. See Tables 3.5 and 3.6.
5	Converse as usual.	

NOTES:

1. The call that was ringing the DSS line is connected to the user's answering line and is disassociated from the called line. The LED associated with the DSS button at the answering instrument is updated to reflect the new current status of the associated DSS line, e.g., the status will indicate busy.
2. When the user "peeks" at more than one ringing DSS line, the call source of the last depressed DSS button is displayed and persists until another user action occurs or until the status of the DSS line is changed to nonringing.
3. Depression of ringing DSS buttons provides no automatic line seizure capability. This restriction permits the user to "peek" at the call source display.
4. If DSS answering is denied because there is no ringing phone, an error tone is heard and NO CALL is displayed.
5. If station is restricted from picking up ringing line, REQUEST DENIED is displayed.

Direct Station Selection - Origination

STEP	OPERATION	RESULT
	Begin with the instrument indicating a preference line choice (secondary status LED is lit for the associated line) and the instrument in an off-hook condition with handset on-hook.	
1	Depress the desired DSS button.	(a) DSS LED on. (b) Speakerphone or monitor and associated LED will be activated. (c) Preference line seized. (d) Primary and secondary status LEDs on.

STEP	OPERATION	RESULT
		(e) Digits associated with the DSS pushbutton will be sent to the switch and are simultaneously displayed.

NOTES:

1. The selected DSS station must be either idle (LED off) or busy (LED on). If the station is being signaled (LED flashing), the DSS pushbutton should not be depressed except to answer the call.

2. Unlike a repertory dial button, alphanumeric characters are not stored for a DSS button. Therefore, only the digits associated with the DSS button will be displayed when they are sent to the switch.

2	Proceed as with a normal station-to-station call.	

See also: Preference

3.7.17 Dual Access "DUAL ACCESS" (10)

Dual Access permits a user to perform two features with one button, e.g., activate Call Hold/cancel Call Hold.

STEP	OPERATION	RESULT
1	Depress DUAL ACCESS button.	(a) DUAL ACCESS LED on. (b) Confirmation tone heard. (c) Feature 1 activated (see note).
2	To activate second feature, depress DUAL ACCESS.	(a) Confirmation tone heard. (b) Feature 2 activated. (c) DUAL ACCESS LED off (see note).

NOTE: If the Dual Access button was previously programmed, the repertory dial number previously programmed will appear in the display.

3.7.18 Elapsed Time "TIMER" (14)

Elapsed Time allows the user to measure and display the duration of both incoming and outgoing calls. Additionally, a given duration may be programmed which, upon expiration, causes the instrument to produce an audible and visual alarm, along with a duration alarm display.

Elapsed Time (ET) - Outgoing Calls

STEP	OPERATION	RESULT
	Begin with the phone on-hook.	
1	Go off-hook and seize a line.	(a) Associated line LED on. (b) The phone's internal elapsed timer is reset and started.
2	To read ET, depress TIMER any time after going off-hook.	(a) TIMER LED on for 1 second. (b) Elapsed time displayed for 2 seconds "HH:MM:SS".
3	To reset ET, depress TIMER a second time.	(a) TIMER LED on 1 additional second. (b) "00:00:00" ET reset. (c) Updated and running ET displayed "HH:MM:SS" until another user action causes a change in status.
4	To include the called party on a 3-way conference, place the call on soft hold (depress HOLD once).	Timer continues accumulating time regardless of flashes, etc., until call termination.
NOTE:	If call is placed on hard hold (depressing HOLD twice) and user causes an on-hook condition, the timer is stopped. Depression of TIMER will recall ET of previous call.	
5	Upon termination of call.	(a) TIMER LED on. (b) Elapsed time displayed "HH:MM:SS"
6	Depress TIMER.	Display and LED off.

NOTE: When the station is idle, a depression of TIMER will display the total elapsed time of the last call. A second depression will turn off the LED and clear the display.

See also: Call Hold
Conference

Elapsed Time (ET) - Incoming Calls

STEP	OPERATION	RESULT
	Begin with the acceptance of an incoming call.	
1	Any time after answering the incoming call, depress TIMER.	(a) TIMER LED on for 1 second. (b) ET from previous call displayed for 2 seconds "HH:MM:SS".

STEP	OPERATION	RESULT
2	To reset the timer, depress TIMER again within the l-second period.	(a) TIMER LED on for an additional 1 second. (b) Display updated and current ET displayed for 2 seconds "HH:MM:SS".
3	Continue as with Step 4 above.	

Duration Alarm

STEP	OPERATION	RESULT
1	Depress and hold TIMER.	(a) Beep heard. (b) "TIMER/ALARM"
2	Release.	ALARM = 00:00:00"
3	Enter HH, #, MM, #, SS.	"ALARM = HH:MM:SS"
4	Depress and hold TIMER.	(a) Beep heard. (b) "TIME SET"
5	Release.	Display clears or reverts to the time and date.
6	Upon expiration,	(a) TIMER LED winks. (b) Continuous beep heard. (c) "ALARM EXPIRED"
7	To turn off alarm, depress TIMER.	(a) TIMER LED off. (b) Beep off if not timed out. (c) Display clears or reverts to the time and date.

Timer/Elapsed Time Notes:

1. The elapsed time counter accumulates up to 23 hours, 59 minutes, and 59 seconds. Elapsed times greater than this length wrap around and continue accumulating starting from 00 hours, 00 minutes, and 00 seconds.

2. Only one call at a time may be accumulating elapsed time, i.e., once a call is initiated on another line, the timer will be reset and started for the new call; calls answered on another line cause the timer to stop.

3. The duration alarm timer allows timing duration up to 23 hours, 59 minutes, and 59 seconds.

4. The duration alarm may only be canceled setting the alarm duration to 00 hours, by 00 minutes, and 00 seconds.

5. The display will show "ALARM EXPIRED" and the TIMER LED will wink.

3.7.19 Executive Priority Call "PRIOR" (35)

Executive Priority Call enables a properly classmarked Integrated Featurephone to attempt a call that cannot be forwarded and will override Forced Busy at the destination. The called party will receive distinctive Intercom ringing. If the called party is busy and has Call Waiting, Call Waiting will apply, otherwise a busy signal will be returned.

STEP	OPERATION	RESULT
1	Go off-hook.	(a) LINE LED on. (b) "PLEASE DIAL"
2	Depress PRIOR.	"NUMBER = "
3	Enter desired number.	(a) NUMBER = XXX&" (b) "RINGING"
4	Continue as in a normal line-to-line call.	

NOTES:

1. The party invoking Executive Priority Call must be classmarked for Busy Override. Both features are controlled by the same classmark.
2. The Executive Priority call may not be forwarded.
3. The Executive Priority feature activation will not affect the current call origination other than as described above.
4. The Executive Priority Call will not be seen by other lines as ringing.

3.7.20 Extended Group Select (12)

Extended Group Select allows a DSS operator to access an extended group.

STEP	OPERATION	RESULT
1	To activate, depress EXT.GRP.SEL.	(a) EXT.GRP.SEL. LED on. (b) Extended Group activated.

STEP	OPERATION	RESULT
2	To deactivate, depress EXT.GRP.SEL. again.	(a) EXT.GRP.SEL. LED off. (b) EXTENDED GROUP deactivated.

3.7.21 Flash "FLASH" (03)

Flash permits the user to interrupt the switch to get dial tone and/or to operate the transfer, consult, and three-way features by using only one feature button, i.e., the FLASH feature button. The user may choose to use the hookswitch to generate a manual flash to obtain the same results as if the FLASH button were depressed.

Consult via Flash

STEP	OPERATION	RESULT
	Begin with the phone off-hook and engaged in a normal two-party call. The STATUS and I-USE LEDs are on.	
1	Depress FLASH button.	(a) FLASH LED flashes at 60 IPM. (b) The switch places the other party on hold (excluded from communications). (c) Dial tone heard. (d) "PLEASE DIAL"
2	Enter the number of the party to be consulted using any of the various methods available.	(a) "XXXX" as entered. (b) Ringback tone heard.
3	If called station is unavailable (for whatever reason), depress FLASH again.	(a) FLASH LED off. (b) Original party reconnected. (c) "HOLD RETRIEVAL"
4	If the station is answered.	(a) FLASH LED flashing. (b) Transmission established. (c) Original call on soft hold.
5	Consultation may be made and the user now may choose to: (a) Establish a 3-way call (b) Transfer the party on hold to the consulted party (c) Return to the party on hold	

See Three-Way Call via Flash and Transfer via Flash for procedures.

Three-Way Call via Flash

STEP	OPERATION	RESULT
	Begin as in Step 4 of Consult via Flash.	
1	To establish a three-way call, depress FLASH.	(a) FLASH LED on. (b) Three-way connection established.
2	To release consulted party, depress FLASH.	(a) FLASH LED off. (b) Two-party call resumes.
3	To release from both parties, go on-hook.	(a) FLASH LED off. (b) Associated LINE LED extinguished. (c) Two remaining parties connected.

Transfer via Flash

STEP	OPERATION	RESULT
	Begin with a connected call.	
1	To transfer consulting party to party on soft hold, go on-hook.	(a) FLASH LED off. (b) Associated LINE LED extinguished. (c) Two remaining parties connected.

NOTES:

1. Any time the FLASH feature button is depressed and the phone is off-hook, the phone will send a FLASH response to the switch.

2. The instrument will time an on-hook followed by an off-hook transition and send a FLASH response to the switch if the transition interval is within the generic hookswitch flash interval.

3. The user may also effect a transfer of the party on soft hold to the called consulted party if he releases while the called party (party to be consulted) is being rung. The party on hold is then connected to the called party and hears the ringback tone. When the consulted party answers, the consulted party is connected to the original party on hold. The call source display indicates the call was transferred and the station number of the transferred-from station.

4. The FLASH feature button (or hookswitch flash) is also used in conjunction with the adding of parties to the multi-party conference bridge. Refer to CONFERENCE for more information.

5. If the user's set is equipped with a CONSULT/TRANSFER button and/or a three-way button, the user may intermix the FLASH button with the other buttons to perform the desired operation. The LED associated with the last button used reflects the status of the latest condition. For example, instead of the user depressing the FLASH button to establish a three-way call, the user could have depressed the three-way button. The THREE-WAY LED turns on steady and the FLASH LED is extinguished for this case.

6. If the user uses the hookswitch to generate a FLASH response, the FLASH LED shall be updated to reflect the latest status as if the FLASH button itself had been used.

3.7.22 Forced Busy "MK BUSY" (36)

Activation of Forced Busy will cause all calls to the phone to be directed to busy tone. Calls incoming to non-control lines and DSS lines' appearances on the instrument are not affected by this feature.

STEP	OPERATION	RESULT
	Begin from any state.	
1	To initiate forced busy, depress FORCED BUSY.	(a) FORCED BUSY LED on. (b) Forced busy initiated.
2	To deactivate, depress FORCED BUSY a second time.	(a) FORCED BUSY LED off. (b) Forced busy deactivated.

NOTES:

1. Executive priority calls override forced busy when the desired line is truly idle. Call announcing operates regardless of forced busy state.

2. If the forced busy feature is active, call waiting will not be allowed.

3.7.23 Intercom Call "ICOM" (22)

Intercom Call provides the ability to make calls within a restricted group of phones and receive priority call termination handling. Distinctive ringing provides the terminal user with the knowledge that the call is distinct from a normal line-to-line call and comes from within a restricted group.

The Intercom Call differs from a normal call in several aspects:

(a) The call cannot be forwarded.

(b) The call cannot be diverted with call pickup.

(c) Distinctive intercom ringing is supplied at the called phone.

(d) The call origination can be completed even though the called line is busy (pseudo call waiting).

(e) The call will automatically release privacy on a current line call.

Intercom Call - Origination

STEP	OPERATION	RESULT
	Begin from off-hook talking state or off-hook receiving dial tone and "PLEASE DIAL".	
1	Depress I COM, if off-hook receiving dial tone.	(a) I COM LED on. (b) Dial tone heard. (c) "WHAT NUMBER"
2	Enter destination via a DSS button number and proceed as with a normal line-to-line call.	(a) I COM LED same. (b) "XXXX" as programmed in DSS.
3	If off-hook talking.	(a) Current call is placed on hard hold and privacy is released. (b) I COM LED flashes at 120 IPM. (c) Dial tone heard. (d) "WHAT NUMBER"
4	Depress a DSS button to call the desired number.	(a) I COM LED on. (b) "XXXX" as programmed in DSS.
5	Proceed as with a normal line-to-line call.	

NOTES:

1. Keypad or repertory dialing is not permitted, and attempts to use them will cause an error tone.
2. Intercom calls cannot be placed from CO lines.

Intercom Call - Termination, Multi-Line Set

STEP	OPERATION	RESULT
1	Depress LINE button.	(a) LINE LED flashes. (b) Distinctive ringing heard.
2	Continue as with a normal line-to-line call.	

NOTE: If the control line is busy, and there is a call waiting button, the call waiting LED will flash, a distinctive call waiting beep will be heard, and the display will show the information for the waiting call. The intercom call will be treated as **a** waiting call.

intercom Call - Termination, Single-Line Set

STEP	OPERATION	RESULT
1	Go off-hook.	Distinctive ringing heard.
2	Continue as with a normal line-to-line call.	

NOTE: If your phone is busy, and you do not have **a** call waiting button, you will not receive the intercom call. The caller will receive a busy tone.

Automatic Intercom "A ICOM"

The AUTOMATIC INTERCOM button combines the INTERCOM button with the dialing capability of the DSS button. Pressing an AUTOMATIC INTERCOM button originates an intercom call to a single specific party whose identity has been programmed in the button.

The call origination sequence is the same as for the INTERCOM feature except that pressing a destination button (DSS or LINE) is not required.

The AUTOMATIC INTERCOM button is not used for call termination.

NOTES:

1. Instruments programmed for AUTO INTERCOM must be members of an Intercom group (data base configuration).
2. Intercom calls are not forwarded if the called line is busy or does not answer.
3. Intercom calls will appear only at the destination instrument and not at any associated instruments in the group.
4. Intercom calls may not be answered using CALL PICKUP.
5. There may be only one INTERCOM button on a phone, although there may be multiple AUTO INTERCOM buttons.
6. An intercom call placed to a POTS phone will proceed as a normal POTS termination.
7. AUTO INTERCOM calls cannot be placed from CO lines.

3.7.24 Line Buttons (Multi-Line Sets Only)

Line Buttons are required on a multi-line set. Each line button has two LEDs associated with it: the PRIMARY status indicator, or UPPER LED; and the SECONDARY status indicator, or LOWER LED. The primary LED shows the IDLE, RINGING, HOLD, EXCLUSIVE HOLD, and BUSY status of the line. Prior to going off-hook, the PREFERENCE LED will show the I-USE condition, if applicable.

Line buttons, once programmed, utilize both LEDs to provide status information for the line. This information represents the state of the line on the user instrument as well as on other instruments (multi-set lines). Status and I-USE LEDs are shown in Table 3.13.

I-USE LEDS

The lower LED is called the I-USE LED. It represents the preference selection status of the line at each particular instrument. Refer to Table 3.14 and Figure 3.3.

3.7.25 Line Preference

Preference, when enabled, will partially or completely eliminate the need for a user to select a line manually. The choice is automatically made by the instrument according to the rules preprogrammed by the instrument user. The user may choose to enable none, one or more, or all of the preference features. The five different kinds of preference, and associated notes are listed in Programming Control Parameters, Preference paragraph 3.4.5.

Preference Line Choice

STEP	OPERATION	RESULT
	Begin with preference feature programmed and phone on-hook. Secondary status LED is on to indicate preference line choice.	
1	Go off-hook via handset, speaker-phone or monitor.	Preference line choice is connected.
2	If line choice was due to an incoming call, answer the call.	(a) Primary and secondary line LEDs on. (b) "XXXX" call source.
3	If line choice was for placing a call,	(a) Primary and secondary line LEDs on. (b) Dial tone heard. (c) "PLEASE DIAL"
4	Continue as for a normal line-to-line call.	

Table 3.13 Status LEDs

The upper LED is considered the Status LED and presents the following indications:

LED CONDITION	STATUS
STEADY LED	The line is busy, either at this instrument, or at another phone for multi-set lines.
WINKING LED	The line is ringing, and the alerting type is delayed.
FLASHING LED	The line is ringing, and the alerting type is immediate.
HOLDING LED	The line has a call on non-Exclusive Hold, i.e., the held call may be retrieved from any instrument at which this multi-set line appears.
EXCLUSIVE HOLD LED	The line has a call on Exclusive Hold, i.e., this held call may be retrieved only from this instrument.

See also: Multi-line Set
Multi-set Line
Visual Signals, Table 3.1

Table 3.14 I-USE LEDs

PREFERENCE CHOICE	I-USE LED
Prime	I-USE LED on Prime Control line on steady to indicate that this is the line that will be automatically selected when going off-hook.
Last Line Used	I-USE LED on the last line used on steady to indicate that this is the line that will be automatically selected when going off-hook.
Idle	I-USE LED on any idle line on steady to indicate that this is the line that will be automatically selected when going off-hook.
Incoming Ringing or Non Ringing	I-USE LED on when receiving a call to that particular line. On steady indicates that this is the line that will be automatically selected when going off-hook.

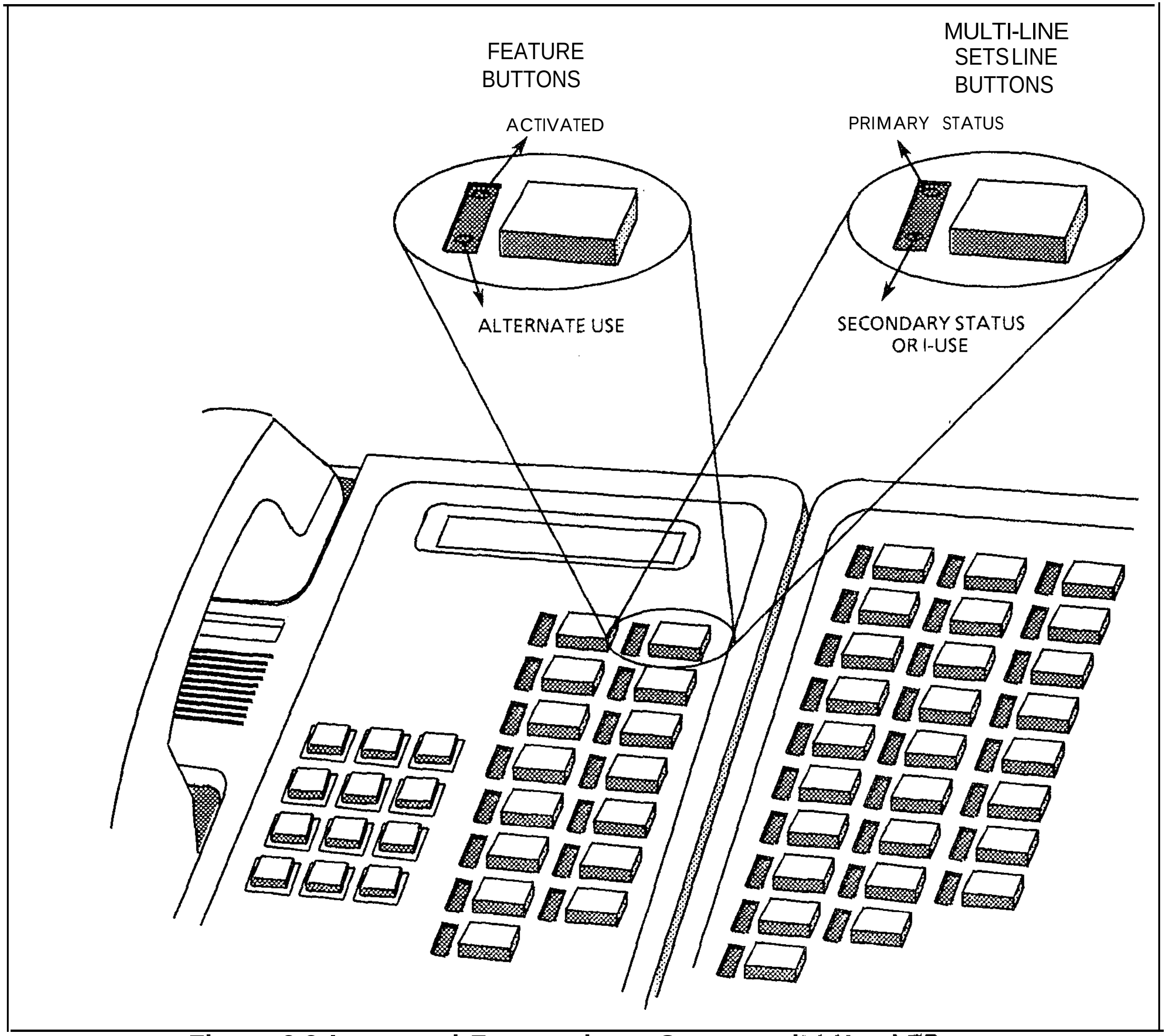


Figure 3.3 Integrated Featurephone Status and I-Use LEDs

NOTES:

1. With the exception noted in note 2 below, preference features shall apply to all PABX or CO LINE buttons (prime, control, and non-control lines). DSS buttons are not considered as line buttons.

2. Ringing line preference and incoming call preference shall also apply to the INTERCOM button on those instruments so equipped.

3. If preference (or manual preselection) chooses a ringing line, the call source shall be displayed during the alerting interval. This display is the basis for the user to determine how to handle the call.

4. Secondary line appearances are to be treated as independent lines with regard to the preselection and/or preference features.

3.7.26 Line Seizure

There are various ways that an instrument user may seize a line, i.e., originate or answer a call by presenting an off-hook condition to any of the virtual lines connecting the instrument to the serving switch. Basically, the user may manually generate the off-hook condition, or the instrument may automatically generate the off-hook condition due to some user action. Both the manual and automatic line seizure methods are given.

(a) Manual Line Seizure

Manual line seizure results from any of the following user actions:

(1) Handset going off-hook.

(2) Depressing the SPEAKERPHONE feature button with handset on-hook.

(3) Depressing the MONITOR feature button with handset on-hook.

(4) Depressing the NEW CALL feature button with the handset off-hook and the line in a busy condition (presenting an off-hook condition to the switch); for this case the current call will be released before reseizure occurs.

(5) Depressing the NEW CALL feature button with the handset off-hook and the line in a released condition (indicated by the RELEASE button's associated LED being on).

(6) Depressing the RELEASE feature button with the handset off-hook and the line in a released condition (indicated by the RELEASE button's associated LED being on).

NOTES:

1. For 1 through 3 above, a particular line must be selected (manually or automatically) either prior to or after the user action in order for a line seizure to occur. For 4 through 6, the line currently selected (indicated by the associated LINE button's I-USE LED being on) is the line that will be reseized. Refer to line selection for information on how lines are manually and automatically selected.

2. Selection of the one line on a single-line set is done automatically by the phone for all of the above cases.

(b) Manual Line Seizure - Handset On-Hook

STEP	OPERATION	RESULT
	Begin from an idle on-hook state, handset in cradle.	
1	Select an idle line (multi-line sets only).	Associated I-USE LED on.
	OR	
	Use the line selected automatically by the preference feature.	Indicated by the associated I-USE LED on.
2	Either (a) Lift the handset (b) Depress MONITOR OR (c) Depress SPEAKERPHONE	(a) MONITOR LED on . OR (b) SPEAKERPHONE LED on (c) "PLEASE DIAL" (d) Primary line LED on.
3	Proceed as with a normal line-to-line call.	

NOTES:

1. If the line selected is a non-dial line, i.e., a personal CO line, the prompt is not displayed. Instead, CO dial tone will be heard as confirmation of successful CO seizure.

2. In the event a line (for multi-line sets) has not been selected (automatically or manually) in Step 1 and the user performs Step 2, then the message "WHICH LINE" will be displayed to prompt the user to select a line.

(c) **Manual Line Seizure - Handset Off-Hook/Line Released**

STEP	OPERATION	RESULT
	Begin with handset off-hook and line previously released by RLS button.	(a) Associated I-USE LED on. (b) RLS LED on.
1	Depress either NEW CALL or RLS button.	(a) Associated LED on. (b) Line is seized. (c) Dial tone heard. (d) "PLEASE DIAL" (e) NEW CALL or RLS LED off.
2	Proceed as with a normal line-to-line call.	

(d) **Manual Line Seizure - Handset Off-Hook/Line Busy**

STEP	OPERATION	RESULT
	Begin with handset off-hook and the selected line busy with a call.	Associated I-USE and BUSY LEDs on.
1	Depress NEW CALL.	(a) NEW CALL LED on. (b) Line released and reseized. (c) Dial tone heard. (d) "PLEASE DIAL" (e) NEW CALL LED off.
2	Proceed as with a normal line-to-line call.	

(b) Automatic Line Seizure

See Automatic Line Seizure, Paragraph 3.7.3.

3.7.27 Line Selection

This feature describes the various ways that a phone (multi-line set) user may select a line for seizing and originating, answering, removing a call from hold, or displaying the source of an incoming call. Basically, the user may manually select the desired line or may choose the line that has been automatically selected by any of the preference features.

(a) **Manual Line Selection**

There are two kinds of manual line selection: preselection and manual selection.

Preselection is defined as the manual selection of a line when the user depresses the LINE button while the instrument is in an on-hook condition, i.e., the handset is on-cradle, and the speakerphone and monitor features are idle. Preselection overrides any previous preselected line or any line choice made by any of the preference features which may be in effect. The instrument guarantees the preselection choice for at least 10 seconds before allowing any of the preference features, if active, to override the choice. Otherwise, the preselection choice remains indefinitely or until the user selects another line.

Manual selection is defined as the manual selection of a line when the user depresses the LINE button while the instrument is in an off-hook condition, i.e., either the handset is off-cradle or the speakerphone or monitor features have been activated. Two other conditions are possible when the user depresses the LINE button:

a. If a line is currently seized, that line will be released and the new line will be seized.

b. If a line is not currently seized, the selected line will be seized. When the user returns to an on-hook condition, the line now behaves as a preselection choice, i.e., the line continues to be the chosen line for at least 10 seconds if any preference features are in effect. Thereafter, the choice may be overridden by any possible line choice conforming to the rules established by the preference features. Otherwise, the line remains the chosen line indefinitely or until the user selects another line.

(b) **Manual Line Selection - Preselection**

STEP	OPERATION	RESULT
	Begin with the phone in an on-hook state and neither speakerphone or monitor active.	
1	Depress desired line button.	(a) Associated I-USE LED on. Any other I-USE LED on by a previous preselection or by any active preference feature will be off. (b) "PLEASE DIAL"
	If no preference features are activated.	(c) The above choice remains in effect and I-USE LED on until the user makes another choice.
	If preference features are in effect.	(d) The preselection choice is guaranteed for 20 seconds along with associated LED being on steady.

STEP	OPERATION	RESULT
	After 10 seconds,	(e) The preselection choice may be overridden by any possible combination programmed for preference.

NOTES:

1. Once a line seizure has taken place, preselection has no meaning until the user returns to an on-hook condition.
2. After going on-hook, the line shall remain as the selected line for 10 seconds, allowing a new call to take place without changing lines. The user may override the choice by selecting another line. The I-USE LED remains on for as long as the line is chosen.
3. After the 10 seconds have expired, preference, if in effect, may override the current line choice by any possible line choice conforming to the rules established for the preference features. The I-USE LED remains on for as long as the line is chosen.

(c) Manual Line Selection - Manual Selection

STEP	OPERATION	RESULT
	Begin with the phone in the off-hook position with the currently selected line busy with another call.	Associated I-USE LED on.
1	Depress another LINE button.	(a) Previous line disconnected and LED off. (b) New line selected and seized. (c) Associated LINE LED on. (d) “PLEASE DIAL”
2	Proceed as with a normal line-to-line call.	

NOTES:

1. Manual line selection does not apply to single-line sets because the off-hook condition is automatically related to the only line on the instrument.
2. Manual line selection must be employed before each and every call on multi-line sets for which all preference features are disabled.

3. If the user chooses a ringing line via preselection, a call source display shall be displayed during the alerting interval. This display is the basis for the user to determine how to handle the call.

4. Anytime the user depresses a LINE button while on another line, the other line is released and the line associated with the depressed LINE button is seized, if idle.

3.7.28 Line Types

Several types of lines may appear on the Featurephone. These are: single-set line, multi-set line, secondary line, CO line, and PABX line.

(a) Single-Set Line

A single-set line is a line having a line appearance on only one Featurephone instrument. The appearance may be on a single-set line, in which case the line must be a PABX line (see section on PABX Line); or the appearance may be as a line button on a multi-line set, in which case the line may be a PABX line or a CO line (see section on CO line).

Single-set lines have no special requirements in and of themselves. Therefore, no step-by-step operations are described for these lines. Refer to the section on A Basic Call and associated subsections.

NOTE: Although a single-set line may appear as a line on only one instrument, it may have secondary lines on that same instrument (provided the instrument is a multi-line set).

(b) Multi-Set Line

A multi-set line is one which appears on more than one multi-line Integrated Featurephone. This line arrangement allows the switch to emulate the basic operation of lines with multiple appearances as in a key system.

Up to 16 Featurephone lines may appear as line and/or DSS appearances on up to eight Featurephones (single-line set or multi-line set). The rest of the Featurephone lines may appear on up to four Featurephones. Status changes for any multi-set line have an appearance in order to keep the instruments and instrument users current as to line status.

A multi-set line may be configured as a PABX line or a CO line (see later sections). POTS lines can have two DSS appearances.

(c) Secondary Line

The secondary line arrangement offers a means of terminating multiple call attempts to a single PABX directory number. This is accomplished by using a multi-line set, and assigning the prime line or a control line and each secondary appearance to individual line buttons on the set.

A call to an instrument with secondary lines is alerted on one of the secondary lines when the prime line is busy. There is no need to provide call waiting operation; terminations occur to idle appearances as long as any are available, and the line is treated as busy when all line appearances are busy. Should alerting occur while the user is in conversation on a line, the active call may be placed on hold and the ringing line appearance answered.

A call may be originated from any idle line appearance.

(d) Secondary Line Operation

The user places an initial call from an idle instrument by going off-hook on a selected line.

Once in conversation, the user may access customary PABX features in handling his call, including use of consult, transfer, hold, etc. The user may in fact place a call on hold, and then place an entirely different call from another appearance of his line.

NOTES:

1. A prime or control line may only have secondary lines if the prime or control line is a single-set line.

2. A prime or control line may have up to a maximum of three secondary lines (prime/control line + three secondary lines = a total of four buttons for the line).

See also: A Basic Call
Line Selection
Prime Line
Secondary Line

(e) Central Office (CO) Line

The Featurephone provides a means for the user to access a dedicated Central Office line. The feature operates to provide capabilities similar to those for CO lines on key systems. Seizure of a CO line causes an immediate connection to the associated CO line, with dial tone being returned from the CO. Ringing of the CO line causes immediate connection to and signal of the Featurephone CO line. This operation emulates CO line operation on standard key systems, differing only on the basis of guaranteed access.

The user of a Featurephone CO line will be permitted to access PABX features via hookswitch flash or feature buttons. CO calls may therefore be placed on hold or transferred to other PABX stations. Flashes will not be echoed into the Central Office; therefore no special features can be provided for in the CO.

CO lines can be assigned only on multi-line sets. The appearance of a CO line need not, however, be restricted to a single instrument. As with PABX lines, CO lines can be configured in multi-set arrangements, so that a particular CO line can be accessed from more than one station instrument. When the line is busy, other attempts to use the line by the sharing stations will result in a trunk busy tone being received at these stations.

(f) CO Line Operation - Call Origination

1. The user selects a CO line by depressing the CO LINE feature button on the Integrated Featurephone instrument and then going off-hook.

2. Assuming the associated CO line is idle, the system establishes a network path between the Integrated Featurephone instrument and the CO line causing seizure.

3. The Central Office returns dial tone to the Integrated Featurephone user. The user may now dial as he chooses within the public network.

4. If the associated CO line is busy, the trunk busy tone will be returned.

(g) CO Line Operation - Call Termination

1. A call from the public network terminating on an Integrated Featurephone dedicated CO line causes ringback tone to the originator and distinctive ringing on the integrated Featurephone.

2. Called party answer causes CO ring-trip and an audio connection to be established.

3. If an incoming CO line call is not accepted by the Integrated Featurephone or accepted but not answered, ringback tone continues to the originator. If the terminating phone is busy, busy tone is returned. In both cases, tones are supplied by the CO.

(h) PABXLine

A PABX line has a one-to-one correlation to the directory number dialed to reach that line. PABX lines may have a single appearance on station apparatus or may have multiple appearances.

(i) PABX Line Operation

PABX lines as a class introduce no additional requirements upon the system. Therefore, no feature operations are described for these lines. Refer to A Basic Call and other related sections.

3.7.29 Manual Signaling "BUZZ" (39)

Manual Signaling (BUZZ) permits the user of one Integrated Featurephone to signal the user of another Integrated Featurephone with a buzz tone without originating a call and regardless of the call related state of the phones.

STEP	OPERATION	RESULT
	Begin from any state.	
1	Depress BUZZ.	(a) BUZZ LED on for 1/2 second. (b) BUZZ tone sounded at destination for 1/4 second. If busy, BUZZ is muted. (Operation may be repeated).

3.7.30 Message Leaving "LV MSG" (25)

LEAVE MESSAGE enables the user to leave a message at an Integrated Featurephone having a LEAVE MESSAGE button programmed. The phone leaving the message may be either an Integrated Featurephone or a POTS phone.

Visual indication of a stored message will be provided on the called instrument to alert the user. Up to eight messages may be stored.

(a) **Message Leaving - Integrated Featurephone Origination**

STEQ	OPERATION	RESULT
	Begin from a busy or no-answer call attempt to an Integrated Featurephone.	
1	To leave message depress LV MSG. Use * to scroll through menu.	(a) LV MSG LED will flash at 120 IPM. (b) "MESSAGE = ?" 1 = PLS CALL XXXX 2 = XXXX CALLED 3 = XXXX SEE ME 4 = XXXX WILL CALL 5 = XXXX URGENT 6 = XXXX RETURNED
2	Enter a message code number representing the message to be left (i.e., "1 = PLS CALL XXXX", and depress #.	(a) LED flashes at 60 IPM. (b) Confirmation tone heard. (c) "MESSAGE = 1*"
	If no message is left within five seconds or "#" is entered,	"1 = PLS CALL" (default message)

STEP	OPERATION	RESULT
	If message leaving is unsuccessful,	(a) Reorder tone heard. (b) "NO MESSAGE LEFT"

NOTES:

1. If the call has been diverted to another station (i.e., call forwarding, call pickup, etc.), the message will be left at the original called instrument.
2. The above procedure may also be performed by either party while the call is in an answered state. However, when involved in a stable two-way conversation, the Integrated Featurephone leaving the message (only Integrated Featurephones can initiate MSG LEAVE while in a two-way state) will only receive the display showing that the message was left or not left; no tones will be placed into the two-way conversation.
3. Message leaving cannot be activated during any calls that contain a third party, nor should the feature be used on two-party calls that have at one time involved a third party. Use of the feature under this condition could result in a misrouted message.
4. Message leaving destination must be an Integrated Featurephone.

(b) Message Leaving - POTS Phone Origination

STEP	OPERATION	RESULT
	Begin from on-hook condition.	
1	Go off-hook, dial the message leaving access code, the number representing the message to be left, and the directory number of the party to receive the message.	(a) If successful, confirmation tone heard, (b) If unsuccessful, reorder tone heard.
2	Go on-hook.	
	OR	
	Begin from off-hook condition.	
3	Flash.	Dial tone heard.

STEP	OPERATION	RESULT
4	Dial the message leaving access code, the number representing the message to be left, and the directory number of the party to receive the message.	(a) If successful, confirmation tone heard. (b) If unsuccessful, reorder tone heard.

(c) Message Retrieval - Integrated Featurephone Only

STEP	OPERATION	RESULT
	Leave MSG button flashing.	
1	Depress LEAVE MSG.	Oldest message first displayed.
2	To see time and date of message, depress #.	TOD displayed.
3	To scroll to next message, depress #.	Second message left is displayed.
4	To delete message and display next one, depress and hold # until beep is heard or depress LEAVE MSG.	(See note.)

NOTE: If there are more messages, LEAVE MSG LED will continue to flash until all messages have been cancelled.

(d) Message Waiting - Retrieval

STEP	OPERATION	RESULT
	Begin from an idle station either on- or off-hook with MS CTR LED flashing at 60 IPM.	
	Depress MS CTR button.	(a) Line seized. (b) Associated LEDs on. (c) "Message Center Number" (d) Connection established.
	Listen to recorded announcement or converse with attendant.	MS CTR LED off.

NOTES:

1. The MESSAGE WAITING (MS CTR) LED on the instrument that is the prime or control appearance for a directory number will be lit for a message waiting request.

2. If the instrument does not have a MESSAGE WAITING (MS CTR) LED, the message will be ignored.

3.9.31 Monitor **"MONTR"** (04)

The MONITOR button controls the activation/deactivation of the monitor amplifier. The associated LED indicates the monitor is on.

The first depression of the MONITOR feature button will initiate the monitor function in an on-hook or off-hook condition. In an on-hook condition, the monitor function will be on and the selected line seized for on-hook dialing. In an off-hook condition, the monitor function will be on standby in preparation for the monitor function which occurs when the handset is placed on-hook.

The monitor function is turned off by a second depression of the MONITOR, or depression of RELEASE, or SPEAKERPHONE feature buttons, or by removing the handset from the cradle. If the MONITOR button is depressed while in the on-hook condition, the line seizure will be withdrawn and the call abandoned.

There is interaction between the optional Speakerphone and the MONITOR. Refer to the Speakerphone paragraph for details.

STEP	OPERATION	RESULT
1	To dial on-hook, select a line and depress MONITOR.	(a) Associated LINE LED on. (b) MONITOR LED on. (c) "PLEASE DIAL"
2	Enter desired number or depress REP DIAL.	
3	Come off-hook when party answers.	
4	To listen on hold, depress MONITOR.	(a) MONITOR LED on. (b) "PLEASE HANG UP"
5	Go on-hook, listen to call on hold.	"MONITOR ON"
6	To retrieve, go off-hook.	MONITOR LED off.
	OR	
	To abandon call, depress MONITOR.	(a) MONITOR LED off. (b) Call released.

NOTES:

1. On a multi-line set, whenever the instrument is off-hook or when the speakerphone or monitor are on and the instrument initiates ringing, the ringing shall be abbreviated to present only three cycles of ringing to the instrument user. Although the ringing is stopped, the visual alerting on the line button will continue until answered.

2. The message "PLEASE HANG UP" is displayed when the user turns the monitor (or speakerphone) feature on and the handset is off-hook. The message persists until the user places the handset on-hook.

3. The monitor volume control is the slide control lever on the front of the instrument.

3.7.32 New Call "N CALL" (05)

The NEW CALL button emulates an on-hook signal followed by an off-hook signal and results in disconnect reseizure and dial tone. If depressed while on-hook, an off-hook signal is generated and the monitor or speakerphone is activated yielding dial tone via the speaker.

STEP	OPERATION	RESULT
1	To release a call without going on-hook, depress NEW CALL.	(a) NEW CALL LED on for 1/2 second. (b) Line reseized. (c) Dial tone heard. (d) "PLEASE DIAL"
2	Proceed as with a normal line-to-line call.	

NOTE: When the user of a multi-line set depresses the NEW CALL feature button after having placed a call on hard hold the ON-HOOK signal is generated for the line on hard hold. The off-hook signal is withheld until the user manually selects another line on which the new call will be placed. The users will be prompted to select a line with the message "WHICH LINE".

3.7.33 PABX Features

(a) Code Calling

Code Calling allows an authorized user to access the PABX code calling functions. This involves audible signals in one or more areas.

STEP	OPERATION	RESULT

Begin from an idle off-hook state receiving dial tone and "PLEASE DIAL".

STEP	OPERATION	RESULT
1	Enter, via keypad or a repertory dial button, the code calling access code, zone code, and code number to be signaled.	(a) "NUMBER ENTERED" (b) Code number is signaled over audible code calling system.
2	Continue as with Basic Calling (Internal/External).	

(b) **Dictation Access**

Dictation Access feature allows users to access the centrally located dictation equipment.

STEP	OPERATION	RESULT
	Begin from an idle off-hook state receiving dial tone and "PLEASE DIAL".	
1	Enter the access code for dictation access via keypad or repertory dial button.	(a) "XXXX" dictation access code. (b) Ready tone heard.
2	Enter, via the keypad, the necessary codes for start, stop, rewind and playback.	Display depends on code entered.
3	To release, go on-hook.	Display clears, connection released.

(c) **Hunt Groups**

Station Hunting allows for the routing of a call to the first idle station of a prearranged group of stations (a Hunt Group). Integrated Featurephones may be used in either terminal or circular hunt groups and operate basically the same as POTS instruments do in Hunt Groups.

Integrated Featurephones do require that the following situations be considered:

(a) Programming of call waiting or secondary line arrangements (e.g., multiple calls) may be active on a given instrument at any given time.

(b) The Force Busy feature can be used to remove a Featurephone temporarily from a Hunt Group. This is analogous to activation of call forwarding to remove a POTS phone from a hunt group.

(c) Call forwarding within a hunt group is valid only if the Featurephone is the last member of a terminal hunt group without camp-on.

(d) Meet-Me Conference

The Meet-Me Conference feature enables station users to establish or enter a conference call by dialing a special access code. Trunk parties may be added to the conference by the attendant.

STEP	OPERATION	RESULT
1	Inform the conferees of the access code, bridge digit, and time of conference.	
2	At the designated time go off-hook and enter access code and bridge digit.	(a) Dial tone. (b) "PLEASE DIAL" (c) "ENTERED CODE AND DIGIT" (d) Placed in conference.

See also: Conference

(e) Paging Access

PAGING ACCESS provides the user access to customer-owned voice paging equipment.

STEP	OPERATION	RESULT
	Begin from an off-hook idle state receiving dial tone and "PLEASE DIAL"	
1	Enter paging access code via the keypad or repertory dial button.	(a) "ENTERED ACCESS CODE" (b) User connected to paging apparatus.
2	Make necessary announcement. To disconnect, go on-hook.	(a) Voice heard. (b) Disconnected from paging apparatus. (c) Display cleared.

(f) Trunk Off-Hook Queuing

TRUNK OFF-HOOK QUEUING allows users to be queued automatically when a particular trunk group is busy.

See section on Station Equipment, Trunk Queuing, paragraph 2.2.5.

(g) Universal Night Answering

UNIVERSAL NIGHT ANSWER (UNA), when activated, allows users to answer attendant directed calls. When the attendant activates the feature, all calls directed to the attendant or transferred to the attendant will cause audible general alerting. The call may then be answered by any station.

STEP	OPERATION	RESULT
1	The attendant has activated UNA.	Incoming calls will cause general audible alerting.
2	To answer an incoming call. (a) Go off-hook. (b) Enter the UNA access code via keypad or repertory dial button.	(a) Dial tone heard. (b) "PLEASE DIAL" (c) Connected to incoming call.
3	Proceed with any call.	

3.7.34 Privacy (46)

On multi-line sets, each set has automatic privacy on the control line appearance during calls which have been initiated by that set. Once a line has been seized by one set, no other set may barge into the connection except the set which has the control appearance of that line.

(a) Privacy Release - Activation/Deactivation

STEP	OPERATION	RESULT
1	To activate privacy release while on the control line, depress PRIVACY RELEASE. A second phone may now go off-hook on this line and be connected in **a** three-way call.	PRIVACY RELEASE LED on.
2	To cancel privacy release, depress PRIVACY RELEASE.	PRIVACY RELEASE LED off.

(b) **Barge In**

On multi-set lines, each set has automatic privacy during calls which have been initiated or answered on that set. However, the control appearance of the line may Barge In by going off-hook when the line is busy on a different set. The control appearance will then be automatically placed in a three-way connection with the call previously established on the non-control appearance of the same line.

STEP	OPERATION	RESULT
	Begin with a multi-set line busy on a non-control appearance, privacy not released and all associated busy LEDs ON.	
1	User of control appearance goes off-hook on the line. If the current call is not a three-way call.	(a) Both parties hear break-in tone. (b) Control appearance forms a three-way conversation. (c) Control appearance phone display blank.
	If any party releases,	Remaining parties are connected in a two-way conversation.
	If the non-control appearance of the line depresses the HOLD button,	"ACTION DENIED"
	If the current call is in the ringing, dialing, consultation or transfer mode,	(a) Control appearance hears busy tone. (b) "ACTION DENIED"

3.7.35 **Release "RLS" (17)**

RELEASE allows the disconnect/connect of a line while using the optional headset.

STEP	OPERATION	RESULT
1	To release the line, depress RELEASE.	(a) RELEASE LED on. (b) "LINE RELEASED"
2	To connect to the line, depress RELEASE.	(a) Dial tone heard. (b) "PLEASE DIAL"
3	Continue as with Basic Calling (Internal/External).	

3.7.36 Reminder “REMIND” (13)

REMINDER permits the user to receive an audible and visual alarm at a previously scheduled time. The alarm may be scheduled on a daily, weekly, or calendar event basis. In addition to the alarm, the user may choose to have a preprogrammed reminder message to accompany the alarm. **A** reminder message consists of three components:

- alphanumeric text
- time and date of the reminder
- digit pattern to be dialed

The user may program up to eight reminder messages.

STEP	OPERATION	RESULT
	Reminder Entry	
1	Depress and hold REMIND.	(a) Beep heard. (b) “REMIND” (c) REMIND LED on.
2	Release.	(a) “INDEX = X" (b) REMIND LED flashes.
3	Enter an index number from 1 to 8, e.g., 1.	“INDEX = I”
4	Depress #.	
5	Enter frequency of reminder, e.g., 1. 1 = Daily Reminder 2 = Weekly Reminder 3 = Event Reminder	“DAILY = 00:00A" 1 = “DAILY = 00:00A" 2 = “WEEKLY = 00:00A DAY” 3 = “EVENT = 00:00A 00/00"
	Daily Reminder	
1	Enter two-digit number to set hour.	“DAILY = 02:00A DAY”
2	Depress # to change fields.	“DAILY = 02:00A DAY”
3	Enter two-digit number to set minutes.	“DAILY = 02:30A DAY”
4	Depress #.	“DAILY = 02:30A DAY”
5	Enter 1 (AM) or 2 (PM).	“DAILY = 02:00P DAY”
6	Depress #.	“I = "

STEP	OPERATION	RESULT
	Weekly Reminder	
1	Enter two-digit number to set hour.	"WEEKLY = 02:00A DAY"
2	Depress # to change fields.	"WEEKLY = 02:00A DAY"
3	Enter two-digit number to set minutes.	"WEEKLY = 02:30A DAY"
4	Depress #.	"WEEKLY = 02:30A DAY"
5	Enter 1 (AM) or 2 (PM).	"WEEKLY = 02:00P DAY"
6	Depress #.	"WEEKLY = 02:30P DAY"
7	Enter day as follows: 1 = Sunday 2 = Monday 3 = Tuesday 4 = Wednesday 5 = Thursday 6 = Friday 7 = Saturday	"WEEKLY = 02:30P MON"
8	Depress #.	"1 = "
	Event Reminder	
1	Enter two-digit number to set hour.	"EVENT = 02:00A 00/00"
2	Depress # to change fields.	"EVENT = 02:00A 00/00"
3	Enter two-digit number to set minutes.	"EVENT = 02:30A 00/00"
4	Depress #.	"EVENT = 02:30A 00/00"
5	Enter 1 (AM) or 2 (PM).	"EVENT = 02:00P 00/00"
6	Depress #.	"EVENT = 02:00P 00/00"
7	Enter month, #, and date.	"EVENT = 02:00P 06/84"
8	Depress #.	"1 = "

NOTES:

1. If an alarm or reminder is not retired by the user within 8 hours of having matured, it will automatically be retired by the instrument.

2. If another alarm/reminder comes due while one is currently active, it will be placed in queue until the current alarm/reminder is retired, at which time the queued alarm/reminder is activated. The queue shall allow a total of eight alarms or reminders to be queued and processed on a first-in, first-out basis.

3.7.37 Remote Tone "TONES" (44)

Remote Tone Send permits the instrument user to cause succeeding keypad button depressions resulting in end-to-end DTMF tone signals over the tip-and-ring wire pair. This feature, when activated, also causes the cancellation of any interdigital timing and digit processing being performed by the switch.

STEP	OPERATION	RESULT
	Begin in off-hook dialing mode. Busy and I-USE LEDs on, "PLEASE DIAL" and dial tone heard.	
1	Depress TONES.	(a) TONES LED on for 1/2 second. (b) Notice of intention to transmit tones sent to switch.
2	Send TONES.	Tone pairs heard.
3	To exit tone send mode, go on-hook.	

3.7.38 Repeat Last Number "LAST #" (06)

Last Number allows the user to initiate a call automatically to the last number manually dialed by the user by retaining in its memory the last number manually dialed. Depression of the LAST # feature button will cause a repeat of the dialing in the same number as if this feature button were a repertory dial feature button.

STEP	OPERATION	RESULT
	Begin in an off-hook receiving dial tone with "PLEASE DIAL" state	
1	Depress LAST #.	(a) LAST # LED on. (b) Last number displayed "XXXX". (c) Digits sent to switch.

STEP	OPERATION	RESULT
		(d) LAST # LED off. (e) Display clears or reverts to the time and date.
2	Continue as with a normal line-to-line call.	

NOTE: If the user depresses the LAST # feature button with the instrument on-hook, the instrument shall initiate automatic line seizure followed by on-hook dialing. Refer to LINE SEIZURE for more information.

3.7.39 Repertory Dial "REP DIAL" (09)

Repertory Dial provides for sending a programmed repertory number to the switch. The associated LEDs are on during programming and digit sending sequence.

If REP DIAL is depressed while on-hook, the SPEAKERPHONE (if equipped) or MONITOR LED will light and on-hook dialing will be initiated. The repertory dial button provides the capability of storing a sequence of 16 digits and pauses and also provides the capability of storing a 12-character alphanumeric identifier to be associated with the number.

STEP	OPERATION	RESULT
	Begin in an off-hook dialing mode. BUSY and I-USE LEDs on, dial tone heard, "PLEASE DIAL" displayed.	
1	Depress desired REP DIAL button.	(a) REP DIAL LED on during sending sequence. (b) DTMF signaling heard. (c) "XXXX---" followed by "AAA---" as programmed.
2	Proceed as with a normal line-to-line call.	

NOTE: The user may have multiple buttons on the instrument and/or on the 30-button Add-On Module (if equipped), programmed as REPERTORY DIALING buttons.

3.7.40 Ringer Cutoff "'NO RING" (01)

NO RING allows the station ringer to be silenced for the current call or for all future calls. If the NO RING button is depressed while the phone is ringing, the ring is silenced only for that call. If the phone is silent when the NO RING button is depressed, all future calls are silenced. Incoming calls are then signaled by a winking LED and may be answered in the normal manner.

STEP	OPERATION	RESULT
1	To deactivate the station ringer, depress RINGER CUTOFF.	(a) RINGER CUTOFF LED on. (b) "RINGER OFF"
2	To reactivate the station ringer, depress RINGER CUTOFF.	(a) RINGER CUTOFF LED off. (b) "RINGER ON"

3.7.4-l Station Speed Calling "SPD" (08)

STATION SPEED CALLING allows the user to have up to 20 numbers programmed which, upon activation, causes the instrument to send the selected number to the switch automatically.

STEP	OPERATION	RESULT
	Begin in an off-hook dialing mode. BUSY and I-USE LEDs on, dial tone heard, "PLEASE DIAL" displayed.	
1	Depress SPEED.	(a) SPEED LED on. (b) "CODE = "
2	Enter two-digit code associated with the desired destination (10 to 29).	(a) "CODE = XX" (b) "XPXXX----" number dialed as programmed. (c) "ABCD---" alpha information. (d) SPEED LED off.
3	Depress #.	
4	Proceed as with a normal line-to-line call.	

NOTES:

1. There can be only one Station Speed Calling feature button per instrument.
2. In the event the user has not programmed an alphanumeric character string with the number, the display will show only the speed calling code.

3. If code number entered has no number stored, display shows "HANG UP and REDIAL".

3.7.42 System Speed Calling "SYS CL" (47)

SYSTEM SPEED CALLING allows the user to access system group speed calling numbers by use of SYS CL button and a code number to identify any one of the 1000 numbers.

STEP	OPERATION	RESULT
	Begin in an off-hook dialing mode. BUSY and I-USE LEDs on, dial tone heard, "PLEASE DIAL" displayed.	
1	Depress SYS CL button.	(a) SYS CL LED flashes at 120 IPM. (b) "CODE = "
2	Enter, via the keypad, the three-digit code (000 - 999) associated with the desired number.	(a) SYS CL LED on. (b) "CODE = XXX" (c) "XPXXX--" number as programmed. (d) SYS CL LED off.
3	Depress #.	(a) SYS CL LED off. (b) "DIALING"
4	Proceed as with a normal call.	

NOTES:

1. There can only be one SYSTEM SPEED CALLING feature button per instrument.
2. The range of the three-digit system speed calling code is specified by data base programming in the switch. Thus, the instrument will not verify the digits.

3.7.43 Saved Number Redial "SAVE #" (07)

The SAVED NUMBER button functions as a REP DIAL button, except the saved number redial button is associated with a user programmed number, or the number associated with a message that was saved while being displayed.

STEP	OPERATION	RESULT
1	To save a number, depress SAVE # (after dialing and before hanging up).	

STEP	OPERATION	RESULT
2	To call a saved number, depress SAVE #.	**(a)** Number dialed appears. (b) Number is automatically dialed.

(a) **Saved Number Redial - To Store Number**

STEP	OPERATION	RESULT
1	Depress and hold SAVE NO.	(a) SAVE NO. LED on. (b) Beep heard. (c) "SAVE # REDIAL"
2	Release.	(a) SAVE # LED flashes. (b) "XXXX" or blank if no number saved.
3	Enter desired number.	"XXX--" as entered.
4	Depress and hold SAVE NO.	(a) SAVE NO. LED on steady (b) Beep heard. (c) "NEW ENTRY STORED"
5	Release.	(a) SAVE NO. LED off. (b) Display clears or reverts to the time and date.

See also: Special Data Programming/Entry

3.7.44 Shift "SHIFT" (43)

The SHIFT button changes the mode of the remaining feature buttons to their preprogrammed alternate features. This mode will remain active until another feature button is depressed or the SHIFT button is depressed again, returning the feature buttons to their main programmed function.

STEP	OPERATION	RESULT
	Begin from any state.	
1	Depress SHIFT.	(a) SHIFT LED on.
2	Depress any feature button.	(a) BOTTOM LED on.

STEP	OPERATION	RESULT
		(b) Any feature button programmed for dual mode use is shifted to its alternate function. (c) Feature name appears.
3	To exit shift mode, depress SHIFT a second time.	(a) SHIFT LED off. (b) BOTTOM LED off.
4	Depress feature button.	Feature is functional.

NOTES:

1. Only one SHIFT button may appear on the keyboard.
2. The SHIFT button (on the instrument itself or on the add-on module) may not have an alternate function.
3. LINE buttons (which can only appear on the instrument itself) may not have an alternate function.
4. Any button on the instrument (with the exceptions cited in items 1 and 2, above) may be programmed to have the DSS function as a primary and/or an alternate function.
5. Buttons on the 30-button add-on module (if equipped) may not have an alternate function.
6. When entering the programming mode while the shift is active, the shift mode will remain active throughout the entire programming sequence, whether programming new alternate functions to the buttons and/or new data to the existing alternate functions. Once programming the alternate functions is complete, simply depress the SHIFT button once, and the SHIFT LED will be extinguished.

3.7.45 Speakerphone "SPKRPH" (18)

The Featurephone is available with an optional speakerphone which shares the amplifier speaker and volume control with the Monitor Circuit. A SPEAKERPHONE button depression will activate the speaker and microphone. If the user wishes to mute (i.e., turn off the microphone), a depression of the MONITOR button will disable the microphone.

STEP	OPERATION	RESULT
1	To answer an incoming call, depress SPKRPH.	(a) SPKRPH LED on. (b) Two-way voice path established through Speakerphone.

STEP	OPERATION	RESULT
		(c) "FX XXXX--" Call Source.
2	Converse with caller.	
3	To terminate call, depress SPKRPH a second time.	(a) SPKRPH LED off. (b) Display clears.
4	To initiate a call, seize a line and dial by any of the methods previously discussed, depress SPKRPH to converse.	Varies.
5	To mute the microphone while on a Speakerphone call, depress MONTR.	(a) MONTR LED on. (b) Microphone disabled.

NOTES:

1. Whenever the instrument is off-hook or when the Speakerphone or Monitor are on, and the instrument initiates ringing on a multi-line set, the ringing shall be abbreviated to present only three cycles of ringing to the instrument user. Although the ringing is stopped, the visual alerting on the line button shall continue until answered.

2. The message "PLEASE HANG UP" is displayed when the user turns the Speakerphone (or Monitor) feature on and the handset is off-hook. The message remains until the user places the handset on-hook.

3.7.46 Station Lock "LOCK" (02)

The STATION LOCK feature enables the user to lock the Integrated Featurephone to restrict usage by unauthorized parties. When the station is locked:

1. No calls of any type may be originated.
2. The keypad and repertory dialing functions will not operate.
3. The other features, resident and system, will not operate (including FLASH, NEW-CALL, HOOKSWITCH FLASH).
4. Incoming calls will cause ringing and associated displays.
5. Incoming voice paging will operate.
6. Incoming calls may be answered.
7. The UNLOCK feature will still be accessible.

The phone can be unlocked by the user upon entering the proper code or by the system (switch).

(a) **Station Lock - Lock Operation**

STEP	OPERATION	RESULT
	Begin in an idle on-hook state.	
1	To activate the lock, depress LOCK.	(a) LOCK LED on. (b) "STATION LOCKED" (c) Display clears or reverts to the time and date.
2	Any attempt to access features locked will cause	(a) Triple beep heard. (b) "STATION LOCKED"

(b) **Station Lock - Unlock Operation**

STEP	OPERATION	RESULT
	Begin from an idle on-hook or talking off-hook condition.	
1	Depress LOCK.	(a) LOCK LED flashes at 120 IPM. (b) "LOCK CODE = ?"
2	Enter the 3-digit lock password previously programmed. If code is correct,	(a) "LOCK CODE = ###" Code is not displayed. (b) LOCK LED off. (c) "STATION UNLOCKED" (d) Display clears or reverts to the time and date.
	If code is incorrect,	(e) LOCK LED on. (f) Continuous beep heard.
	If code is not completely entered within ten seconds, or no entry after one minute,	(g) Triple beep heard. (h) LOCK LED on. (i) Phone locked.

NOTES:

1. The operation of the LOCK feature shall not interfere with a call in progress.
2. The factory preprogrammed unlock code is 000.
3. If a user forgets the unlock code, contact a telephone maintenance person.

3.7.47 Time and Date "CLOCK"

The time and date feature button displays the correct time, day, and date whenever depressed. A second depression of the button, while the time/date display is active, will restore the display to its former use.

In addition to the time and date button, the instrument can be programmed to display automatically the correct time, day, and date whenever the phone is idle. The time/date display will remain until preempted by a higher priority function (i.e., phone goes off-hook, user enters programming mode, expiration of a reminder/alarm message, etc.). See Table 4.1 for the time, month, day, and time code conversion.

STEP	OPERATION	RESULT
	Begin from an idle on-hook state.	
1	To display time and date, depress CLOCK.	(a) CLOCK LED on. (b) "HH-MM-SS MO DA" Time and Date.
2	To revert to previous display, depress CLOCK a second time.	(a) CLOCK LED off. (b) Original Display.

3.7.48 Types of Sets

A Featurephone may be either a single-line set or a multi-line set. A single-line set has only one line associated with the set, while a multi-line set has from 2 to 16 lines associated with the set giving the user access to any of the lines associated with the set.

(a) Single-Line Set

A single-line set has only one line associated with it. It may have DSS buttons, but it is not permitted to have any line buttons. Therefore, going off-hook shall directly seize the line if it is idle, or answer it if it is ringing.

(b) Multi-Line Set

A multiline set has more than one line associated with it. Therefore, it will have buttons assigned as line buttons. There shall be a one-to-one correspondence between line buttons and directory numbers. Going off-hook shall only result in seizing or answering a line if that particular line has been selected (manually or automatically) before or after going off-hook.

Since these instruments do have line buttons with associated status LED indicators, the five conditions that a line may be in will be indicated by the primary status indicator:

(1) Line In Use
(2) Line Ringing

(3) Call On Hold
(4) idle Line
(5) Request For Input

The secondary status indicator will provide the I-USE indication when the user is off-hook on a particular line, the I-HOLD indication when the user has placed a call on hard hold, or it will indicate the preselected or preference line choice when the instrument is on-hook.

NOTES:

1. The single-line on the single-line set may be either a single-set line or a multi-set line.

2. The single-line on the single-line set must be a prime control line.

3. Secondary line appearances are only allowed on multi-line sets.

4. A multi-line set must have at least one control line appearance.

THIS PAGE INTENTIONALLY LEFT BLANK.

4.0 ATTENDANT CONSOLE

4.1 General

The Attendant Console, shown in Figure 4.1, is a microprocessor-controlled, desk-top console, that provides total control and flexibility for processing calls through the system. The attendant can place the Attendant Console in other control modes during nighttime, maintenance, or unattended periods.

The Attendant Console contains compact electronics and all controls and indicators needed for complete monitoring and controlling of calls through the system. Both incoming and outgoing calls are answered, extended, camped on, put on hold, or released by depressing various pushbuttons.

A 32-character alphanumeric display across the top of the Attendant Console furnishes full call information to the attendant. The type of call indication is present as long as the attendant is servicing the call/recall. The type of call display shows class of service and system fault alarm information.

The Attendant Console is equipped with a headset/handset receptacle located at the rear and on one side. The rear receptacle is used for supervisory headset/handset monitoring; the side receptacle is used for the attendant headset or handset. When the attendant headset or handset is removed, the Attendant Console is in the night answer mode.

The Attendant Console includes the following features:

(a) Twenty-six pushbutton keyboard with associated LEDs

(b) Twelve-key dial pad

(c) Thirty-two character alphanumeric LED display

(d) Headset/handset and associated jacks

(e) Time-of-day display and update

(f) Internal self-test diagnostics

4.2 Controls, Indicators, and Displays

Various controls, indicators, and displays located on the Attendant Console (Figure 4.2) are listed below with a brief functional description.

ON/OFF TOGGLE SWITCH. Located at the left rear and used to turn the Attendant Console ON or OFF. When the Attendant Console does not respond properly, the power can be turned off and on again to re-initialize the Attendant Console.

FUSE HOLDER AND FUSE. Circuit protection device located on the left rear of the Attendant Console.

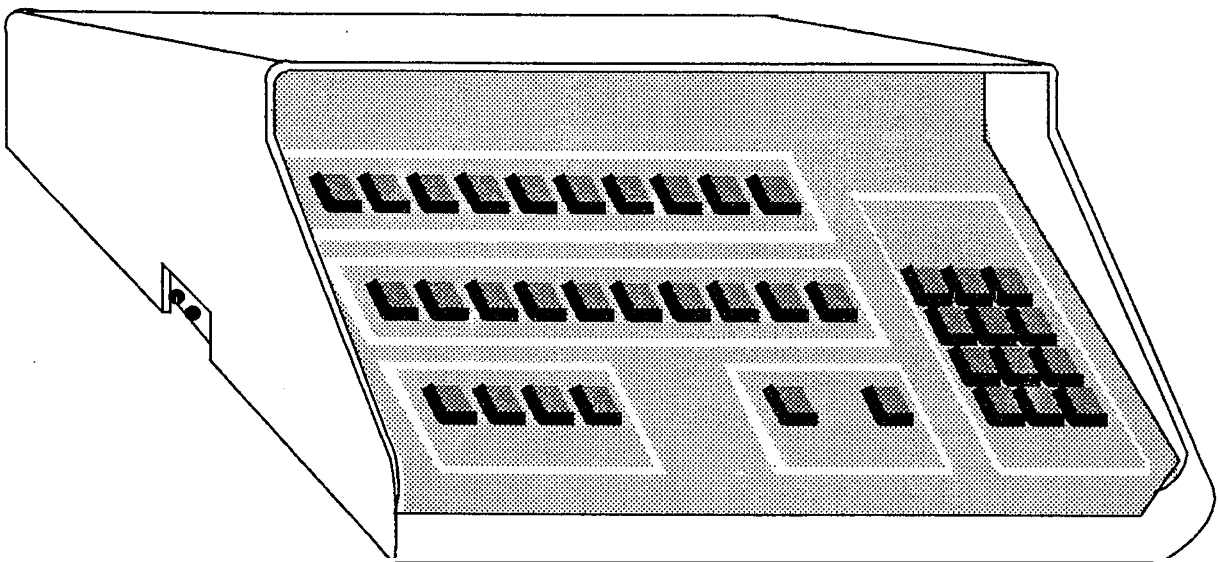


Figure 4.1 Attendant Console

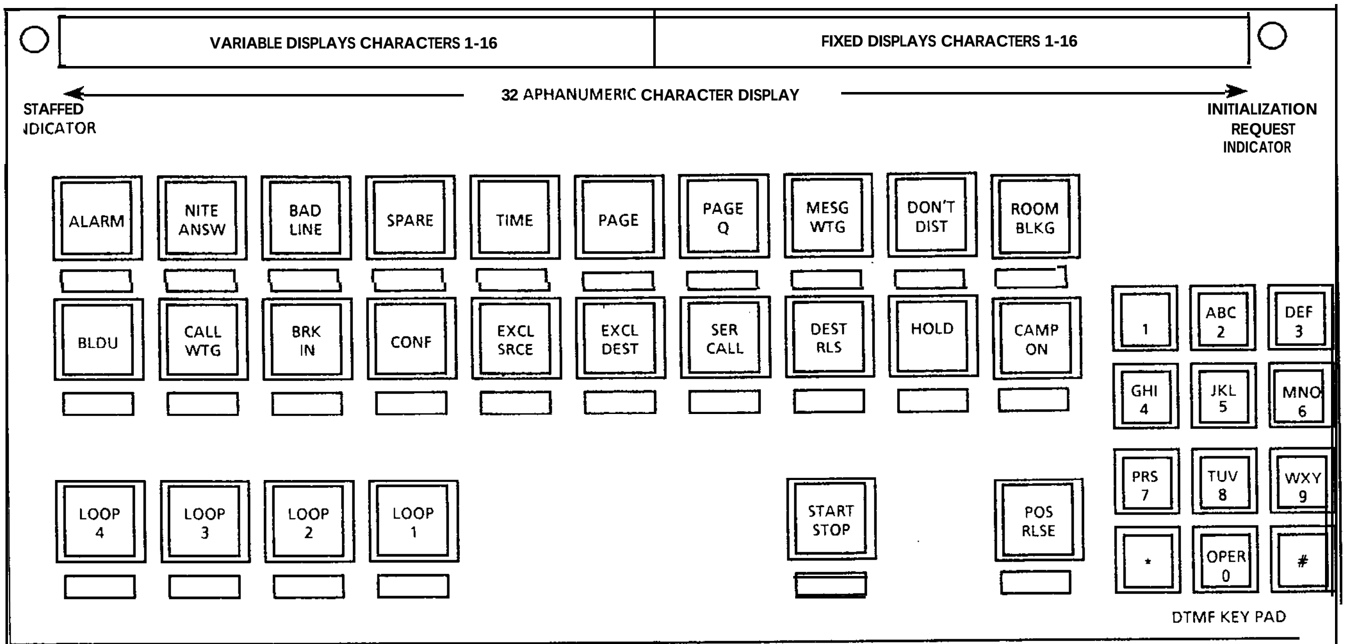


Figure 4.1 Attendant Console, Controls, and Indicators

ELECTRONIC RINGER VOLUME CONTROL. Slide bar located on the right rear of the Attendant Console and used only to increase or decrease the ringer volume.

REQUEST FOR INITIALIZATION LED. Flashes when the Attendant Console is initialized, and goes off when complete.

STAFFED LED. Lights only if the position is staffed

DISPLAY FIELD. A 32-character alphanumeric display that has two 16-character fields (fixed, right; variable, left).

DTMF KEYPAD. The standard 12-button keypad layout is used for attendant DTMF signaling to the system.

Pushbuttons and LEDs. The Attendant Console pushbuttons are switches that do not light and when depressed, make contact with internal circuits. An associated LED is located directly below each pushbutton for visual indication/confirmation purposes.

ALARM. LED lights to indicate a fuse alarm or power failure, and flashes to indicate trouble with a particular line or trunk or group of lines and/or trunks.

NITE ANSW (NIGHT ANSWER). Activates the night answer mode level of operation (entered as a single digit on the keypad), or marks the Attendant Console as being unstaffed. The level of night answer mode is displayed on the type of call display. If the night answer pushbutton is depressed when the LED is on, the night answer mode is returned to the day mode.

BAD LINE. Allows the posting of a bad line report on the maintenance teletypewriter if the pushbutton is depressed when a complete connection is present.

SPARE. Reserved for future enhancements.

TIME [TIME OF DAY (TOD)]. Allows the attendant to override any variable display with the time-of-day display in hours and minutes.

PAGE. Allows direct access to the paging system as long as the pushbutton is depressed.

PAGE Q (PAGE QUEUE). Allows placement of an incoming call in a page-queue hold condition.

MESG WTG (MESSAGE WAITING). Allows the user to cause a telephone-associated LED to flash an indication to a station that a message is waiting. This pushbutton generally is used with a feature enhancement.

DON'T DIST (DO NOT DISTURB). Allows blockage of all calls to a station when requested. This pushbutton is reserved for the hotel/health care feature package.

ROOM BLKG (ROOM BLOCKING). Allows blockage of incoming calls to all stations and diverts these calls to 120 IPM tone. This pushbutton is reserved for the hotel/health care feature package.

BLDU (BUSY LAMP DISPLAY UNIT). Allows selection of the Hundreds Group for display on the busy lamp line field.

CALL WTG (CALL WAITING). LED on condition indicates that the first level of calls waiting has occurred. The LED flashing indicates that the second level of calls waiting has occurred. Depressing the pushbutton during either of these conditions will distribute a call to an Attendant Console that is in the night answer mode.

BRK IN (BREAK IN). Allows break-in on two-way conversations.

CONF (CONFERENCE). Allows addition of stations, trunks, or the attendant to the first conference circuit. LED indicates the status of either conference circuit.

EXCL SRCE (EXCLUDE SOURCE). Allows private consultation with the calling party. The pushbutton can also be used to control the three-way connection between the called station, the attendant, and the outside calling party.

EXCL DEST (EXCLUDE DESTINATION). Allows private consultation with the called party after the call has been completed. The pushbutton can also be used to control the three-way connection between the called station, the attendant, and the outside calling party.

SER CALL (SERIES CALL). Depressing this pushbutton after extending an incoming call automatically causes the trunk to recall the attendant when the station hangs up so the trunk can be extended to the next station in the series call sequence. Alternatively, the attendant can call or be called from a station (or have a station diverted to the Attendant Console due to an ACOF restriction on an accessed trunk group) and extend a trunk to the station using this pushbutton, re-initiating the automatic recall sequence to notify the attendant that the trunk is available for the next user. The LED on indicates action confirmation.

DEST RLS (DESTINATION RELEASE). Allows release of a called station or trunk if a dialing error is made or a busy signal is received.

HOLD. Used to place a call in a waiting condition (on hold) until the attendant can return to the call.

CAMP-ON. Allows placement of an incoming call in a waiting condition if the called station is busy. When camp-on is used, the system automatically extends the calling party to the called station when the called station becomes idle.

LOOP 1 through **LOOP** 4. LEDs which indicate status of the loop.

START/STOP. In the START mode, this pushbutton activates the keypad and provides dial tone and in the STOP mode, removes tick tone on outgoing calls.

POS RLSE (POSITION RELEASE). Allows the attendant to release from any type of call.

Class-of-Service Display. When a LOOP pushbutton is depressed to answer an incoming call from a station or tie line, a one-digit display indicates the displayable class of service (0 through 9), and a two-digit display indicates the N-displayable class of service (00-15) of the incoming call. In a no-answer situation, the class-of-service display of the called station appears after the LOOP pushbutton is depressed.

The class-of-service operation allows/disallows access to trunks and features of the system. On a per-line basis, each line is marked with a class identifying the line as 0 of 9 displayable classes of service and 00 of 15 N-displayable classes of service. If a line with a restricted access code as a function of classmark is dialed, the call is routed to an intercept facility (recorded announcement, attendant, reorder tone etc.). The displayable and N-displayable classes of service are displayed at the Attendant Console every time the line is serviced by an attendant.

Displayable Classes of Service. The following can be combined, as desired, to establish 10 types of call combinations that can be allowed or disallowed as displayable classes of service.

(a) Dictation equipment access

(b) Meet-me conference access

(c) Progressive conference access

(d) Paging access

(e) Station-to-station access

(f) Toll restriction

(g) Switched direct line service

(h) Code call access

(i) On-Network Most Economical Route Selection (ON-NET MERS) access

(j) Off-Network Most Economical Route Selection (OFF-NET MERS) access

(k) Maintenance access

(l) Trunk groups 00 through 63 access

(m) Release Link Trunk (RLT) access (CAS option)

(n) Modem line access

N-Displayable Class of Service. The following can be combined as desired to establish 16 types of call combinations that can be allowed or disallowed.

(a) Attendant information calls

(b) Station camp-on with automatic recall

(c) Data line security

(d) Dial call pickup

(e) Universal Night Answer (UNA)

(f) Executive override

(g) Speed calling

(h) Terminating call waiting, DID

(i) Terminating call waiting, non-DID

(j) Originating call waiting

(k) Call forwarding, fixed

(l) Call forwarding, variable

(m) DID allowed

(n) Terminating service only

(o) Originating service only

(P) Hookswitch flash

(q) Call park

(r) Call hold

(s) Secondary directory number access

(t) Administrative telephone

(u) Service telephone

(v) Display telephone

(w) Station activation of timed-reminder feature

(x) Message-center call-waiting with automatic clearance of the message-waiting function

(y) Computer access

(z) Special common carrier access

(aa) Receive DID

Trunk or Station Display. A four-digit display of the trunk or line identity that appears in the variable left portion of the display-characters field. When the LOOP pushbutton is depressed to answer a call from a station or a tie line, this four-digit display gives the number of the calling station or calling trunk. The number of the called station in camp-on, busy, and no-answer situations is displayed when the LOOP pushbutton is depressed.

Type-of-Call Displays. A display of predetermined messages that appear in the fixed portion (right) of the display characters field (e.g., INFORMATION, FX, NO ANSWER, etc.).

NOTE: The left 16 characters display variable data: line data, class-of-service data, etc. The right 16 characters display fixed predetermined messages and input from the keypad.

LOCAL. Indicates an incoming CO call. The associated LOOP LED flashes at 120 IPM.

INFORMATION. Indicates that a station desires assistance from the attendant and has dialed 0. The associated LOOP LED flashes at 30 IPM.

NOTE: When the attendant recall-on-hold has timed out, the original display flashes at 60 IPM. The associated LOOP LED winks at 60 IPM.

SERIES CALL. At the completion of an incoming series call, the station hangs up and the attendant is called. On answer, the SERIES CALL message and the trunk number appear on the display, and the attendant can converse with the calling party. At the completion of an outgoing series call, the station hangs up, the trunk is released, and the attendant is called. On answer, the SERIES CALL message flashes on the display along with the previous caller's station number and COS.

NO ANSWER. Provides for answering a call for service resulting from a time out on an unanswered call. The type of call could be a normal attendant-extended call (information or incoming), delayed call, camp-on call, or station-transferred call (transferring party on-hook) either busy or unanswered. The associated LOOP LED flashes at 60 IPM.

STILL BUSY. Refers to a camp-on call remaining busy for a predetermined period (data base programmable), and the call being diverted to the attendant or an attempt to transfer a call to a busy or unequipped device. The associated LOOP LED flashes at 60 IPM.

TIE LINE. Indicates an incoming tie-line call. The associated LOOP LED flashes at 120 IPM.

FX (Foreign Exchange). Indicates an incoming call from a foreign exchange. The associated loop LED flashes at 120 IPM.

INTERCEPT. Available when required. The associated LOOP LED flashes at 30 IPM.

LD RESTRICTION (Long Distance Restrict). Diverts attempted toll calls from toll-restricted stations to the attendant. The LD RESTRICTION display identifies these calls. The associated LOOP LED flashes at 30 IPM.

TRANSFER. Provides for attendant recall via hookswitch-flash-dial 0 operation. The associated LOOP LED flashes at 120 IPM.

WATS. Indicates an incoming call from a WATS trunk. The associated LOOP LED flashes at 120 IPM.

CALL WAS PARKED. Indicates a call was placed in a call-park queue. The display flashes at 60 IPM.

DIVERT FROM ACOF (Attendant Control of Facility). Indicates that dial access by all stations or tie trunks to a trunk group has been diverted to the Attendant Console.

DND ACTIVE (Do Not Disturb active).

DND NOT ACTIVE (Do Not Disturb not active).

DND and MG WT ACT (Do Not Disturb and Message Waiting active).

RR BLK ACTIVE (Room-to-Room Blocking active).

RR BLK NOT ACT (Room-to-Room Blocking not active).

MSG WAITING, NO MSG WAITING. Indicates the status of message waiting.

PRIORITY CALL. Indicates that the attendant has answered a priority call. LOOP LED flashes at 120 IPM.

NO DIAL ALARM. Indicates that the attendant has answered a no-dial alarm; LOOP LED flashes at 60 IPM.

DAY MODE. Indicates that the Attendant Console is in the day mode.

ENTER 1 or 2 (data base programmable). Indicates night answering service has been selected and directs selection of Night Answer 1 or Night Answer 2.

NIGHT ANSWER 1 (data base programmable). Indicates that the Attendant Console has assumed PNA or UNA mode of night answer service.

NIGHT ANSWER 2 (data base programmable). Indicates that the Attendant Console has assumed UNA or PNA mode of night answer service.

ENTER 100's. Indicates that the BLDU pushbutton has been depressed and directs that a 100's group (00-99) be selected.

NOTES:

1. Most of the pushbuttons on the keyboard are depressed and released to perform a function. The PAGE and BRK IN pushbuttons are exceptions, however, and must be held in the depressed position for the duration of the function. When released, these pushbuttons are automatically cleared.

2. The fixed portion of the display is cleared when the START/STOP key is depressed to enter digits. Then each pushbutton entry on the keypad is displayed in this portion of the display. If more than 16 entries occur, the input digits are scrolled off of the 16-character display on a first-in, first-out basis. (This situation is likely to occur during group speed-calling updates.)

3. If two keys are depressed simultaneously, neither key is recognized. Keys operate in a two-roll-over mode. In the page or break-in case, the appropriate key is held down while the function is activated. The depression of any other key that occurs while a page or break-in is in effect, is ignored. The page or break-in key release code is sent before the rolled-over key is processed.

4. Certain combinations of key activations are illogical and should not occur. A key activation is indicated by the associated key LED being on (or flashing). If the attendant attempts an illogical combination, the latest key depressed is ignored.

5. All pushbuttons are nonlocking. All LEDs light, with the exception of HOLD, DEST RLS and CAMP ON.

4.3 Headset/Handset

The headset/handset can be plugged into the left side or the right rear of the Attendant Console. When the headset/handset is not inserted, the Attendant Console functions in the unstaffed mode. All LEDs are off and the appropriate unstaffed message is automatically sent to the system. When the Attendant Console is in the unstaffed mode, the user can request specific stand-alone maintenance diagnostics from the Attendant Console keypad. These diagnostics test all of the displays (LEDs and alphanumeric display), display Attendant Console parameters, and show the contents of the various internal registers. Refer to Section TL-130200-I 001 for diagnostic procedures.

4.4 Multiple Attendant Console Operation

When two Attendant Consoles are used, one is designated as the master. The master Attendant Console provides control of the night-answer selection.

Each Attendant Console is assigned a directory number so calls can be directed to a specific Attendant Console. If one Attendant Console calls the other Attendant Console, the call is placed at the top of the called Attendant Console call-waiting queue.

4.4.1 Call-Waiting Queue

Calls are generally distributed evenly to the Attendant Consoles. Attendant Console 1 receives the first call and Attendant Console 2 the next. When the Attendant Consoles remain busy until the second level in the call-waiting queue is reached, the CALL WTG LEDs on the Attendant Consoles flash. (Option: If the system is programmed for overflow, the universal-night-answer mode also is activated when the second call-waiting queue is reached.)

4.4.2 Queued-Call Diversion

If the operator at an Attendant Console other than the master Attendant Console enters into the night-answer mode and the second level of the call-waiting queue is reached, the operator can divert calls one at a time to the Attendant Console by depressing the flashing CALL WTG LED. The operator can then process calls until the overflow is relieved and still remain in the night-answer mode.

4.5 Preparing Attendant Console for Service

4.5.1 Day Service

STEP	OPERATION	RESULT
1	Ensure OFF/ON toggle switch is in the ON position.	All displays on momentarily.

STEP	OPERATION	RESULT
2	Plug handset/headset into Attendant Console.	(a) If not in night answer mode, STAFFED LED on. (b) ALARM LED flashes at 60 IPM. (c) INITIALIZATION LED flutters for 30 seconds (all other LEDs off). (d) Initialization is finished when the following occurs: 1. INITIALIZATION LED off. 2. NIGHT ANSWER 1 appears on display. 3. NITE ANSW LED on. 4. ALARM LED flashes.
3	Depress NITE ANSW pushbutton.	(a) STAFFED LED on. (b) ALARM LED off. (c) NITE ANSW LED off. (d) NIGHT ANSWER 1 message cleared. (e) Within two minutes TOD appears on display.

4.5.2 Internal Time-of-Day (TOD) Clock Display

STEP	OPERATION	RESULT
1	Depress and hold TIME pushbutton if TOD is not being displayed.	(a) Variable display clears. (b) TOD displayed in variable display field. (See Table 4.1 for time code conversion.)
2	Release TIME pushbutton.	(a) TOD cleared. (b) Original variable display restored.

4.5.3 System Real-Time Clock Display

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton.	(a) LOOP LED on. (b) POS RLSE LED on. (c) Dial tone heard.

Table 4.1 Time Code Conversion

TIME STANDARD	TIME MILITARY
Midnight - 12:59 a.m.	0000-0059
1 a.m. - 1:59 a.m.	0100-0159
2 a.m. - 2:59 a.m.	0200-0259
3 a.m. - 3:59 a.m.	0300-0359
4 a.m. - 4:59 a.m.	0400-0459
5 a.m. - 5:59 a.m.	0500-0559
6 a.m. - 6:59 a.m.	0600-0659
7 a.m. - 7:59 a.m.	0700-0759
8 a.m. - 8:59 a.m.	0800-0859
9 a.m. - 9:59 a.m.	0900-0959
10 a.m. - 10:59 a.m.	1000-1059
11 a.m. - 11:59 a.m.	1100-1159
12 p.m. - 12:59 p.m.	1200-1259
1 p.m. - 1:59 p.m.	1300-1359
2 p.m. - 2:59 p.m.	1400-1459
3 p.m. - 3:59 p.m.	1500-1559
4 p.m. - 4:59 p.m.	1600-1659
5 p.m. - 5:59 p.m.	1700-1759
6 p.m. - 6:59 p.m.	1800-1859
7 p.m. - 7:59 p.m.	1900-1959
8 p.m. 8:59 p.m.	2000-2059
9 p.m. 9:59 p.m.	2100-2159
10 p.m. - 10:59 p.m.	2200-2259
11 p.m. 11:59 p.m.	2300-2359

NOTE: Enter single-digit day and month codes with a leading zero (i.e., Jan = 01).

STEP	OPERATION	RESULT
2	Dial access code for system real-time clock display plus digit = 1. (See Table 4.1 for time code conversion.)	(a) Year and month displayed on attendant variable display field. (b) Dialed digits appear on fixed display.
3	Dial digit 2.	(a) Month and day displayed on attendant variable display field. (b) Dialed digits appear on fixed display.

STEP	OPERATION	RESULT
4	Dial digit 3.	(a) Day and hour displayed on attendant variable display field. (b) Dialed digits appear on fixed display.
5	Dial digit 4.	(a) Hour and minute displayed on attendant variable display field. (b) Dialed digits appear on fixed display.
6	Depress POS RLSE pushbutton.	(a) POS RLSE, START/STOP and LOOP LEDs go off. (b) Fixed and variable displays clear. (c) TOD displayed.

4.54 Real-Time Clock Update

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (l-4).	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial access code for real-time clock update.	Fixed display shows dialed digits.
3	Dial digits of time data YYMMDDHHMM for years, months, days, hours, and minutes. Hours and minutes are in military time (0000 to 2359). (See Table 4.1 for time code conversion.)	Fixed display shows dialed digits.
4	Dial execution digit (digit 1).	Confirmation tone heard.
5	Depress POS RLSE pushbutton.	(a) LOOP, POS RLSE and START/STOP LEDs off. (b) Display clears. (c) Displays TOD.

NOTE: To verify time entered, refer to Real-Time Clock display operation.

4.6 Emergency Service

4.6.1 Power Failure

Attendant Console has power and ALARM LED is on, indicating that fuse alarm or power failure has occurred.

STEP	OPERATION	RESULT
1	Depress ALARM pushbutton.	ALARM LED on.
2	Place a few test calls to stations to determine the extent of the trouble, then call repair service to report conditions and trouble found.	

4.6.2 Faults in System

Attendant Console has power and ALARM LED is flashing, indicating faults in system.

STEP	OPERATION	RESULT
1	Depress ALARM pushbutton.	Trouble number (three- or four-digit) may appear on the left portion of the display, but the ALARM LED continues to flash.
2	Record number and depress ALARM pushbutton again.	Another trouble number may appear.
3	Record number again and continue process until all fault numbers have been displayed.	(a) Another trouble number may appear. (b) ALARM LED will stop flashing when all fault numbers are indicated.
4	Call repair service to report trouble numbers recorded.	

4.6.3 Attendant Console Has Power But Is Inoperative

STEP	OPERATION	RESULT
1	Unplug and reinsert handset/headset plug in jack on left.	(a) If the Attendant Console is in day mode, STAFFED LED goes off and then on. (Go to Step 3 if this condition exists.)

STEP	OPERATION	RESULT
		(b) If Attendant Console is in NIGHT ANSWER mode, NIGHT ANSWER 1 (or 2) appears on the display and the NITE ANSW LED is on.
2	Depress NITE ANSW pushbutton.	STAFFED LED on and display and NITE ANSW LED off.
3	If service is not restored, turn power switch off, then back on.	INITIALIZATION LED flutters, and after a short time goes off, NIGHT ANSWER 1 appears on the display and ALARM LED flashes.
4	Depress NITE ANSW pushbutton.	STAFFED LED on and display and NITE ANSW LED off.
5	If above steps fail to restore Attendant Console to service, either the Attendant Console and/or its related circuits have a failure or a major system failure exists. Repair service must be contacted immediately.	

4.6.4 Power Source Fault or Major System Failure

When Attendant Console is without power and all controls and indicators are inoperative, indicating either a power source fault or a major system failure, contact repair service.

4.7 Answering an, Incoming Call

4.7.1 Incoming Station Call

STEP	OPERATION	RESULT
1	Call placed to Attendant Console.	(a) Ringing heard. (b) LOOP LED flashes at 30 IPM, indicating a call.
2	Depress flashing LED LOOP pushbutton.	(a) LOOP LED on steady. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Station number displayed. (e) “INFORMATION” appears in fixed variable display.

STEP	OPERATION	RESULT
		(f) Ringing stops. (g) Attendant and called party connected. (h) (Message Waiting option) MSG WTG pushbutton LED flashes at 60 IPM and “MSG WAITING” displayed if station has message-waiting feature activated.
3	Answer calling party and provide assistance.	
4	Depress POS RLSE pushbutton to release call.	(a) LOOP LED off. (b) START/STOP LED off. (c) POS RLSE off. (d) EXCL SRCE or EXCL DEST LED off if on. (e) Fixed display cleared (f) Trunk and station identification cleared from variable display. (g) Class-of-service digits cleared if present. (h) TOD displayed.

NOTE: The attendant must depress POS RLSE pushbutton to release the loop when calling party disconnects first.

4.7.2 Incoming Trunk Call

STEP	OPERATION	RESULT
1	Incoming CO trunk call is received by an Attendant Console.	(a) Ringing heard. (b) LOOP LED (l-4) flashes at 120 IPM.
2	Depress flashing LED LOOP pushbutton.	(a) Depressed LOOP LED on. (b) “LOCAL” appears in fixed display field. (c) Trunk number appears in variable display field. (d) START/STOP LED on. (e) POS RLSE LED on. (f) Ringing stops. (g) Transmission established between trunk and Attendant Console.

NOTE: In case of a misdial for any call, station, or trunk, depress DEST RLS pushbutton, then depress START/STOP pushbutton. Listen for dial tone and enter correct number. Dial tone is not applied after DEST RLS.

4.8 Placing Call on Hold

4.8.1 Place Station Call on Hold

STEP	OPERATION	RESULT
1	Ask the calling party to wait and depress the HOLD pushbutton.	(a) LOOP pushbutton LED winks slowly at 30 IPM. (b) TOD displayed.

NOTE: On call hold time out, attendant recalled and LOOP wink changes to 60-IPM wink.

4.8.2 Return to Call on Hold

STEP	OPERATION	RESULT
1	Depress winking LOOP LED pushbutton.	(a) LOOP LED on. (b) START/STOP LED on. (c) POS RLSE LED on. (d) Display indicates the original calling party information. (e) Original display flashes in the type-of-call display. (f) Music, if provided, is removed. (g) Transmission established between attendant and caller.
2	Report results to calling party and proceed as required.	

NOTE: To continue call on hold, repeat the hold procedure.

STEP	OPERATION	RESULT
3	Depress POS RLSE pushbutton to release.	(a) LOOP LED off. (b) START/STOP LED off. (c) POS RLSE LED off. (d) EXCL SRCE or EXCL DEST LED off if on. (e) Fixed display cleared. (f) Trunk and station identification cleared from variable display.

STEP	OPERATION	RESULT
		(g) Class-of-service digits cleared if present. (h) TOD displayed.

4.8.3 Trunk Calls and Automatic Recall

STEP	OPERATION	RESULT
1	Incoming CO trunk call received by Attendant Console.	(a) Ringing heard. (b) LOOP LED (l-4) flashes at 120 IPM.
2	Depress flashing LOOP LED pushbutton (l-4).	(a) LOOP LED on. (b) "LOCAL" appears in fixed display field. (c) Trunk number appears in variable display field. (d) START/STOP LED on. (e) POS RLSE LED on. (f) Ringing stops. (g) Transmission established between trunk and Attendant Console.
3	Depress HOLD pushbutton.	(a) HOLD LED on until key is released. (b) LOOP LED (l-4) winks at 30 IPM. (c) START/STOP LED on. (d) POS RLSE LED off. (e) Fixed display off. (f) Trunk identification number off. (g) Trunk placed on hold and music is heard (if customer-provided).
4	Depress winking loop pushbutton before time out occurs.	(a) Music, if provided in Step 3, ceases. (b) Transmission established between trunk and Attendant Console. (c) Attendant Console display reverts to results (a) through (d) of Step 2.
5	Depress HOLD pushbutton.	Results are the same as in Step 3.
6	Allow time out to occur.	(a) Ringing heard. (b) LOOP LED flashes at 60 IPM.
7	Depress flashing LED LOOP pushbutton.	Results are same as Step 2.

STEP	OPERATION	RESULT
8	Depress POS RLSE pushbutton.	(a) LOOP LED off. (b) POS RLSE LED off. (c) START/STOP LED off. (cl) EXCL SRCE or EXCL DEST LED off if on. (e) Fixed display cleared. (f) Trunk and station identification cleared from variable display. (g) TOD displayed.

4.9 Extending a Call

4.9.1 Extension of Trunk-to-Trunk Calls

STEP	OPERATION	RESULT
1	Incoming CO trunk call having disconnect supervision received by Attendant Console.	(a) Ringing heard. (b) LOOP LED (l-4) flashes at 120 IPM.
2	Depress flashing LED LOOP pushbutton (l-4).	(a) LOOP LED on. (b) "LOCAL" appears in display field. (c) Trunk number appears in variable display field. (d) START/STOP LED on. (e) POS RLSE LED on. (f) Ringing stops. (g) Transmission established between trunk and Attendant Console.
3	Dial trunk group access code.	(a) EXCL SRCE LED on. (b) Fixed display cleared. (c) Trunk placed on hold in exclude-source state. (d) Attendant Console receives CO dial tone and, optionally, tick tone heard.
4	Dial number desired.	(a) CO ringback tone heard. (b) Digits appear on fixed display.
5	Call answered at outside station.	(a) Ringback tone stops. (b) Private consultation established between Attendant Console and second trunk.

STEP	OPERATION	RESULT
6	Depress POS RLSE pushbutton,	(a) POS RLSE LED off. (b) START/STOP LED off. (c) LOOP LED off. (d) Transmission established between trunks.

4.9.2 **Extension of Information Calls (Option 1)**

STEP	OPERATION	RESULT
1	Initiate station-to-attendant call.	(a) Station hears ringback tone. (b) Ringing heard. (c) LOOP LED flashes at 30 IPM.
2	Depress flashing LED LOOP pushbutton (1-4).	(a) Ringback tone stops at station. (b) Ringing stops. (c) LOOP LED on. (d) START/STOP LED on. (e) POS RLSE LED on. (f) Originating station number and classes of service displayed. (g) "INFORMATION" appears in fixed display field. (h) Transmission established between the station and Attendant Console.
3	Dial trunk group access code.	(a) EXCL SRCE LED on. (b) Station placed on hold. (c) Fixed display cleared and digits appear on the fixed display. (d) CO dial tone and, optionally, tick tone heard. (e) Digits appear on fixed display.
4	Dial remaining digits.	(a) CO ringback tone heard. (b) Called station rings.
5	Depress POS RLSE pushbutton.	(a) EXCL SRCE LED off. (b) POS RLSE LED off. (c) LOOP LED off. (d) START/STOP LED off. (e) Station extended to the CO ringback tone. (f) TOD displayed.

4.9.3 Extension of Information Calls (Option 2)

STEP	OPERATION	RESULT
1	Station-to-attendant call initiated.	(a) Station hears ringback tone. (b) Ringing heard. (c) LOOP LED (l-4) flashes at 30 IPM.
2	Depress flashing LED LOOP pushbutton (1-4).	(a) Ringback tone stops. (b) Ringing stops. (c) LOOP LED on. (d) START/STOP LED on. (e) POS RLSE LED on. (f) Originating station number and classes of service displayed. (g) "INFORMATION" appears in fixed display field. (h) Transmission established between station and Attendant Console.
3	Dial trunk group access code. (Refer to data base program.)	(a) EXCL SRCE LED on. (b) Station placed on hold. (c) Fixed display cleared. (d) CO dial tone and, optionally, tick tone heard. (e) Digits appear on fixed display.
4	Depress POS RLSE pushbutton.	(a) POS RLSE LED off. (b) LOOP LED off. (c) Station extended to CO dial tone and tick tone (64-ms of busy tone every 2 seconds if office data base is marked for tick tone). (d) TOD displayed.

4.9.4 Call Transfer to IVMS

STEP	OPERATION	RESULT
1	When outside party or another station is on the Console line, dial IVMS access code.	(a) EXCL SRCE LED on. (b) Calling party placed on hold. (c) Ringback or busy tone heard. (d) IVMS is being called.
2	Depress POS RLSE pushbutton.	(a) All Console lamps go off. (b) Calling party is connected to IVMS.

4.10 Announcing Call to Station

4.10.1 Excluding Calling Party When Extending Call

STEP	OPERATION	RESULT
1	Incoming call received by Attendant Console.	(a) Ringing heard. (b) LOOP LED (l-4) flashes.
2	Depress LOOP pushbutton associated with flashing LOOP LED.	(a) LOOP LED on. (b) Source appears in display. (c) POS RLSE LED on. (d) START/STOP LED on. (e) Ringing stops. (f) Transmission established between trunk and Attendant Console.
3	Dial station number.	(a) EXCL SRCE LED on. (b) Fixed display cleared. (c) Calling party placed on hold in exclude-source state. (d) Ringback tone heard. (e) Dialed digits appear on fixed display. (f) START/STOP LED off.
4	Remain with call until station answers, and announce call.	Conversation private (two-way), calling party excluded.
5	Depress POS RLSE pushbutton to release call.	(a) EXCL SRCE, POS RLSE and LOOP LEDs off. (b) Trunk-and-station-number display off. (c) Type-of-call display off. (d) Called station and calling party connected. (e) TOD displayed.

4.10.2 Including Calling Party When Extending Call (Three-Party Conference)

STEP	OPERATION	RESULT
1	Incoming call received by Attendant Console.	(a) Ringing heard. (b) LOOP LED (l-4) flashes.
2	Depress LOOP pushbutton associated with flashing LOOP LED.	(a) LOOP LED on. (b) Source appears in display. (c) POS RLSE LED on. (d) START/STOP LED on.

STEP	OPERATION	RESULT
		(e) Ringing stops. (f) Transmission established between trunk and Attendant Console.
3	Dial station number	(a) EXCL SRCE LED on. (b) Fixed display cleared. (c) Calling party placed on hold in exclude-source state. (d) Ringback tone heard. (e) Dialed digits appear on fixed display. (f) START/STOP LED off.
4	Remain with call until station answers, and announce call.	Conversation private (two-way calling party excluded).
5	Depress EXCL SRCE pushbutton.	(a) EXCL SRCE LED off. (b) Attendant, calling party and called station connected in three-party conversation.
6	Depress POS RLSE pushbutton to release call.	(a) EXCL SRCE, POS RLSE, and LOOP LEDs off. (b) Trunk-and-station number display off (c) Type-of-call display off. (d) Called station and calling party connected. (e) TOD displayed.

4.10.3 Alternate Exclusion of Calling Party or Called Party When Extending Call

STEP	OPERATION	RESULT
1	Incoming call received by Attendant Console.	(a) Ringing heard. (b) LOOP LED (l-4) flashes.
2	Deeress LOOP pushbutton associated with flashing LOOP LED.	(a) LOOP LED on. (b) Source appears in display. (c) POS RLSE LED on. (d) START/STOP LED on. (e) Ringing stops. (f) Transmission established between trunk and Attendant Console.

STEP	OPERATION	RESULT
3	Dial station number.	(a) EXCL SRCE LED on. (b) Fixed display cleared. (c) Calling party placed on hold in exclude-source state. (d) Ringback tone heard. (e) Fixed display contains dialed digits. (f) START/STOP LED off.
4	Remain with the call until the station answers and announce the call.	(a) Calling party excluded. (b) Conversation private (two-party).
5	Depress EXCL DEST pushbutton.	(a) Called party excluded. (b) EXCL DEST LED on. (c) EXCL SRCE LED off. (d) Conversation private (two-party).
6	Depress EXCL SRCE pushbutton.	(a) Calling party excluded. (b) EXCL SRCE LED on. (c) EXCL DEST LED off. (d) Conversation private (two-party).
NOTE:	Alternate exclusion of calling party or called party can be continued by performing Steps 5 and 6 above. If three-way conversation is desired, perform Step 7 operation, otherwise, go to Step 8.	
7	Depress either the EXCL SRCE or EXCL DEST pushbutton that has its associated LED on.	(a) Associated LED off. (b) Three-way conference established.
NOTE:	Either Step 5 or 6 position may again be performed to exclude either the calling or called party.	
8	Depress POS RLSE pushbutton.	(a) EXCL SRCE, EXCL DEST, POS RLSE and LOOP LEDs off. (b) Display clears. (c) Called and calling parties connected. (d) TOD displayed.

4.11 Called Station Does Not Answer

4.11.1 Attendant Stays With Calling Party When Extending a Call

STEP	OPERATION	RESULT
1	Incoming call received by Attendant Console.	(a) Ringing heard. (b) LOOP LED (l-4) flashes.
2	Depress LOOP. pushbutton associated with the flashing LOOP LED.	(a) LOOP LED on. (b) Source appears in display. (c) POS RLSE LED on. (d) START/STOP LED on. (e) Ringing stops. (f) Transmission established between trunk and Attendant Console.
3	Dial station number.	(a) EXCL SRCE LED on. (b) Fixed display cleared. (c) Calling party placed on hold in the exclude-source state. (d) Ringback tone heard. (e) Dialed digits appear on fixed display. (f) START/STOP LED off. (g) Called station does not answer.
4	Depress DEST RLS pushbutton.	(a) Ringback tone stops. (b) START/STOP LED on.
5	Depress lighted EXCL SRCE LED pushbutton.	EXCL SRCE LED off.
6	Converse with calling party to determine if additional assistance is required.	

4.11.2 Automatic Recall From Calling Party When Extending Call

STEP	OPERATION	RESULT
1	Incoming call is received by Attendant Console.	(a) Ringing heard. (b) LOOP LED (l-4) flashes.
2	Depress LOOP pushbutton associated with the flashing LOOP LED.	(a) LOOP LED on. (b) Source appears in display. (c) START/STOP LED on. (d) POS RLSE LED on.

STEP	OPERATION	RESULT
		(e) Ringing heard. (f) Transmission established between trunk and Attendant Console.
3	Dial station number.	(a) EXCL SRCE LED on. (b) Fixed display cleared. (c) Calling party placed on hold in exclude-source state. (d) Ringback tone heard. (e) Dialed digits appear on fixed display. (f) START/STOP LED off. (g) Called station does not answer
4	Depress POS RLSE pushbutton.	(a) POS RLSE LED off. (b) LOOP LED off. (c) Calling party extended to station and hears ringback tone. (d) TOD displayed. (e) Called station does not answer after a predetermined amount of time. (f) LOOP LED flashes at 60 IPM. (g) Ringing heard.
5	Depress LOOP pushbutton associated with the flashing LOOP LED.	(a) LOOP LED on. (b) NO ANSW appears in display. (c) START/STOP LED on. (d) POS RLSE LED on. (e) Ringing stops. (f) Called station number and COS displayed. (g) Transmission established between trunk and Attendant Console.
6	Converse with calling party to determine what additional assistance is required.	

4.11.3 Attendant Consults With Calling Party When Extending Call to a Busy Station

STEP	OPERATION	RESULT
1	Incoming call received by Attendant Console.	(a) Ringing heard. (b) LOOP LED (l-4) flashes.
2	Depress LOOP pushbutton associated with flashing LOOP LED.	(a) LOOP LED on. (b) Source appears in display. (c) START/STOP LED on. (d) POS RLSE LED on. (e) Ringing stops. (f) Transmission established between trunk and Attendant Console.
3	Dial station number.	(a) EXCL SRCE LED on. (b) Fixed display cleared. (c) Calling party placed on hold in exclude-source state. (d) Dialed digits appear on fixed display. (e) START/STOP LED off. (f) Busy tone heard.
4	Depress DEST RLS pushbutton.	(a) Dialed digits display is cleared. (b) Called station connection broken, (c) START/STOP LED on. (d) Busy tone ceases.
5	Depress EXCL SRCE pushbutton.	(a) EXCL SRCE LED off. (b) Transmission established between attendant and calling party.

NOTE: The connection to the called station is maintained, allowing automatic camp-on by depressing the CAMP-ON pushbutton. If the calling party wishes to be connected to another station, proceed with Step 6.

STEP	OPERATION	RESULT
6	Converse with calling party to determine what additional assistance is required. If calling party wishes to be connected to another station, proceed with Step 7. If no further assistance is required and the calling party wishes to cancel call, go to Step 8.	
	Dial station number.	(a) EXCL SRCE LED on. (b) Fixed display cleared. (c) Calling party placed on hold in exclude-source state. (d) Ringback tone heard.

STEP	OPERATION	RESULT
		(e) Dialed digits appear on the display. (f) START/STOP LED off.
8	Depress POS RLSE pushbutton.	(a) POS RLSE LED off. (b) LOOP LED off. (c) EXCL SRCE off. (d) Display cleared. (e) TOD displayed. (f) Calling party hears ringback.

4.12 Break-In

The attendant exclusion and attendant break-in features are mutually exclusive. The attendant exclusion feature denies the attendant break-in capabilities by making the BRK IN pushbutton function inoperative. If this feature is used, the attendant can enter a station-to-trunk connection only if recalled by the station. Attendant exclusion is sometimes called attendant lockout.

NOTE: Break-in is not possible if the system has the attendant-exclusion feature activated, if three-way calling is in progress, or if the station has the data-line security feature. A break-in denied message is displayed in these cases.

4.12.1 Break-h on Busy Station

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (l-4).	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial number of station known to be busy in a two-way call.	(a) 60-IPM busy tone heard. (b) Dialed digits appear on fixed display. (c) START/STOP LED off.
3	Depress and hold BRK IN pushbutton.	(a) BRK IN LED on. (b) 60-IPM busy tone heard. (c) A one-second burst of tone heard by attendant and both parties.

STEP	OPERATION	RESULT
4	Allow warning tone time out to occur.	(a) Warning tone to attendant ceases. (b) Transmission established between attendant and both stations (three-way conversation).
5	Release BRK IN pushbutton.	(a) BRK IN LED off. (b) Original two-way conversation continues. (c) 60-IPM busy tone heard in console headset.
6	Depress POS RLSE pushbutton.	(a) LOOP LED off. (b) POS RLSE LED off. (c) EXCL SRCE or EXCL DEST LED off if on. (d) Fixed display cleared. (e) Trunk and station identification cleared from variable display. (f) Class-of-service digits cleared if present. (g) TOD displayed. (h) 60-IPM busy tone ceases.

4.12.2 Break-h Not Allowed Condition

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (l-4) where break-in is not equipped.	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial number of a station known to be busy in a two-way call.	(a) 60-IPM busy tone heard. (b) Dialed digits appear on fixed display. (c) START/STOP LED off.
3	Depress and hold BRK IN pushbutton.	(a) BRK IN LED flutters until released. (b) “BREAK-IN DENIED” appears on display. (c) 60-IPM busy tone heard.
4	Release BRK IN pushbutton.	BRK IN LED off.
5	Depress POS RLSE pushbutton.	(a) LOOP LED off. (b) POS RLSE LED off.

STEP	OPERATION	RESULT
		(c) EXCL SRCE or EXCL DEST LED off if on. (d) Fixed display cleared. (e) Trunk and station identification cleared from variable display. (f) Class-of-service digits cleared if present. (g) TOD displayed.

4.12.3 Break-h on a Broken-In Line Condition

STEP	OPERATION	RESULT
1	Depress an idle LOOP pushbutton (1-4) on Attendant Console 1.	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial number of a station known to be busy in a two-way call.	(a) 60-IPM tone heard. (b) Dialed digits appear on fixed display. (c) START/STOP LED off.
3	Depress and hold BRK IN pushbutton.	(a) BRK IN LED on. (b) 60-IPM tone heard. (c) A one-second burst of tone heard by attendant and both parties. (d) Transmission established between attendant and both stations (three-way conversation).
4	Depress idle LOOP pushbutton (l-4) on second Attendant Console position.	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
5	Dial number of station already being broken into by Attendant Console 1.	(a) 60-IPM busy tone heard. (b) Numbers appear on fixed display. (c) START/STOP LED off.
6	Depress and hold BRK IN pushbutton on Attendant Console 2.	(a) BRK IN LED flutters until released. (b) “BREAK-IN DENIED” appears on display. (c) 60-IPM busy tone heard.

STEP	OPERATION	RESULT
7	Release BRK IN pushbutton on Attendant Console 2.	BRK IN LED off.
8	Depress POS RLSE pushbutton on Attendant Console 2.	(a) LOOP LED off. (b) POS RLSE LED off. (c) EXCL SRCE or EXCL DEST LED off if on. (d) Fixed display cleared. (e) Trunk and station identification cleared from variable display. (f) Class-of-service digits cleared if present. (g) TOD displayed.
9	Release BRK IN pushbutton on Attendant Console 1.	(a) BRK IN LED off. (b) Original two-way connection continued.
10	Depress POS RLSE pushbutton.	(a) LOOP LED off. (b) POS RLSE LED off. (c) EXCL SRCE or EXCL DEST LED off if on. (d) Fixed display cleared. (e) Trunk and station identification cleared from variable display. (f) Class-of-service digits cleared if present. (g) TOD displayed.

4.12.4 Break-in on Data Secure Line Condition

STEP	OPERATION	RESULT
1	Call another station from data-line secure station.	Transmission established between stations.
2	Depress idle LOOP pushbutton (l-4).	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
3	Dial number of data-line secure station used in Step 1.	(a) 60-IPM dial tone heard. (b) Numbers appear on fixed display field. (c) START/STOP LED off.

STEP	OPERATION	RESULT
4	Depress and hold BRK IN pushbutton.	(a) BRK IN LED flutters when released. (b) "BREAK-IN DENIED" appears on display. (c) 60-IPM busy tone heard.
5	Release BRK IN pushbutton	BRK-IN LED off.
6	Depress POS RLSE pushbutton.	(a) LOOP LED off. (b) POS RLSE LED off. (c) EXCL SRCE or EXCL DEST LED off if on. (d) Fixed display cleared. (e) Trunk and station identification cleared from variable display. (f) Class-of-service digits cleared if present. (g) TOD displayed.

4.13 Camp-On

The number of calls that can be camped on to a station is dependent on data base programming. All calls over the camp-on maximum are placed on hold.

If a call is not released from the Attendant Console, the camp-on is denied and the call must be placed on hold, extended to another party, or canceled.

4.13.1 Attendant Camp-On

STEP	OPERATION	RESULT
1	Place incoming CO trunk call to a number.	(a) Ringing heard. (b) LOOP LED flashes at 120 IPM.
2	Depress flashing LED LOOP pushbutton (l-4).	(a) Depressed LOOP LED on. (b) "LOCAL" appears in fixed display field. (c) Trunk number appears in variable display field. (d) START/STOP LED on. (e) POS RLSE LED on. (f) Ringing stops. (g) Transmission established between trunk call and attendant.

STEP	OPERATION	RESULT
3	Dial number of station known to be busy.	(a) EXCL SRCE LED on. (b) Dialed digits displayed in fixed display. (c) START/STOP LED off. (d) 60-IPM busy tone heard.
4	Depress CAMP-ON pushbutton.	(a) POS RLSE LED off. (b) EXCL SRCE LED off. (c) LOOP LED off. (d) CAMP-ON LED on until key reteased. (e) Trunk is camped on busy line. (f) Camped-on station hears burst of camp-on tone. (g) TOD displayed.
5	Allow time out to occur, while keeping station off-hook.	(a) Ringing heard. (b) LOOP LED flashes.
6	Depress flashing LOOP LED pushbutton (l-4).	(a) Ringing stops. (b) LOOP LED on. (c) START/STOP LED on. (d) POS RLSE LED on. (e) "STILL BUSY" appears in fixed display field. (f) Called-station number appears in variable display field. (g) Transmission established between trunk and Attendant Console.
7	Re-establish camp-on condition by keeping station off-hook in a two-way conversation and depressing CAMP-ON pushbutton.	(a) POS RLSE LED off. (b) LOOP LED off. (c) Station hears burst of camp-on tone. (d) TOD displayed.
8	Go on-hook but do not answer camp-on.	(a) After predetermined period of time (data base programmed), ringing heard. (b) LOOP LED flashes at 60 IPM.
9	Depress flashing LED LOOP pushbutton.	(a) Ringing stops. (b) LOOP LED on. (c) START/STOP LED on. (d) POS RLSE LED on. (e) "NO ANSWER" appears in fixed field display. (f) Called-station number appears in variable display field. (g) Transmission established between trunk and attendant.

STEP	OPERATION	RESULT
10	From attendant, make a call to busy station. Depress CAMP-ON pushbutton.	(a) POS RLSE LED off. (b) LOOP LED off. (c) TOD displayed. (d) Station hears burst of camp-on tone.
11	Before the time out occurs, place station on-hook.	**(a)** Trunk receives ringback tone. (b) Camped-on station receives ringing.
12	Answer camp-on call at station.	(a) Ringback tone to trunk stops. (b) Ringing ceases. (c) Transmission established between trunk and station.
13	Place station on-hook.	(a) Station returns to idle state. (b) Trunk returns to idle state.

4.13.2 Attendant Camp-On Data Secure Lines

STEP	OPERATION	RESULT
1	Place incoming CO trunk call to a number.	(a) Ringing heard. (b) LOOP LED (l-4) flashes at 120 IPM.
2	Depress flashing LED LOOP pushbutton (l-4).	(a) Depressed LOOP LED on. (b) "LOCAL" appears in fixed display field. (c) Trunk number appears in variable display field. (d) START/STOP LED on. (e) POS RLSE LED on. (f) Ringing stops. (g) Transmission established between trunk and attendant.
3	Dial number of station that is data secure and known to be busy.	(a) EXCL SRCE LED on. (b) START/STOP LED on. (c) Dialed digits displayed in fixed display (d) START/STOP LED off. (e) 60-IPM busy tone heard.
4	Depress CAMP-ON pushbutton.	(a) POS RLSE LED off. (b) EXCL SRCE LED off. (c) LOOP LED off. (d) Trunk camped on busy line.

STEP	OPERATION	RESULT
		(e) Camped-on station does not hear burst of camp-on tone. (f) TOD displayed.
5	Before time out occurs, place station on-hook.	(a) Trunk receives ringback tone. (b) Camped-on station receives ringing.
6	Answer camp-on call at station.	(a) Ringback tone to trunk stops. (b) Ringing stops. (c) Transmission established between trunk and station.
7	Go on-hook.	(a) Station returns to idle state. (b) Trunk returns to idle state.

4.13.3 Attendant Camp-On Not Equipped

STEP	OPERATION	RESULT
1	Incoming CO trunk call received.	(a) Ringing heard. (b) LOOP LED (l-4) flashes at 120 IPM.
2	Depress flashing LED LOOP pushbutton. (l-4)	(a) Depressed LOOP LED on. (b) "LOCAL" appears in fixed display field. (c) Trunk number appears in variable display field. (d) START/STOP LED on. (e) POS RLSE LED on. (f) Ringing stops. (g) Transmission is established between trunk and attendant.
3	Dial number of station known to be busy.	(a) EXCL SRCE LED on. (b) Dialed digits displayed in fixed display. (c) START/STOP LED off. (d) 60-IPM busy tone heard.
4	Depress the CAMP-ON pushbutton on Attendant Console not equipped with camp-on.	(a) Message indicating that camp-on denied displayed. (b) Camped-on station does not hear burst of camp-on tone. (c) CAMP-ON LED remains on while depressed.

STEP	OPERATION	RESULT
5	Depress POS RLSE pushbutton.	(a) POS RLSE LED off. (b) LOOP LED off. (c) TOD displayed. (d) EXCL SRCE off. (e) Fixed display cleared.

4.14 Answering Station Recall and Transferring to Inside Party

STEP	OPERATION	RESULT
1	Incoming call to Attendant Console has been extended to a station, and station is now recalling by flashing hookswitch and dialing attendant access code.	LOOP LED flashes at 60 IPM.
2	Depress flashing LOOP LED pushbutton to answer recalling party.	(a) LOOP LED on. (b) POS RLSE LED on. (c) Type-of-call display indicates "TRANSFER". (d) Recalling station number displayed. (e) Class-of-service digits displayed. (f) EXCL DEST LED on.

NOTE: If the station releases connection before attendant answers, type-of-call display indicates TRANSFER, and trunk- and station-number display indicates number of trunk or station that originated call.

STEP	OPERATION	RESULT
3	Depress EXCL DEST pushbutton.	(a) Three-way call established. (b) EXCL DEST LED off.
4	Recalling party requests attendant to transfer trunk party to another station and places station on-hook.	Two-way call established.
5	Depress START/STOP pushbutton.	(a) Dial tone heard. (b) START/STOP LED on. (c) EXCL SRCE LED on. (d) Fixed display cleared.
6	Dial requested station number	(a) START/STOP pushbutton LED off. (b) Ringback tone heard. (c) Dialed digits appear on fixed display.

STEP	OPERATION	RESULT
NOTE:	In case of misdial, depress DEST RLS pushbutton and then depress START/STOP pushbutton again. When dial tone heard, dial correct station number.	
7	Depress POS RLSE pushbutton to release call. Attendant does not need to remain with call until called station answers.	(a) LOOP and POS RLSE LEDs off. (b) Trunk-and-station-number display off. (c) Type-of-call display off. (d) Class-of-service digits display off if station or tie-line call. (e) Ringing station and calling party connected. (f) EXCL SRCE LED off. (g) TOD displayed.

4.15 Placing a Call Through the Attendant Console

4.15.1 Outside-to-Inside Call

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (l-4).	(a) LOOP LED on. (b) POS RLSE LED on. (c) Dial tone heard.
2	Dial outside access code.	(a) START/STOP LED on. (b) Second dial tone and tick tone (see note 1) heard.
3	Dial number of outside party.	(a) Ringback and tick tone heard (if the option is used). (b) Dialed digits appear *on* fixed display.
4	Depress START/STOP pushbutton or # sign to remove tick tone.	(a) START/STOP LED off. (b) Tick tone stops.
5	When outside party answers, inform party of call and ask party to wait while call is completed.	
6	Depress START/STOP pushbutton.	(a) Dial tone heard. (b) START/STOP LED on. (c) EXCL SRCE LED on (outside party excluded).
7	Dial number of station.	(a) Receives ringback tone. (b) Dialed digits appear on fixed display.

STEP	OPERATION	RESULT
		(c) START/STOP LED off.
8	inform station user of nature of call when station user answers.	
9	Depress POS RLSE pushbutton to release call.	(a) LOOP, POS RLSE, and EXCL SRCE LEDs off. (b) Fixed display clears. (c) TOD displayed.

NOTES:

1. Provision of tick tone is a data base programmable option and shall be provided together with block transmission data base option.

2. If three-way connection desired, depress EXCL SRCE pushbutton before releasing call (EXCL SRCE LED goes off),

4.15.2 Inside-to-Outside Call

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (l-4).	(a) LOOP LED on. (b) POS RLSE on. (c) Dial tone heard. (d) START/STOP LED on.
2	Dial number of station.	(a) EXCL SRCE LED on. (b) Ringback tone heard. (c) Dialed digits appear on fixed display. (d) START/STOP LED off.
3	When inside party answers, inform party of call and ask party to wait while call is completed.	
4	Depress START/STOP pushbutton.	(a) START/STOP LED on. (b) Dial tone heard. (c) EXCL SRCE LED on (first party excluded).
5	Dial outside access code.	Second dial tone and tick tone heard.

STEP	OPERATION	RESULT
6	Dial directory number of the outside option party.	(a) Ringback tone and tick tone heard (if option provided). (b) Dialed digits appear on fixed display.
7	Depress START/STOP pushbutton.	(a) START/STOP LED off. (b) Tick tone stops.
8	When outside party answers, inform party of nature of call.	
9	Depress POS RLSE pushbutton to release call.	(a) LOOP, POS RLSE and EXCL SRCE LEDs go off. (b) Display clears. (c) TOD displayed.

NOTE: Provision of tick tone is a data base programmable option and shall be provided together with block transmission data base option.

4.15.3 Inside-to-Inside Call

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton.	(a) LOOP LED on. (b) POS RLSE on. (c) Dial tone heard. (d) START/STOP LED on.
2	Dial number of the station.	(a) EXCL SRCE LED on. (b) Ringback tone heard. (c) Dialed digits appear on fixed display (d) START/STOP LED off.
3	When inside party answers, inform party of call and ask party to wait while call is completed.	
4	Depress START/STOP pushbutton,	(a) Dial tone heard. (b) START/STOP LED on. (c) EXCL SRCE LED on (first party excluded).
5	Dial number of station.	(a) Ringback tone heard. (b) Dialed digits appear on fixed display. (c) START/STOP LED off.

STEP	OPERATION	RESULT
6	When station user answers, inform station user of nature of call.	
7	Depress POS RLSE pushbutton to release call.	(a) LOOP, POS RLSE and EXCL SRCE LEDs off. (b) Display clears. (c) TOD displayed.

4.15.4 Outside-to-Outside **Call**

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (l-4).	(a) LOOP LED on. (b) POS RLSE on. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial outside access code.	Second dial tone and tick tone heard.
3	Dial directory number of outside party.	(a) Ringback tone and tick tone heard (if option implemented). (b) Dialed digits appear on fixed display.
4	Depress START/STOP pushbutton or # sign.	(a) START/STOP LED off. (b) Tick tone stops.
5	When outside party answers, inform party of call and ask party to wait while call is completed.	
6	Depress START/STOP pushbutton.	(a) START/STOP LED on. (b) Dial tone heard.
7	Dial outside access code.	Second dial tone and tick tone heard.
8	Dial directory number of outside party.	(a) Ringback tone and tick tone heard (if option implemented). (b) Dialed digits appear on fixed display.
9	Depress START/STOP pushbutton or # sign.	(a) START/STOP LED off. (b) Tick tone stops.
10	When outside party answers, inform party of nature of call.	

STEP	OPERATION	RESULT
11	Depress POS RLSE pushbutton to release call.	(a) LOOP, POS RLSE, and EXCL SRCE LEDs off. (b) Display clears. (c) TOD displayed. (d) Two trunk parties are connected. (e) If neither trunk party has supervision, LOOP LED flashes.

NOTE: Provision of tick tone is a data base programmable option and shall be provided together with block transmission data base option.

4.16 Series Calling

4.16.1 Trunk to Line

STEP	OPERATION	RESULT
1	Incoming CO trunk call placed to Attendant Console.	(a) Ringing heard. (b) LOOP LED flashes at 120 IPM.
2	Depress flashing LOOP LED pushbutton (l-4).	(a) LOOP LED on. (b) "LOCAL" appears in fixed display field. (c) Trunk number appears in variable display field. (d) START/STOP LED on. (e) POS RLSE LED on. (f) Ringing stops. (g) Transmission established between trunk and attendant.
3	Dial number of next station in series call.	(a) Fixed display cleared. (b) EXCL SRCE LED remains on. (c) Trunk placed on hold. (d) Dialed digits displayed in fixed display. (e) START/STOP LED off. (f) 60-IPM ringback tone heard. (g) Called station receives distinctive ringing (1 second on, 1 second off).
4	Depress SER CALL pushbutton.	(a) POS RLSE LED off. (b) EXCL SRCE LED off. (c) LOOP LED off. (d) SER CALL LED momentarily on. (e) Trunk receives ringback tone.

STEP	OPERATION	RESULT
5	Called station goes off-hook.	(a) Ringing removed from station. (b) Ringback tone removed from calling trunk. (c) Transmission established between station and trunk.
6	Place called station on-hook.	(a) Ringing heard. (b) LOOP LED flashes at 60 IPM. (c) Trunk receives ringback tone. (d) Trunk diverted to Attendant Console.
7	Depress flashing LED LOOP pushbutton.	(a) Ringing stops. (b) LOOP LED on. (c) START/STOP LED on. (d) POS RLSE LED on. (e) Trunk number appears in variable display field. (f) "SERIES CALL" appears in fixed display field. (g) Ringback tone removed from trunk. (h) Transmission established between trunk and attendant.
8	Repeat Steps 3 through 7 for at least one more call.	Results are the same as in Steps 3 through 7.
9	Repeat Step 3.	Results are the same as in Step 3.
10	Depress POS RLSE pushbutton.	(a) LOOP, POS RLSE and EXCL SRCE LEDs off. (b) Display clears. (c) Trunk receives ringback tone. (d) TOD displayed. (e) Called station is ringing.
11	Repeat Step 5.	Results are the same as in Step 5.

4.16.2 Line to Trunk and Series Call Release

STEP	OPERATION	RESULT
1	Call attendant from one of the station lines,	(a) Ringing heard. (b) LOOP LED (l-4) flashes at 30 IPM.

STEP	**OPERATION**	**RESULT**
2	Depress flashing LED LOOP pushbutton (l-4).	(a) LOOP LED on. (b) '"INFORMATION" appears in fixed display field. (c) Line number appears in variable display field. (d) START/STOP LED on. (e) POS RLSE LED on. (f) Ringing stops. (g) Transmission established between line and attendant.

NOTE: As an alternate to Steps 1 and 2, the attendant can initiate the call to the station and then proceed with Step 3.

STEP	OPERATION	RESULT
3	Dial number of outgoing trunk group access code followed by trunk number.	(a) Fixed display cleared. (b) EXCL SRCE LED on. (c) Line placed on hold. (d) Dialed digits displayed in fixed display. (e) Ringback tone heard. (f) Called trunk accessed.
4	Depress START/STOP pushbutton if on.	START/STOP LED off.
5	Depress SER CALL pushbutton.	(a) POS RLSE LED off. (b) EXCL SRCE LED off. (c) LOOP LED off. (d) SER CALL LED momentarily on. (e) Station receives ringback tone.
6	Called trunk answers.	(a) Ringing stops. (b) Ringback tone stops. (c) Transmission established between station and trunk.
7	Go on-hook.	(a) Console ringing heard. (b) LOOP LED flashes at 60 IPM. (c) Ringback tone heard. (d) Station remains disconnected.
8	Depress flashing LED LOOP pushbutton.	(a) Console ringing stops. (b) LOOP LED on. (c) POS RLSE LED on. (d) Station number appears in variable display field. (e) "SERIES CALL" flashes in fixed number display field. (f) Ringback tone stops.

STEP	OPERATION	RESULT
9	Depress POS RLSE pushbutton.	(a) LOOP LED off. (b) POS RLSE LED off. (c) EXCL SRCE or EXCL DEST LED off if on. (d) Fixed display cleared. (e) Trunk and station number cleared from variable display. (f) Class-of-service digits cleared if present. (g) TOD displayed.

4.16.3 Line to Line and Series Call Release (Invalid Condition)

STEP	OPERATION	RESULT
1	Attendant called from one of the station lines.	(a) Ringing heard. (b) LOOP LED (l-4) flashes at 30 IPM.
2	Depress flashing LED LOOP pushbutton (l-4).	(a) LOOP LED on. (b) “INFORMATION” appears in fixed display field. (c) Line number appears in variable display field (d) START/STOP LED on. (e) POS RLSE LED on. (f) Ringing stops. (g) Transmission established between line and Attendant Console.
3	Dial number of next call of the series call in fixed display.	(a) Fixed display clears. (b) EXCL SRCE LED remains on. (c) Line placed on hold. (d) Dialed digits displayed. (e) START/STOP LED off. (f) Ringback tone heard. (g) Called line accessed.
4	Depress SER CALL pushbutton.	(a) “SERIES CALL DENIED” displayed. (b) SER CALL LED on while depressed.
5	Depress POS RLSE pushbutton.	(a) LOOP LED off. (b) POS RLSE LED off. (c) EXCL SRCE or EXCL DEST LED off if on. (d) Fixed display cleared.

STEP	OPERATION	RESULT
		(e) Trunk andstation numbers cleared from variable display. (f) Class-of-service digits cleared if present. (g) TOD displayed.

4.17 Extending Information Call on Outside Number, WATS Line, or FX Line

4.17.1 Attendant Completes Call

STEP	OPERATION	RESULT
1	Station user requests outside number.	
2	Dial appropriate access code.	(a) EXCL SRCE LED on. (b) Fixed display cleared. (c) Second dial and tick tone heard. (d) Dialed digits appear on fixed display.
3	Dial outside number.	(a) Ringback and optional tick tone heard. (b) Digits appear on fixed display.
4	Depress START/STOP pushbutton to remove tick tone.	(a) START/STOP LED off. (b) Tick tone stops.
5	Announce call when outside party answers.	
NOTE:	Disconnect can be made as soon as ringback tone heard.	
6	Depress POS RLSE pushbutton to release call.	(a) LOOP LED off. (b) POS RLSE LED off. (c) EXCL SRCE or EXCL DEST LED off if on. (d) Fixed display cleared. (e) Trunk and station number cleared from variable display. (f) Class-of-service digits cleared if present. (g) TOD displayed.

4.17.2 Station User Completes Call

STEP	OPERATION	RESULT
1	Station user requests assistance to reach outside number.	(a) Acknowledge request. (b) inform station user that outside number can be dialed from station upon hearing outside dial tone.

NOTES:

1. If the attendant extended the station to the trunk, the station's class of service is used for toll restriction.
2. If the attendant wants to bypass toll restriction, the entire number must be dialed before extending the trunk to the station.

STEP	OPERATION	RESULT
2	Dial the appropriate access code.	(a) EXCL SRCE LED on. (b) Fixed display cleared. (c) Second dial tone and tick tone heard. (d) Digits appear on fixed display.
3	Depress the POS RLSE pushbutton.	(a) LOOP LED off. (b) POS RLSE LED off. (c) EXCL SRCE or EXCL DEST LED off if on. (d) Fixed display cleared. (e) Trunk and station numbers cleared from variable display. (f) Class-of-service digits cleared if present. (g) TOD displayed.

4.18 Extending information Call Over Tie Line

STEP	OPERATION	RESULT
1	Station user requests tie-line number.	
2	Dial appropriate tie-line access code.	(a) EXCL SRCE LED on. (b) Fixed display cleared. (c) Second dial tone and optional tick tone heard. (b) Digits appear on fixed display.
3	Dial tie-line number of desired party.	(a) Ringback tone heard. (b) Digits appear on fixed display.

STEP	OPERATION	RESULT
4	When tie-line party answers, announce call.	
NOTE:	Disconnect when ringback tone heard.	
5	Depress lighted EXCL SRCE LED pushbutton. .	EXCL SRCE LED off.
6	Depress POS RLSE pushbutton to release from call.	(a) Fixed display cleared. (b) LOOP and POS RLSE LEDs off. (c) TOD displayed. (d) Station and the Tie Line party are talking.

4.19 Attendant Conference

4.19.1 Establish Conference Call for Inside Party

STEP	OPERATION	RESULT
1	Obtain all names and station numbers of conferees.	(a) LOOP LED on. (b) START/STOP LED on. (c) POS RLSE LED on.
2	Depress CONF pushbutton.	(a) CONF LED flashes. (b) EXCL SRCE LED on. (c) Attendant placed in conference.
3	Depress EXCL SRCE pushbutton.	(a) EXCL SRCE LED off. (b) Calling station placed in conference.
4	Depress the flashing CONF LED pushbutton.	(a) Variable display cleared. (b) CONF LED on. (c) Attendant removed from conference.
NOTE:	CONF LED flashing indicates that attendant is in conference bridge.	
5	Dial number of second conferee. If inside party, dial station number. If an outside party, dial access code, receive second dial tone, then dial outside number. If outside party, depress START/STOP pushbutton.	(a) Fixed display cleared. (b) Ringback tone heard. (c) Dialed digits appear on display. (d) START/STOP LED off.
6	Announce conference call when second conferee answers.	

STEP	OPERATION	RESULT
7	Depress CONF pushbutton.	(a) CONF LED flashes. (b) EXCL SRCE LED on. (c) Beak-in tone heard by conferee. (d) Attendant placed in conference.
8	Depress EXCL SRCE pushbutton.	(a) The EXCL SRCE LED off. (b) Called party placed in conference.
9	Depress flashing CONF LED pushbutton.	(a) CONF LED on. (b) Attendant removed from conference.
NOTE:	Trunks that do not guarantee disconnect supervision remain in conference bridge until last station hangs up regardless of when trunk party actually hangs up. All trunks are removed from bridge when last station hangs up. Bridge becomes idle and the CONF LED goes off.	
10	Repeat Steps 5 through 10 until all conferees have been placed in conference bridge.	Variable display cleared.
11	Depress POS RLSE pushbutton to release loop.	(a) LOOP and POS RLSE LEDs off. (b) TOD displayed.

4.19.2 Outside Party Connection

STEP	OPERATION	RESULT
1	Obtain all information concerning conference names and station numbers.	
2	Proceed as directed in Steps 2 through 10 (Paragraph 4.19.1).	

4.20.3 Placing Call in Page Queue

STEP	OPERATION	RESULT
1	Incoming trunk call placed to Attendant Console.	(a) Ringing heard. (b) LOOP LED (l-4) flashes at 120 IPM.
2	Depress flashing LED LOOP pushbutton.	(a) LOOP LED on. (b) “LOCAL” appears in fixed display field. (c) Trunk number appears in variable display field. (d) START/STOP LED on. (e) POS RLSE LED on. (f) Ringing stops. (g) Transmission established between trunk and attendant.
3	Depress PAGE Q pushbutton.	(a) PAGE Q LED on. (b) Trunk placed on hold in page queue. (c) Fixed and variable displays cleared.
4	Depress POS RLSE pushbutton.	(a) POS RLSE and LOOP LEDs off. (b) TOD displayed.
5	Depress and hold the PAGE push-button and page the desired party.	(a) PAGE LED on. (b) Attendant access to paging system allowed.
6	Release PAGE pushbutton.	(a) PAGE LED off. (b) Attendant Console disconnected from paging system.
7	Dial page queue access code from any station.	(a) PAGE Q LED off. (b) Transmission established between trunk and station (paged party).
8	Dial attendant from any station as an alternative to Step 7.	(a) Ringback tone heard. (b) LOOP LED (l-4) flashes at 30 IPM. (c) Ringing heard.
9	Depress flashing LOOP LED pushbutton.	(a) LOOP LED on. (b) START/STOP LED on. (c) POS RLSE LED on. (d) Ringing stops. (e) Page queue party remains. (f) Calling station number and class of service displayed. (g) “INFORMATION” appears in fixed display field. (h) Transmission established between attendant and calling party.

STEP	OPERATION	RESULT
10	Depress PAGE Q pushbutton.	(a) The PAGE Q LED off. (b) Three-way conversation established between attendant, trunk, and station.
11	Depress POS RLSE pushbutton.	(a) The LOOP LED off. (b) POS RLSE LED off. (c) EXCL SRCE or EXCL DEST LED off if on. (d) Fixed display cleared. (e) Trunk and station identification cleared from variable display. (f) Class-of-service digits cleared if present. (g) TOD displayed. (h) Trunk and station are connected.
12	Dial attendant from any station.	(a) Ringing heard. (b) LOOP LED (l-4) flashes at 30 IPM.
13	Depress flashing LOOP LED pushbutton.	(a) LOOP LED on. (b) START/STOP LED off. (c) POS RLSE LED on. (d) Ringing stops. (e) The calling station number and class of service displayed. (f) "INFORMATION" appears in fixed display field. (g) Transmission established between attendant and station.
14	Repeat Steps 3 through 7.	Results same as Steps 3 through 7.

4.20.4 If Paged Party Does Not Answer

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (l-4).	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on.
2	Depress PAGE Q pushbutton.	PAGE Q LED off
3	Proceed as requested by calling party.	

4.20.5 If Paged Party Calls Attendant

STEP	OPERATION	RESULT
1	Depress flashing LED LOOP pushbutton.	(a) LOOP LED on. (b) START/STOP LED on. (c) POS RLSE LED on. (d) Station number displayed in trunk-and-station-number display. (e) The class-of-service digits displayed in class-of-service display. (f) "INFORMATION" displayed in type-of-call display.
2	After answering paged party, depress PAGE Q pushbutton.	(a) PAGE Q LED off. (b) Three-way call established.
3	To release call, depress POS RLSE pushbutton.	(a) LOOP and POS RLSE LEDs off. (b) START/STOP LED off. (c) Trunk-and-station number display off. (d) Class-of-services display off. (e) Type-of-call display off. (f) TOD displayed.

NOTE: If paged party dials page-answer code, paged party is automatically connected to party in page queue. The PAGE Q LED goes off.

4.20.6 Attendant Denied Placing Call In Page Queue Condition

STEP	OPERATION	RESULT
1	Incoming trunk call placed to an attendant that has PAGE Q equipped.	(a) Ringing heard. (b) LOOP LED (l-4) flashes at 120 IPM.
2	Depress flashing LOOP LED pushbutton.	(a) LOOP LED on. (b) "LOCAL" appears in fixed display field. (c) Trunk number appears in variable display field. (d) START/STOP LED on. (e) POS RLSE LED on. (f) Ringing stops. (g) Transmission established between trunk and attendant.

STEP	OPERATION	RESULT
3	Depress PAGE Q pushbutton.	(a) PAGE Q access denied. (b) Appropriate message displayed indicating failure.
4	Depress START/STOP pushbutton.	START/STOP LED off.
5	Incoming trunk call placed to an attendant that does not have PAGE Q equipped.	(a) Ringing heard. (b) LOOP LED (l-4) **flashes at 120 IPM.**
6	Depress flashing LED LOOP pushbutton.	(a) LOOP LED on. (b) "LOCAL" appears in fixed display field. (c) Trunk number appears in variable display field. (d) POS RLSE LED on. (e) Ringing stops. (f) Transmission established between trunk and attendant.
7	Depress PAGE Q pushbutton.	(a) Page Queue access denied. (b) Appropriate message displayed indicating failure.

4.21 Code Calling

4.21.1 Attendant Code Calling

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (1-4)	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial the following information: (a) Code call access code. (b) Zone-paging digit (if applicable). (c) One-, two-, or three-digit code of the called party.	(a) PAGE LED on. (b) Fixed display shows dialed digits.
3	Remain with call until called party answers.	Coded digits are heard by attendant and over loudspeakers.
4	Deliver message when called party answers.	

4.19.3 Station on Outside Party Requests Inclusion in Existing Conference

STEP	OPERATION	RESULT
1	If calling party requests entry in existing conference, depress the lighted CONF LED pushbutton.	(a) CONF LED flashes. (b) EXCL SRCE LED on. (c) Attendant placed in conference.
2	Depress EXCL. SRCE LED pushbutton if conferees are allowed entry.	(a) EXCL SRCE LED off. (b) Calling party placed in conference.
3	Depress flashing CONF LED pushbutton.	(a) Fixed and variable displays cleared. (b) CONF LED on. (c) Attendant removed from conference.
4	Depress POS RLSE pushbutton to release loop.	(a) LOOP and POS RLSE LEDs off. (b) TOD displayed.
5	Depress flashing CONF LED pushbutton if conferees are denied entry.	(a) CONF LED on. (b) Attendant removed from conference.
6	Depress EXCL SRCE pushbutton.	(a) EXCL SRCE LED off. (b) Attendant and calling party connected.
7	Inform calling party of reason for entry denial. Provide assistance as required and depress POS RLSE pushbutton.	(a) LOOP and POS RLSE LEDs off. (b) Trunk-and-station-number display off. (c) Type-of-call display goes off. (d) If station call or tie-line call, class-of-service digits display off. (e) TOD displayed.

4.19.4 Accessing Conference Bridge 2

STEP	OPERATION	RESULT
1	Conference bridge 1 is busy (indicated by CONF LED on).	
2	Depress idle LOOP pushbutton (1-4).	(a) LOOP LED on. (b) POS RLSE LED on. (c) Fixed display cleared. (d) START/STOP LED on. (e) Dial tone heard.

STEP	OPERATION	RESULT
3	Dial conference bridge 2 access code.	(a) Fixed display cleared. (b) If conference bridge 2 is idle, CONF LED off. (c) If conference bridge 2 is busy, CONF LED remains on. (d) Dialed digits appear on fixed display.

4.20 Paging

4.20.1 Universal Paging

STEP	OPERATION	RESULT
1	Depress and hold PAGE pushbutton for duration of page or message.	(a) PAGE LED on. (b) Message or page heard over loudspeakers.
2	Release PAGE pushbutton.	(a) PAGE LED off. (b) Attendant disconnected from paging circuit.

4.20.2 Zone Paging

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (l-4).	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial access paging code, followed by zone-paging digit.	(a) PAGE LED on. (b) Dialed digits appear on fixed display. (c) START/STOP LED off.
3	Page or provide authorized message.	
4	Depress POS RLSE pushbutton.	(a) Fixed display cleared. (b) LOOP, POS RLSE, and PAGE LEDs off. (c) TOD displayed.

STEP	OPERATION	RESULT
NOTE:	If called party does not answer within reasonable time, release call as described in Step 5.	
5	Depress POS RLSE pushbutton to release call.	(a) Fixed display clears. (b) PAGE, LOOP, and POS RLSE LEDs off. (c) TOD displayed.

4.21.2 Code Calling With Outside Party Waiting

STEP	OPERATION	RESULT
1	Inform outside caller of intent to page inside party.	
2	Dial the following: (a) Code call access code. (b) Zone-paging digit (if applicable). (c) One-, two-, or three-digit code of called party.	(a) EXCL SRCE LED on. (b) Fixed display clears. (c) PAGE LED on. (d) Dialed digits appear in fixed display.
3	Remain with call until called party answers.	Coded digits heard over loudspeakers.
4	Announce the call when the called party answers. Conversation is private between the attendant and the announced party.	
NOTE:	If called party does not answer, depress EXCL SRCE pushbutton and inform party.	
5	When conversation with calling party desired, depress EXCL SRCE pushbutton.	(a) EXCL SRCE LED off. (b) Three-way call established.
6	Depress POS RLSE pushbutton to release calls.	(a) PAGE, LOOP, and POS RLSE LEDs off. (b) Trunk-and-station-number display goes off. (c) Type-of-call display goes off. (d) If a tie-line call, class-of-service digits display off. (e) TOD displayed.

4.22 Attendant Trunk Group Restriction

STEP	OPERATION	RESULT
1	Depress an idle LOOP pushbutton (l-4).	(a) LOOP LED on. (b) POS RLSE LED on. (c) STOP/START LED on. (d) Dial tone heard.
2	Dial activation access code followed by number of trunk group to be restricted.	(a) Restriction LED of trunk group indicates in trunk group-status display of the BLDU. (b) 60-IPM tone heard. (c) Dialed digits appear on fixed display. (d) START/STOP LED off.

NOTE: Restriction is removed by attendant dialing a cancellation code followed by trunk group number.

4.23 Attendant Direct Trunk Access

4.23.1 Attendant Access

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (l-4).	(a) Depressed LOOP LED and POS RLSE LEDs on. (b) Fixed display cleared. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial direct trunk access code plus trunk number.	(a) Dialed digits appear on fixed display field. (b) Transmission established between Attendant Console and trunk.
3	Depress POS RLSE pushbutton.	(a) Trunk connection released. (b) LOOP LED off. (c) POS RLSE LED off. (d) EXCL SRCE or EXCL DEST LED off if on. (e) Fixed display cleared. (f) Trunk and station identification cleared from variable display. (g) Class-of-service digits cleared if present. (h) TOD displayed.

STEP	OPERATION	RESULT
4	Repeat Steps 1 through 3 for all trunks in use.	Results are same as in Steps 1 through 3.

NOTE: If the trunk does not have answer supervision and if an MDR call record is desired, the attendant must depress the START/STOP key after all digits have been dialed.

4.23.2 Attendant Control of Facility (ACOF)

STEP	OPERATION	RESULT
1	Depress an idle LOOP pushbutton (l-4).	(a) Depressed LOOP LED and POS RLSE LEDs on. (b) Fixed display cleared. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial the trunk group block on access code and the trunk group number (00 through 63) to block a specific trunk group.	(a) Trunk group blocked. (b) Confirmation tone heard. (c) Dialed digits appear on fixed display. (d) START/STOP LED off.
3	Depress POS RLSE pushbutton.	(a) LOOP and POS RLSE LEDs off. (b) TOD displayed. (c) Fixed display cleared.
4	Repeat Steps 1 through 3 to block additional trunk groups.	Results are same as Steps 1 through 3.
5	Access code of a block trunk group dialed from an unrestricted station.	Station receives 120-IPM busy tone or is diverted to attendant or recorded announcement (data base dependent).
6	Depress an idle LOOP pushbutton (l-4).	(a) LOOP LED and POS RLSE LEDs on. (b) START/STOP LED on.
7	Dial the trunk group block “off” access code followed by trunk group number (00 through 63) to unblock a trunk group.	(a) Dialed digits appear on fixed display. (b) START/STOP, LED off. (c) Trunk group no longer blocked. (d) Confirmation tone heard.
8	Depress POS RLSE pushbutton.	(a) LOOP and POS RLSE LEDs off. (b) TOD displayed. (c) Fixed display cleared.

STEP	OPERATION	RESULT
9	Unblock additional trunk groups by repeating Steps 6 through 8.	Results are same as Steps 6 through 8.
10	Originate station call to a trunk in trunk group taken off block in Step 7.	Station call completed.

4.24 Trunk Busy Verification and Break-in

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (1-4).	(a) LOOP LED on. (b) POS RLSE LED on. (c) Dial tone heard. (d) START/STOP LED on. (e) Fixed display cleared.
2	Dial trunk-busy verification code.	Dialed digits appear on fixed display.
3	Dial number of a trunk known to be idle.	(a) 60-IPM busy tone heard. (b) Dialed digits appear on fixed display.
4	Dial trunk-busy verification code.	Dialed digits appear on fixed display.
5	Dial number of the trunk known to be busy.	(a) 120-IPM busy tone heard. (b) Dialed digits appear on fixed display.
6	Depress and hold BRK IN pushbutton.	(a) BRK IN LED flutters and then goes on. (b) 120-IPM busy tone heard. (c) Called party hears warning tone (dial tone for 1 second).
7	Allow warning tone time out to occur.	(a) Busy tone stops. (b) Transmission established between the attendant and both stations (three-way conversation).
8	Release BRK IN pushbutton.	(a) BRK IN LED off. (b) Two-way connection re-established (see note). (c) 120-IPM busy tone heard.
9	Depress POS RLSE pushbutton.	(a) Display clears. (b) LOOP, POS RLSE and EXCL SRCE or EXCL DEST LEDs off.

STEP	OPERATION	RESULT
		(c) TOD displayed. (d) 120-IPM busy tone ceases.

NOTE: If trunk is engaged in three-way conversation, or if a data-line secure station is involved, the BRK IN LED does not go on after flashing, and a BREAK-IN DENIED message is displayed. If LED trunk is maintenance busy or out of service, BRK IN LED flashes at 120 IPM, a 120-IPM busy tone is received at the Attendant Console, and a BREAK-IN DENIED message is displayed.

4.25 Forced Trunk Release

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton.	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial forced-release access code.	Dialed digits appear on fixed display.
3	Dial number of trunk to be released.	(a) Call on trunk released. (b) START/STOP LED off. (c) Dialed digits appear on fixed display. (d) Confirmation tone heard.
4	Depress START/STOP pushbutton.	(a) START/STOP LED on. (b) Fixed display cleared.
5	Dial trunk-busy verification code.	Dialed digits appear on fixed display.
6	Dial number of the trunk released in Step 3.	(a) Confirmation tone heard. (b) Dialed digits appear on fixed display. (c) START/STOP LED off.
7	Depress POS RLSE pushbutton.	(a) LOOP LED off. (b) POS RLSE LED off. (c) EXCL SRCE or EXCL DEST LED off if on. (d) Fixed display cleared. (e) Trunk and station identification cleared from variable display.

STEP	OPERATION	RESULT
		(f) Class-of-service digits cleared if present. (g) TOD displayed (see Table 4.1 for time code conversion).

4.26 Dynamic FRL Table

4.26.1 Table Update

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (l-4).	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial appropriate access code for FRL assignment map update.	(a) Dial tone ceases. (b) Fixed display contains dialed digits.
3	Dial the new FRL table values.	(a) Dialed digits appear on the fixed display after access code. (b) When all new FRL digits are entered, fixed display clears. (c) Updated dynamic FRL table appears on variable display. (d) Confirmation tone is heard.
4	Depress POS RLSE pushbutton.	(a) LOOP LED off. (b) POS RLSE LED off. (c) Fixed display cleared. (d) Trunk and station identification cleared from variable display. (e) TOD displayed (see Table 4.1 for time code conversion).

4.26.2 Table Display

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (1-4).	(a) LOOP LED on. (b) POS RLSE LED on.

STEP	OPERATION	RESULT
		(c) START/STOP LED on. (d) Dial tone heard.
2	Dial appropriate access code for displaying FRL data.	(a) Dial tone ceases. (b) Fixed display contains dialed digits. (c) When all access code digits are entered, dialed digits display clears. (d) Dynamic FRL table appears on variable display.
3	Depress POS RLSE pushbutton.	(a) LOOP LED off. (b) POS RLSE LED off. (c) Fixed display cleared. (d) Trunk and station identification cleared from variable display. (e) TOD displayed (see Table 4.1 for time code conversion).

4.27 Flexible Night Connections, UNA, PNA

4.27.1 Flexible Night Connections (PNA Directory Number Update)

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton.	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial flexible-night-connection access code, destination number, and night-answer directory number.	(a) START/STOP LED off. (b) Dialed digits appear on fixed display. (c) Confirmation tone heard.
3	Depress POS RLSE pushbutton.	(a) LOOP LED off. (b) POS RLSE LED off. (c) Fixed display cleared.
4	Repeat Steps 1 through 3 for each destination PNA number.	Results are the same as in Steps 1 through 3.

STEP	OPERATION	RESULT
5	Depress NITE ANSW pushbutton and key in 1 or 2 on the Attendant Console that controls night answer (master console). Place any other console to night mode.	(a) Night Answer message displayed in fixed display. (b) Night Answer LED on. (c) STAFFED LED off. System is now in night-answer mode associated with the entered key (see note).

NOTE: Use of UNA and PNA is data base dependent and may be assigned to either Night Answer 1 or 2.

4.27.2 Flexible Night Connections Verification

STEP	OPERATION	RESULT
1	Initiate incoming trunk call.	(a) Call is diverted to night-answer station. (b) Station rings. (c) Trunk receives ringback tone.
2	Answer call at station.	Transmission established between trunk party and station.
3	Repeat Steps 1 and 2 until all night-answer stations have an incoming trunk call.	Results are same as Steps 1 and 2.

NOTE: The data base assigns trunks to the corresponding night answer destination number (O-l 5).

STEP	OPERATION	RESULT
4	Depress NITE ANSW pushbutton.	(a) NITE ANSW LED off. (b) Night Answer message removed from fixed display. (c) STAFFED LED on. Attendant Console is no longer in night-answer mode.

4.27.3 Night Answer 1 Control, UNA Implementation Assumed

STEP	OPERATION	RESULT
1	Depress NITE ANSW pushbutton and key in a 1.	(a) “NIGHT ANSWER 1" displayed in fixed display. (b) NITE ANSW LED on

STEP	OPERATION	RESULT
		(c) STAFFED LED is off. Every station classmarked to receive incoming trunk calls can answer an incoming call by dialing assigned night-answer access code.

NOTE: The N1 position may be a combination of both UNA and PNA for the different trunks.

4.27.4 Night Answer 1 Verification

STEP	OPERATION	RESULT
1	Place incoming trunk call to appropriate number.	(a) Ringback heard. (b) Audible and/or visible night-answer signal received (as provided by customer for UNA).
2	Dial assigned night-answer access code plus the UNA zone.	Transmission established between calling party and station.

4.27.5 Night Answer 2 Control, PNA Implementation Assumed

STEP	OPERATION	RESULT
1	Depress NITE ANSW pushbutton and key in a 2 on the console that controls night answer.	(a) "NIGHT ANSWER 2" displayed on the fixed display. (b) NITE ANSW LED on. Up to 16 PNA stations or station-hunting groups primed for night-answer service. (c) STAFFED LED off.

4.27.6 Night Answer 2 Verification

STEP	OPERATION	RESULT
1	Place incoming trunk call to the appropriate number.	Call diverted to one of PNA stations or station-hunting groups.

STEP	OPERATION	RESULT
2	Answer call at station.	Transmission established between calling party and station.
3	Repeat Steps 1 and 2 for different trunks until all PNA destination numbers have been verified.	Results are same as Steps 1 and 2.

4.28 Group Speed-Calling Update

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (1-4).	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial appropriate access code for updating group speed-calling data.	(a) Dial tone stops. (b) Dialed digits appear on fixed display.
3	Dial group speed-calling list entry number to be modified (000 - 999).	(a) Dial tone heard again, indicating that system is prepared to accept group speed-calling data. (b) Dialed digits appear on fixed display.
4	If group speed-calling list entry is to be deleted, dial a # and the procedure is complete; otherwise, go to Step 5.	Dialed digits appear on fixed display.
5	Dial digit for bypass trunk group access restriction (0 = restrict, 1 = bypass).	(a) Dial tone stops. (b) Dialed digits appear on the fixed display.
6	Dial value for bypass toll restriction (0 = restrict, 1 = bypass).	Dialed digits appear on fixed display.
7	Dial group speed-calling information consisting of 1 to 15 digits.	Dialed digits appear on fixed display.
NOTE:	If a pause or a special character entry is required, dial an *, then dial a 1 for a short pause or a 2 for a long pause (durations determined in the data base) or the * or # character, then continue dialing digits 0 through 9.	
8	Dial a # to indicate procedure complete.	(a) Confirmation tone heard. (b) Dialed digits appear on fixed display.

NOTE: Speed Calling numbers are grouped by data base in 25 different sublists. The access to a particular sublist can be restricted in data base, by group speed calling access class (set of sublists) assigned to the line originating a group speed call.

4.29 Timed-Reminder

4.29.1 Timed-Reminder Activation

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (1-4).	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial timed-reminder access code, station number, and time.	(a) START/STOP LED off. (b) 60-IPM tone heard. If time invalid, 120-IPM tone heard. (c) Dialed digits appear on fixed display.

NOTES:

1. Where station numbers are of three- and four-digit format, four digits must be dialed for all entries. If a station number is of three-digit format, dial a # after the three digits.
2. A five-digit reminder time must be dialed for all entries; the first digit is either 2 (or A) for AM or 7 (or P) for PM, and the remaining four digits are the hour and minute.
3. The time may not exceed 24 hours from the current time.
4. See Table 4.1 for time code conversion.

STEP	OPERATION	RESULT
3	Depress POS RLSE pushbutton.	(a) POS RLSE and LOOP LEDs off. (b) Fixed display off. (c) TOD displayed.

4.29.2 Timed-Reminder Deactivation

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (l-4).	(a) LOOP and POS RLSE LEDs on. (b) START/STOP LED on. (c) Receive dial tone.

STEP	OPERATION	RESULT
2	Dial timed-reminder cancel code and station number.	(a) START/STOP LED off. (b) Dialed digits appear on fixed display.
3	Depress POS RLSE pushbutton.	(a) LOOP and POS RLSE LEDs off. (b) Fixed display goes off. (c) TOD displayed (see Table 4.1 for time code conversion).

4.30 **Message Waiting**

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (l-4).	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
2	Depress MESG WTG pushbutton.	MESG WTG LED on.
3	Dial station directory number. Three digit station number must be preceded by # key.	(a) Fixed display contains dialed digits. (b) If station had message waiting active, MESG WTG LED flashes and MESG WAITING displayed in fixed display, otherwise MESG WTG LED off and NO MESG WTG displayed in fixed display. (c) START/STOP LED off.
4	Depress MESG WTG pushbutton.	Message waiting status changed from Step 3.
5	Depress MESG WTG pushbutton.	Message waiting status changed from Step 4.
6	Depress POS RLSE pushbutton.	(a) MESG WTG LED off. (b) LOOP LED off. (c) POS RLSE LED off. (d) Fixed display field cleared. (e) TOD displayed.

NOTE: If message waiting is activated, the LED associated with the station user telephone continues to flash until the station user dials the message center station number (this can be an attendant). The LED does not flash if the station is ringing or off-hook.

4.31 Most Economical Route Selection (MERS) Active Time Zone Display

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (l-4).	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial access code for active MERS time zone display.	(a) Zone number in use displayed in-calling number display field. (b) Dialed digits appear on fixed display.
3	Dial end-time-display digit (digit 1).	(a) End time for the active time zone displayed in calling-number display field. (b) Confirmation tone heard. (c) START/STOP LED off. (d) Dialed digits appear on fixed display.
4	Depress POS RLSE pushbutton.	(a) POS RLSE and LOOP LEDs off. (b) Variable and fixed displays cleared. (c) TOD displayed. (See Table 4.1 for time code conversion.)

4.32 Most Economical Route Selection (MERS) Time Zone Control

4.32.1 MERS Time Zone Assignment

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (1-4).	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial access code for MERS time zone change.	(a) Letter "L" displayed if no zone was previously entered by attendant. If attendant had entered a zone number, that number is displayed. (b) Dialed digits appear on fixed display.
3	Dial new zone number (l-3).	(a) Beginning time, if any, of previously entered data displayed (Table 4.1). (b) Dialed digits appear on fixed display.

STEP	OPERATION	RESULT
4	Dial first digit of two-digit hours of beginning time data.	(a) End time, if any, of previously entered data displayed. (b) Dialed digit appears on fixed display.
5	Dial remaining beginning-time and end-time data.	(a) Confirmation tone heard. (b) START/STOP LED off. (c) Dialed digits appear on fixed display.
6	Repeat Steps 1 through 5 to verify that entered data is correct.	
7	Depress POS RLSE pushbutton.	(a) POS RLSE and LOOP LEDs off. (b) Fixed and variable displays cleared. (c) TOD displayed.

4.32.2 MERS Time Zone Cancellation

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton.	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial access code for the MERS time zone cancellation.	(a) Confirmation tone heard. (b) The active time zone in use displayed. (c) START/STOP LED off. (d) Dialed digits appear on fixed display.
3	Depress POS RLSE pushbutton.	(a) POS RLSE and LOOP LEDs off. (b) Fixed display cleared. (c) TOD displayed.

4.33 Call Park

STEP	OPERATION	RESULT
1	When conversing with outside party or another station and outside party wishes to speak to another party at another station, dial call-park access code.	(a) EXCL SRCE LED on. (b) Fixed display contains dialed digits.
2	Dial directory number of station to park call.	(a) START/STOP LED off. (b) Original party parked on station. (c) LOOP LED off. (d) POS RLSE LED off. (e) EXCL SRCE or EXCL DEST LED off if on. (f) Fixed display cleared. (g) Trunk and station identification cleared from variable display. (h) Class-of-service digits cleared if present. (i) TOD displayed.
3	Station that call is parked on can still originate or receive other calls.	
4	Contact party whose station call is parked on (this is done by either paging or telephoning the party if that party's location is known).	
5	The party on whose station call is parked answers by dialing call-park answer code and parked-on station number.	
6	Any station can pick up parked call by dialing call-park answer code and parked-on station number.	
7	If call is not picked up within predetermined time (up to 4 minutes), call automatically rings parked-on station. If call is not answered within predetermined time, call is diverted to attendant.	

4.34 Attendant Recall

STEP	OPERATION	RESULT
1	Incoming CO trunk call received at Attendant Console.	(a) Ringing heard. (b) LOOP LED (l-4) flashes at 120 IPM
2	Depress flashing LOOP LED pushbutton.	(a) LOOP LED on. (b) "LOCAL" appears in fixed display field. (c) Trunk number appears in variable display field. (d) POS RLSE LED on. (e) Ringing stops. (f) Transmission established between trunk and attendant.
3	Dial station number.	(a) EXCL SRCE LED on. (b) Incoming trunk placed on hold. (c) Dialed digits displayed in fixed field. (d) START/STOP LED off. (e) Ringback tone heard. (f) Station receives distinctive ringing tone.
4	Call is answered at the station.	(a) Ringing stops. (b) Ringback tone stops. (c) Transmission allowed between station and attendant.
5	Depress POS RLSE pushbutton.	(a) Call released. (b) LOOP, POS RLSE, and EXCL SRCE LEDs off. (c) TOD displayed. (d) The fixed display is cleared. (e) Station and trunk connected.
6	At the station, the hookswitch is flashed and at the Attendant Console all of the LOOP keys are busy.	(a) Distinctive dial tone heard. (b) Trunk placed on hold (excluded from Communications only).
7	Digit 0 dialed at station.	(a) Ringback tone heard at station. (b) CALL WTG LED dialing if first level of call waiting reached and flashes if second level of call waiting reached.
8	Depress POS RLSE on Attendant Console.	(a) Previous call released from Attendant Console. (b) LOOP, POS RLSE, and EXCL SRCE LEDs off. (c) Display cleared. (d) TOD displayed.

STEP	OPERATION	RESULT
		(e) Available LOOP LED flashes. (f) Ringing heard.
9	Depress flashing LED LOOP pushbutton.	(a) LOOP, POS RLSE and START/STOP LEDs on. (b) If station of Step 6 was next in queue, station number, COS, and TRANSFER appear on display. (c) Ringing stops. (d) Transmission established between station and Attendant Console, or as a data base option, a three-way connection established among attendant, station, and trunk.
10	Provide assistance required.	
11	Depress POS RLSE pushbutton.	(a) LOOP, POS RLSE and EXCL SRCE or EXCL DEST LEDs off. (b) Display clears. (c) TOD displayed.

4.35 Attendant Timed Recall

STEP	OPERATION	RESULT
1	Incoming trunk call received at Attendant Console.	(a) Ringing heard. (b) LOOP LED (l-4) flashes at 120 IPM.
2	Depress flashing LOOP LED pushbutton (l-4).	(a) LOOP LED on steady. (b) "LOCAL" appears in fixed display field. (c) Trunk number appears in variable display field. (d) POS RLSE LED on. (e) START/STOP **LED** on. (f) Ringing stops. (g) Transmission established between trunk and attendant.
3	Dial station number into the system.	(a) Incoming trunk placed on hold. (b) EXCL SRCE LED on. (c) Dialed digits displayed on fixed display.

STEP	OPERATION	RESULT
		(d) Ringback tone heard. (e) Called station rings.
4	Depress POS RLSE pushbutton.	(a) Attendant Console released from call. (b) Trunk receives ringback tone. (c) LOOP LED off. (d) POS RLSE LED off. (e) EXCL SRCE or EXCL DEST LED off if on. (f) Fixed display cleared. (g) Trunk and station identification cleared from variable display. (h) Class-of-service digits cleared if present. (i) TOD displayed.
5	Allow time out (data base) to occur.	LOOP LED flashes at 60 IPM.
6	Depress flashing LOOP LED pushbutton.	(a) LOOP LED on steady and POS RLSE on. (b) "NO ANSWER” appears in fixed display field. (c) Called station number and class-of-service marks displayed. (d) Transmission established between trunk and attendant.

4.36 Attendant Access Check of All implemented Stations

STEP	OPERATION	RESULT
1	Depress an idle LOOP pushbutton (1-4).	(a) LOOP LED on. (b) POS RLSE LED on. (c) START/STOP LED on. (d) Dial tone heard.
2	Dial number of station to be tested.	(a) Station receives normal ringing (1 second on, 3 seconds off). (b) Ringback tone heard. (c) Dialed digits appear on fixed display. (d) START/STOP LED off.
3	Take called station off-hook.	Transmission established between attendant and station.

STEP	OPERATION	RESULT
4	Place station on-hook.	(a) Variable display cleared. (b) TOD displayed.
5	Depress POS RLSE pushbutton.	(a) LOOP and POS RLSE LEDs off. (b) Fixed and variable display cleared. (c) TOD displayed.
6	Repeat Steps 1 through 6 for all stations.	

4.37 Attendant Call Splitting

STEP	OPERATION	RESULT
1	Incoming trunk call placed to appropriate number.	(a) Ringing heard. (b) LOOP LED (l-4) flashes at 120 IPM.
2	Depress flashing LED LOOP pushbutton (l-4).	(a) Ringing stops. (b) LOOP LED on. (c) POS RLSE LED on. (d) START/STOP LED on. (e) Trunk number displayed in variable display field. (f) “LOCAL” appears in fixed display field. (g) Transmission established between trunk and attendant.
3	Dial valid three- or four-digit station number.	(a) EXCL SRCE LED on. (b) Trunk placed on hold in exclude-source state. (c) Dialed digits displayed in fixed display. (d) START/STOP LED off. (e) Ringback tone heard. (f) Station receives distinctive ringing.
4	Answer call at station.	(a) Ringing stops at station. (b) Ringback tone stops at Attendant Console. (c) Transmission established between station and attendant.
5	Depress EXCL DEST pushbutton.	(a) EXCL DEST LED on. (b) EXCL SRCE LED off. (c) Station placed on hold in exclude-destination state.

STEP	OPERATION	RESULT
		(d) Transmission established between trunk and attendant.
6	Depress EXCL SRCE pushbutton.	(a) EXCL DEST LED off. (b) EXCL SRCE LED on. (c) Trunk placed on hold in exclude-source state. (d) Transmission established between station and attendant.
7	Depress EXCL SRCE pushbutton.	(a) EXCL SRCE LED off. (b) Transmission established between trunk, attendant and station.
8	Depress POS RLSE pushbutton.	(a) EXCL DEST LED off. (b) POS RLSE LED off. (c) LOOP LED off. (d) Trunk to station connection is maintained. (e) Variable and fixed displays cleared. (f) TOD displayed.

4.38 Priority Calls Directed to the Attendant

STEP	OPERATION	RESULT
1	Access code for priority call is dialed from a station.	(a) Ringing heard. (b) LOOP LED (l-4) flashes at 120 IPM.
2	Depress flashing LOOP LED pushbutton (l-4).	(a) LOOP LED on steady. (b) "PRIORITY CALL" appears in fixed display. (c) Line number and class of service appears in variable display field. (d) POS RLSE LED on. (e) START/STOP LED on. (f) Ringing stops. (g) Transmission established between line and attendant.
3	Depress POS RLSE pushbutton.	(a) LOOP LED off. (b) POS RLSE LED off. (c) EXCL SRCE or EXCL DEST LED off if on. (d) Fixed display cleared.

STEP	OPERATION	RESULT
		(e) Trunk and station identification cleared from variable display. (f) Class-of-service digits cleared if present. (g) TOD displayed.
4	Non-priority calls are placed in call-waiting queue.	CALL WTG LED on when level 1 is exceeded. LED flashes if level 2 is exceeded.
5	From any station, access code for priority call is dialed.	(a) Ringing heard at 60 IPM. (b) CALL WTG LED flashes at 120 IPM.
6	Terminate call in progress.	LOOP LED (l-4) flashes at 120 IPM.
7	Depress flashing LOOP LED pushbutton (l-4).	(a) LOOP LED on. (b) PRIORITY CALL appears in fixed display. (c) Line number and class of service appear in variable display. (d) POS RLSE LED on. (e) START/STOP LED on. (f) Ringing stops. (g) Transmission established between line and attendant.
8	Depress POS RLSE pushbutton.	(a) LOOP LED off. (b) POS RLSE LED off. (c) EXCL SRCE or EXCL DEST LEDs off if on. (d) Fixed display cleared. (e) Trunk and station identification cleared from variable display. (f) Class-of-service digits cleared if present. (g) TOD displayed.

4.39 Attendant Call-Waiting Operation

STEP	OPERATION	RESULT
1	From several stations, Attendant Console dialed. Do not answer any calls.	(a) LOOP LED flashes at 30 IPM. (b) When call-waiting queue reaches first level, CALL WTG LED goes on.

STEP	OPERATION	RESULT
		(c) When call-waiting queue reaches second level, CALL WTG LED flashes and overflow service is alerted (if equipped).
2	Answer calls.	(a) Calls answered in order received (except for priority calls which move to top of queue). (b) When CALL WTG queue is reduced below second level, LED goes on and UNA (if programmed) is canceled. (c) When CALL WTG queue is reduced below level 1, LED goes off.

4.40 Faulty Line

4.40.1 Maintenance Print-out Operation

STEP	OPERATION	RESULT
1	Attendant called from one of station lines.	(a) Ringing heard. (b) LOOP LED (l-4) flashes at 30 IPM.
2	Depress flashing LED LOOP pushbutton (l-4).	(a) LOOP LED on steady. (b) "INFORMATION" appears in fixed display field. (c) Line number appears in variable display field. (d) POS RLSE LED on. (e) Ringing stops. (f) Transmission established between line and attendant.
3	Depress BAD LINE pushbutton.	(a) BAD LINE LED flashes. (b) System records line information on maintenance TTY or CRT. (c) BAD LINE LED on steady.
4	Depress POS RLSE pushbutton.	(a) BAD LINE LED off. (b) LOOP LED off. (c) POS RLSE LED off. (d) TOD displayed.

4.40.2 Invalid Operation

STEP	OPERATION	RESULT
1	Attendant called from one of station lines.	(a) Ringing heard. (b) LOOP LED flashes at 30 IPM.
2	Depress flashing LOOP LED pushbutton (l-4).	(a) LOOP LED on. (b) “INFORMATION” appears in fixed display field. (c) Line number appears in variable display field. (d) POS RLSE LED on. (e) Ringing stops. (f) Transmission established between line and attendant.
3	Depress START/STOP pushbutton.	(a) START/STOP LED on. (b) EXCL SRCE LED on. (c) Dial tone heard. (d) Line placed on hold.
4	Depress BAD LINE pushbutton.	(a) BAD LINE LED flashes. (b) System does NOT record line information on the maintenance TTY or CRT.
5	Depress POS RLSE pushbutton.	(a) BAD LINE LED off. (b) LOOP LED off. (c) POS RLSE LED off. (d) TOD displayed.

4.41 Alarms

4.41.1 System initialization Alarm

STEP	OPERATION	RESULT
1	Bring up Attendant Console for service.	ALARM LED flashes at 60 IPM.
2	Depress ALARM pushbutton.	If present, fault message displayed in variable display field.
3	Repeat Step 2 until ALARM LED ceases to flash.	Results same as Step 2 or LED off.

4.41.2 No Dial Alarm Directed to Attendant

STEP	OPERATION	RESULT
1	Station assigned no dial alarm feature (data base option).	
2	Assigned station left off-hook without dialing.	(a) After time out period, ringing heard. (b) LOOP LED (I-4) flashes at 60 IPM.
3	Depress the flashing LOOP LED pushbutton (I-4).	(a) LOOP LED on steady. (b) “NO DIAL ALARM” displays in fixed display field. (c) POS RLSE LED on. (d) START/STOP LED on. (e) Ringing stops. (f) Transmission established between off-hook station and Attendant Console.

4.42 Removal of Headset During Operation

STEP	OPERATION	RESULT
1	Attendant is called from any equipped station.	(a) Station receives ringback tone. (b) Ringing heard. (c) LOOP LED flashes at 30 IPM.
2	Depress the flashing LOOP LED pushbutton.	(a) LOOP LED on. (b) Station number appears in variable display field. (c) Station class of service displayed in variable display. (d) “INFORMATION” displayed in fixed display field. (e) Transmission established between attendant and station. (f) START/STOP LED on. (g) POS RLSE LED on.
3	Remove the headset from Attendant Console.	(a) STAFFED LED off. (b) All lit LEDs off. (c) Transmission interrupted to station. (d) Attendant Console functions in N1 mode of night answer function.

STEP	OPERATION	RESULT
		(e) Attendant Console available for self-test diagnostics, internal program access.

4.43 **Data Link Recovery Attempts**

STEP	OPERATION	RESULT
1	Attendant Console called from station.	(a) Ringback tone heard. (b) Ringing heard. (c) LOOP LED flashes at 30 IPM.
2	Depress the flashing LOOP LED pushbutton.	(a) LOOP LED on. (b) Station number appears in variable display field. (c) Station class of service displayed in variable display. (d) “INFORMATION” displayed in fixed display field. (e) Transmission established between attendant and station. (f) START/STOP LED on. (g) POS RLSE LED on.
3	Power to Attendant Console turned OFF.	(a) All lit LEDs off. (b) Transmission interrupted to station.
4	Power to the Attendant Console turned ON.	(a) Requests to be initialized. (b) Verify all queued up messages dropped cleanly. (c) Ensure any call stores or digit stores used are returned to idle list. (d) NITE ANSW LED on. (e) “NIGHT ANSWER 1" appears in fixed display. (f) ALARM LED flashes.
5	Depress NITE ANSW pushbutton.	(a) NITE ANSW LED off. (b) Fixed display cleared. (c) STAFFED LED flashes. (d) TOD displayed (within 60 seconds). (e) Call returns to console. LOOP LED flashes.

NOTE: A call placed to the Attendant Console can ring and flash the LOOP LED even though Attendant Console is in night answer mode.

4.44 Attendant Console Internal Program Access

Refer to Section TL-130200-1001 for procedures.

4.45 Initialization and De-Initialization of Attendant Console via Request

STEP	OPERATION	RESULT
1	Place Attendant Console in service via maintenance.	(a) INITIALIZATION LED flutters for approximately 30 seconds. (b) When initialization is finished, the following occur: 1. Initialization LED goes off. 2. ALARM LED flashes.
2	Insert headset.	STAFFED LED on steady.
3	Remove headset.	STAFFED LED off.
4	Place Attendant Console out of service via maintenance request.	(a) Verify Attendant Console out of service. (b) All displays and LEDs off.
5	Return to Step 1 to bring Attendant Console back on line; stop after Step 2.	

4.46 Spare Key Access

STEP	OPERATION	RESULT
1	Depress idle LOOP pushbutton (l-4).	(a) LOOP LED on. (b) POS RLSE LED on.
2	Depress SPARE pushbutton.	No noticeable change should occur to Attendant Console.
3	Depress POS RLSE pushbutton.	(a) LOOP LED off. (b) POS RLSE LED off. (c) Fixed display field cleared. (d) TOD displayed.

STEP	OPERATION	RESULT
4	Repeat Step 2 while performing other tests.	Depressing SPARE pushbutton should not influence the instrument or switch.
NOTE:	Data Base programmable.	

4.47 Attendant Handling of Locked Loop Trunk-to-Trunk Calls

STEP	OPERATION	RESULT
1	Incoming CO trunk call with no disconnect supervision placed to appropriate number.	(a) Ringing heard. (b) LOOP LED (l-4) flashes at 120 IPM.
2	Depress the flashing LOOP LED pushbutton (l-4).	(a) LOOP LED on steady. (b) “LOCAL” appears in fixed display field. (c) Trunk number appears in variable display field. (d) POS RLSE LED on. (e) START/STOP LED on. (f) Ringing stops. (g) Transmission established between trunk and the attendant.
3	Dial trunk group access code for outgoing trunk that does not guarantee disconnect supervision.	(a) EXCL SRCE LED on. (b) Fixed display cleared. (c) Trunk placed on hold in exclude-source state. (d) CO dial tone and, optionally, tick tone heard. (e) Dialed digits appear on fixed display.
4	Dial outside station number.	(a) CO ringback tone heard. (b) Dialed digits appear on fixed display.
5	Answer call at outside station.	(a) Ringback tone stops. (b) Private conversation allowed between attendant and second trunk.
6	Depress POS RLSE pushbutton.	(a) POS RLSE LED off. (b) START/STOP LED off. (c) LOOP LED winks at 30 IPM. (d) EXCL SRCE LED off. (e) Fixed display goes off. (f) Transmission established between trunks.

NOTE: After a period of time, perform trunk busy verification and Break-In on either trunk. If trunks are not busy (reorder tone, quiet, etc.), perform a Forced Trunk Release on either trunk to break connection.

4.48 Busy Lamp Display Unit Access

STEP	OPERATION	RESULT
1	Depress BLDU pushbutton.	Enter Hundreds Group number displayed in fixed display field.
2	Enter **a** 2-digit Hundreds Group number (00-99).	(a) Number entered displayed in fixed display field. (b) If BLDU attached, status of Hundreds Group entered displayed on BLDU. (c) Fixed display is cleared.’ (d) TOD displayed.

NOTES:

1. If any pushbutton on the Attendant Console other than the BLDU pushbutton and the keypad is depressed after Step 1, “DIGIT ENTRY ONLY” is displayed in the Attendant Console fixed display field. When this occurs, enter **a** 2-digit Hundreds Group number and Step 2 results are completed.

2. If the BLDU pushbutton is depressed again after Step 1, the TOD is displayed on the Attendant Console and the BLDU function is canceled.

4.49 Second Chime

STEP	OPERATION	RESULT
1	Station calls Attendant Console.	(a) Ringback tone is heard. (b) Chime is heard. (c) LOOP LED flashes at 30 IPM
2	Depress the flashing LOOP LED pushbutton.	(a) LOOP LED on steady. (b) Station number appears in the variable display field. (c) Station class of service is displayed in the variable display. (d) Information is displayed in the fixed display field. (e) START/STOP and POS RLSE LEDs on. (f) Transmission allowed between attendant and station.

STEP	OPERATION	RESULT
3	Station calls Attendant Console.	(a) Ringback tone is heard. (b) Chime is heard. (c) LOOP LED flashes at 30 IPM.

THIS PAGE INTENTIONALLY LEFT BLANK.

5.0 BUSY LAMP DISPLAY UNIT

5.1 General

The Busy Lamp Display Unit (BLDU) is an optional unit that can be operated as a stand-alone unit or can be positioned on the Attendant Console so that the BLDU display is directly above the Attendant Console display. Both configurations are shown in Figure 5.1. A detailed view of the BLDU showing the LCD line and trunk group fields and the LCD line group selected display is illustrated in Figure 5.2. This view also shows the position of the ten line group selection keys and the small rectangular red LED indicators located immediately below each key. Refer to Table 5.1 for BLDU key functions.

The line and trunk group display fields use LCD elements to block out the appropriate characters. Therefore, if a trunk group is not restricted, the "R" is blocked from view. Alternatively, if all trunks in a trunk group are busy, the "B" is visible. All trunk groups may be presented in the trunk group display field.

The line field display shows the status of a group of 100 lines. Only the last two digits of the station numbers are shown so that any Hundreds Group can be displayed. The seven-segment line group display window shows which Hundreds Group is being displayed in the line field.

Operation of the BLDU is limited to the selection of the line Hundreds Group to be displayed. This is accomplished either by local control at the BLDU or via remote control from the Attendant Console. The line group hundreds keys on the BLDU allow the selection of ten different Hundreds Groups as defined in the data base, and the LED indicates which pushbutton has been operated. The remote control feature allows the attendant to select any Hundreds Group. The line group two-digit display will'show this selection and active LED will be extinguished. Additionally, a power ON/OFF switch is located on the rear of the BLDU.

NOTE: Activation of displays of Hundreds Groups not programmed on one of the BLDU keys can be made only from an Attendant Console associated with the BLDU.

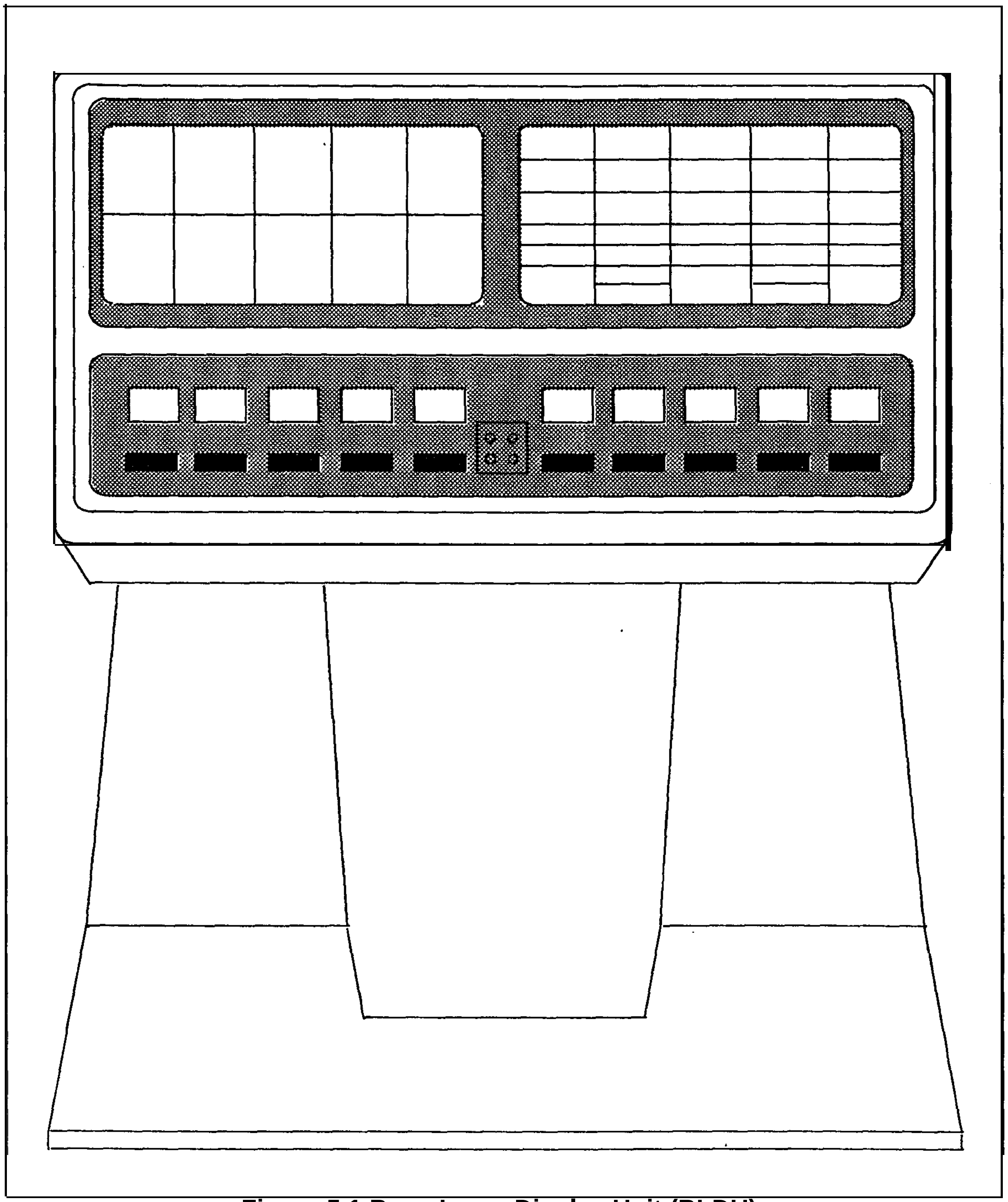


Figure 5.1 Busy Lamp Display Unit (BLDU)

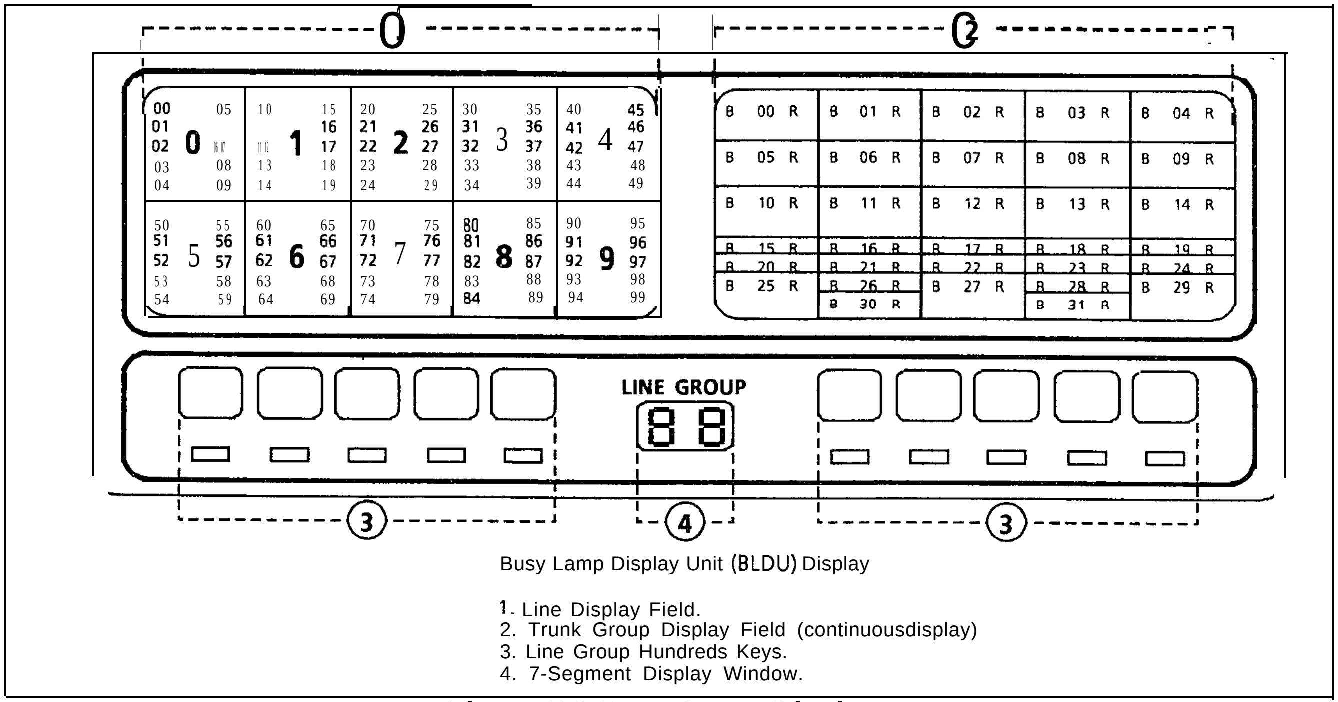


Figure 5.2 Busy Lamp Display

Table 5.1 BLDU Display and Key Function

KEY OR DISPLAY	TYPE	DESCRIPTION
Top-left Field	LCD	Indicates individual station activity.
Top-right Field	LCD	Indicates trunk activity (B = Busy, R = Restricted).
Line Group (seven-segment display)	LCD	Indicates selected 100s group, test status.
Line Group Selection Keys/Indicators	Key & LED	Selects 100s groups. Indicates 100 group selected by a key.
ON/OFF (located on left rear)	Toggle Switch	Controls power supplied to the BLDU.
Fuse Holder and Fuse (located on left rear)	W/3AG, 1.5A Fuse	Provides circuit protection.
Data Link Connections (2) (located on rear)	Ports	Primary data link port in and port out to provide series circuit for additional units up to a total of four.

5.2 BLDU Operating Procedures

5.2.1 BLDU Initialization

STEP	OPERATION	RESULT
1	Turn power ON.	(a) Double dash should be visible on 7-segment display when initialization is occurring. (b) Double dash will be removed and 7-segment display will be blank when initialization complete. (c) Busy or restricted trunk groups will appear in the trunk area of the display.

NOTES:

1. When the BLDU is powered ON, a request for initialization is automatically generated and sequentially completed by the system if normal operating conditions exist. This includes internal operational data plus the status of lines and trunk groups.

2. If the system is idle, the BLDU should complete initialization in approximately 30 seconds. A busy (heavy traffic) system will take longer, but less than 5 minutes.

5.2.2 BLDU Line and Trunk Group Display, Attendant Console Assignment of Hundreds Groups

The trunk group busy display is continuous and cannot be accessed by Attendant Console.

Assignment and change of BLDU ATB/ACOF trunk status and Hundreds Group numbers (other than through the Attendant Console) are accomplished during loading of the data base or through the recent change operation via the TTY or Test Set equipment interface.

(a) From the BLDU

STEP	OPERATION	RESULT
1	Depress line group hundreds key (K1-K10).	(a) Selected line Hundreds Group number appears in 7-segment display window. (b) Active line numbers in selected Hundreds Group light across line display field.

(b) From the Attendant Console

STEP	OPERATION	RESULT
1	Depress BLDU pushbutton on Attendant Console.	ENTER "HUNDREDS GROUP NUMBER" is displayed in fixed display field of Attendant Console.
2	Enter a two-digit Hundreds Group number (00-99).	(a) Number entered is displayed in fixed display field of Attendant Console. (b) If BLDU is attached, status of the Hundreds Group entered will be displayed on the BLDU. (c) Fixed display cleared.

NOTES:

1. If any pushbutton on the Attendant Console, excluding the BLDU pushbutton, and the keypad is depressed after Step 1, "DIGIT ENTRY ONLY" is displayed in the Attendant Console fixed display field. When this occurs, enter a 2-digit Hundreds Group number and Step 2 results are completed.

2. If the BLDU pushbutton is depressed again after Step 1, the time of day (TOD) is displayed on the Attendant Console and the BLDU function is canceled.

5.2.3 Optional BLDU Controls and Indicators Procedures

(a) BLDU Hundreds Group Key Test

STEP	OPERATION	RESULT
1	Depress Hundreds Group keys (K1-K10) across front of BLDU.	Hundreds Group digits will appear in 2-digit, 7-segment display for each programmed group.

(b) BLDU LCDs and LEDs Test

STEP	OPERATION	RESULT
1	Depress last two keys on right of BLDU simultaneously (keys 9 and 10).	All LCDs are lit (no numbers visible in line display field, no Bs or Rs visible in trunk group field, "88" displayed in 7-segment display) and all LEDs are on.

STEP	OPERATION	RESULT
2	Release keys 9 and 10.	All LCDs and LEDs will be extinguished.

(c) BLDU Power-Cycle Operation

STEP	OPERATION	RESULT
1	Turn power ON to OFF, then to ON. BLDU's automatic initialization sequence is engaged.	(a) BLDU clears. (b) BLDU operates normally. (c) If BLDU fails to operate, call maintenance personnel.

6.0 MESSAGE DETAIL RECORDING (MDR)

6.1 General

The MDR feature provides a means to record system call information for outgoing CO, DOD, WATS, Foreign Exchange (FX), and tie lines. MDR will also record call information for incoming CO, DID, WATS, FX, and tie lines. Therefore, MDR may be used as an aid in determining the most efficient and economical use of the system.

For a more detailed description of the MDR function, refer to Sections TL-130000-1001 and TL-130500-1001.

6.2 MDR Output Options

The NSDC Card (FB-20992) can support two independent I/O functions. When MDR is equipped in data base, one port will be dedicated to MDR OUTPUT only. The MDR port will be the port which is not marked as maintenance in data base. (Port 0 is marked maintenance by default so the MDR port will default to port 1.) All system maintenance commands will take place on the maintenance port. The two MDR commands (SET MDR options; DUMP MDR options) will be accessed via the maintenance port only. Transmission data for both ports is set up in Record Code TT. One of two types of MDR OUTPUT can be selected via data base: (1) MDR Call Records can be output in real-time by equipping MDR Port 0 in Data Base, and (2) MDR Call Record Blocks can be output by equipping MDR Port 1 in Data Base.

If MDR Port 0 is implemented in data base, output is provided independently of the condition of the receiving TTY. Header data is output whenever a system start occurs, a change in date or time occurs, or after every 32 double-line call records or 64 single-line call records.

If MDR Port 1 is implemented in data base, then Call Record Blocks will be output to the customer-provided minicomputer or other Remote Polling Devices. Output is in FGBS-modified EBCDIC, which is not suitable for a TTY. Refer to Table 6.1 for a listing of FGBS-modified EBCDIC.

Figure 6.1 and Figure 6.2 illustrate the possible combinations of output from the Narrow Serial Device Controller Card (NSDC). Note that both maintenance and MDR functions are supported in these illustrations.

6.3 Operation

6.3.1 Security Lock

To make changes in the type of MDR output, the user must access a Security Lock. The SI security locks for MDR are the same as those for Recent Change. Refer to Section TL-130200-1001 for a detailed description of these security locks.

Table 6.1 FGBS-Modified EBCDIC

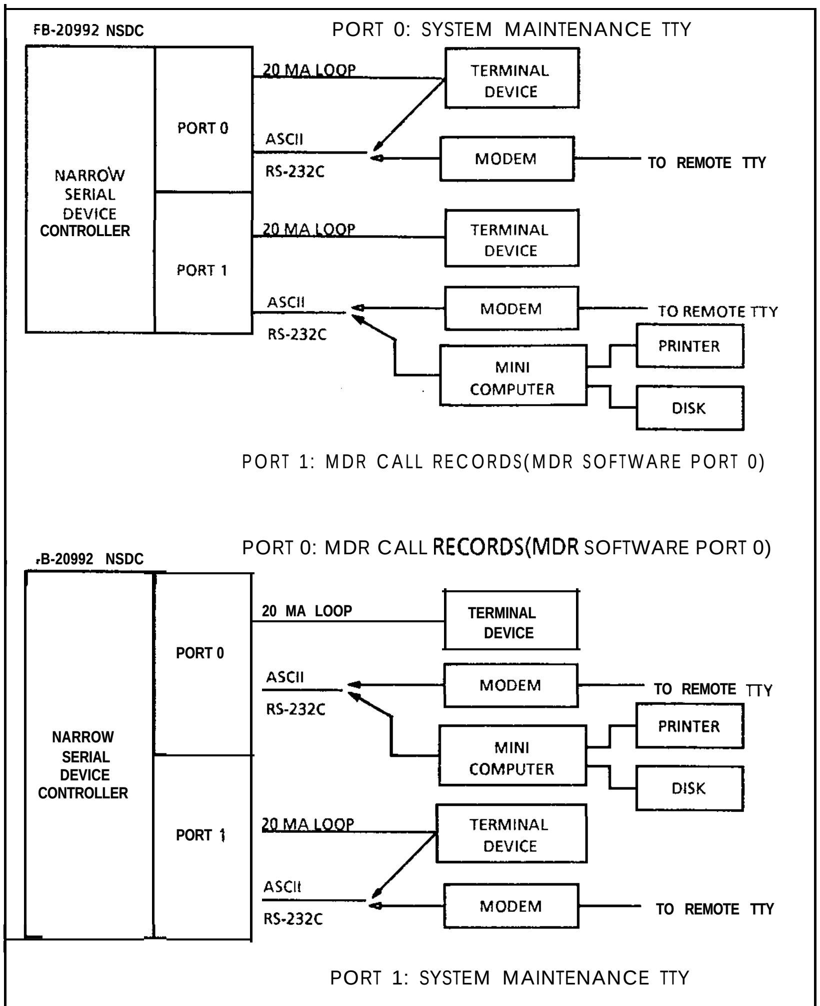
DECIMAL CHARACTER	HEX-ADECIMAL EQUIVALENT	BINARY EQUIVALENT	MDR USE
0	0	0000	The number zero (note 1)
1	1	0001	The number one
2	2	0010	The number two
3	3	0011	The number three
4	4	0100	The number four
5	5	0101	The number five
6	6	0110	The number six
7	7	0111	The number seven
8	8	1000	The number eight
9	9	1001	The number nine
10	A	1010	The number 10 (DIALED DIGIT ZERO) (note 1)
11	B	1011	The number 11 (note 2)
ALPHA (α)/12	C	1100	Identifying character/the number 12 (note 2)
BETA (β)/13	D	1101	Identifying character/the number 13 (note 2)
14	E	1110	The number 14 (note 2)
15, BLANK (b)	F	1111	The number 15, a blank space/identifying character (note 2)

NOTES:

1. The hexadecimal zero is always used for the decimal zero. Hexadecimal A is used as zero in calling and called numbers.

2. Hexadecimals A, B, and E are always used for the decimals 10, 11 and 14, respectively.

   Hexadecimals C, D, and F may have two different uses; however, their meaning is fully defined by the position in the record that they occupy.



'igure 6.1 Possible NSDC Port Assignments with MDR Port 0 (Call Records) Equipped in Data Base

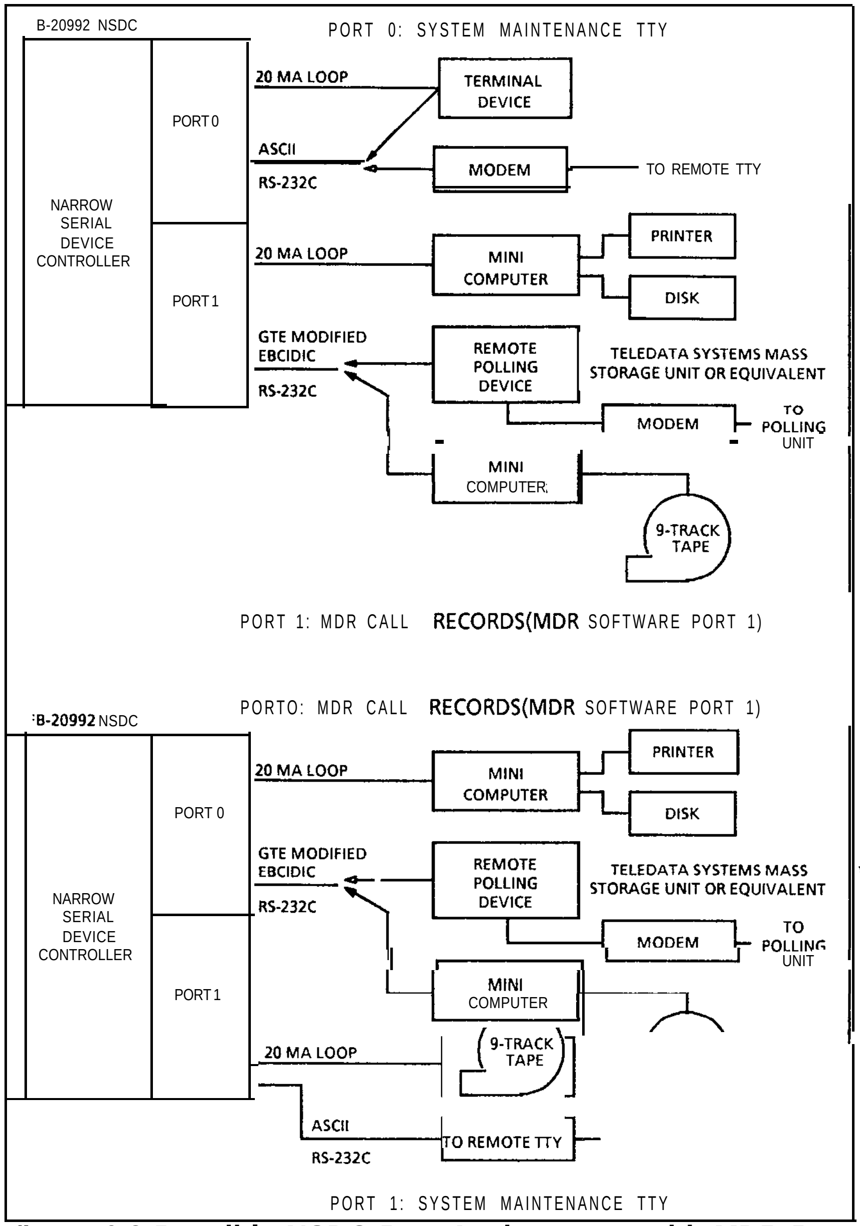


Figure 6.2 Possible NSDC Port Assignments with MDR Port 1 (Call Records) Equipped in Data Base

6.3.2 Time/Date Change

MDR will report the time and date of a call when outputting the call information. The time and date reported are taken from the system Time of Day (TOD) software clock. Refer to Section TL-130200-1001 for instructions to set the TOD.

6.3.3 Set Options Command

The MDR accesses the SET OPTIONS command via the system maintenance terminal. It uses a set of data base variables called screening options to determine which trunk calls are to be recorded. This set of variables consists of the following:

(a) Call duration

(b) Work group number

(c) Outgojng trunk group number

(d) Incoming trunk group number

The set options command is used to change the selection of trunk calls to be recorded by the MDR. When set via the system maintenance TTY, the changes are not permanent unless a Maintenance Request 09 is performed to back up the data base changes on disk.

Two unique sets of screening options exist within the MDR:

(a) Terminal Options. These are used to screen calls that are to be sent to MDR Port 0.

(b) Remote Polling Devices. These are used to screen calls that are to be sent to MDR Port 1.

To enter the SET OPTIONS command, the user types in "SET MDR OPTIONS" or "SE MDR OP" from the maintenance terminal. If MDR Port 0 is equipped in data base, MDR will respond with terminal options. If MDR Port 1 is equipped in data base, MDR will respond with Remote Polling Device.

The prompting formats for both the tape and the terminal are identical. After the MDR responds with REMOTE POLLING DEVICE: or TERMINAL:, it spaces twice and prompts the user with CALL DURATION (O-255 SEC). The call duration is a timing value specified in seconds that is used to determine if a call is to be recorded. Only those calls that exist in a valid two-way conversation for a time period equal to or greater than the call duration value are recorded by the MDR. If a value greater than 255 is entered, the MDR responds with INVALID COMMAND and the set options command is terminated. If the user does not wish to change the present value of the call duration, the present value must be entered.

Once a valid call duration value has been entered, the MDR prompts the user for the work group numbers. A work group number is a number assigned to a uniquely defined set of lines that is used solely by the MDR for screening calls. The MDR will record only those lines that are members of a work group listed in the set options of the data base (Tables T4472 or T4482, Record S1). The maximum number of work groups is 64 (O-63). Member selection for a particular group is data base programmable. However, the set options command does not have the capability of changing line members within a work group. The set options command is only capable of changing the list of work group numbers retained in the data base.

The MDR will request the user to select up to eight work groups at one time as follows: WORK GROUP #S O-7. The user may then enter any number(s) from 0 through 7 in any order followed by a period. Once this is done, the MDR requests the user to select the next set of eight work group numbers as follows: WORK GROUP #S 8-15. Again, the user enters the desired work group numbers within the indicated bounds. If the user does not wish to select any of the work groups within any given eight-group bounds, he can simply enter a period. The MDR then prompts the user for the next eight-group selections. The prompting format continues until all 64 work groups have been presented for user selection.

If a number outside of a designated range is entered, the MDR ignores the out-of-range number and' responds with ENTRY XX DELETED. If the user wishes to enter a new work group number that is in a range of already existing numbers, the existing numbers must be entered along with the new ones. For example, if work group number 10 is to be added, but work group numbers 8 and 15 already exist, the user must enter 8, 10, and 15. If 8 and 15 are not entered along with 10, work group number 10 will be entered, but work group numbers 8 and 15 will be eliminated.

When the work group number selection is completed, the MDR next prompts the user for the outgoing trunk group numbers selection. The MDR uses the outgoing trunk numbers for call-screening purposes. The MDR will record only those outgoing trunks that are members of an outgoing trunk group listed in the screening options portion of the data base.

The prompting format shown as in Figure 6.3 is the same eight-group selection type of format used with the work groups. The same response will result for an out-of-bounds selection. The same procedure as for the addition of a new work group number applies to the addition of a new outgoing trunk group number to a range of existing numbers, i.e., the existing outgoing trunk group numbers must be entered with the new numbers, or the existing numbers will be eliminated from the screening options.

When the outgoing trunk group number selection is completed, the MDR will next prompt the user for the incoming trunk group numbers. The incoming trunk group numbers are identical to the outgoing trunk group numbers in terms of format and usage by the MDR. The only difference is that these numbers are used to screen incoming trunk calls. A format example of the incoming trunk group numbers is shown in Figure 6.4.

In order for a call to be recorded on a given MDR software port, each of the following screening conditions must be satisfied:

(a) The call must involve a trunk in a trunk group which is being recorded by the given port.

(b) If the second party involved in the call is a line or trunk, the associated line work group or trunk group must be marked for recording by the given port.

Set MDR Options.
Remote Polling Device:

Call Duration (O-255 sec) ?	200							
Work Group #S 0 - 7 ?	0	3	7	2	4	5	1	6.
Work Group #S 8 - 15 ?	8	9	10	11	12	13	14	15.
Work Group #S 16 - 23 ?	16	17	18	19	20	21	22	23.
Work Group #S 24 - 31 ?	24	25	26	27	28	29	30	31.
Work Group #S 32 - 39 ?	32	33	34	35	36	37	38	39.
Work Group #S 40 - 47 ?	40	41	42	43	44	45	46	47.
Work Group #S 48 - 55 ?	48	49	50	51	52	53	54	55.
Work Group #S 56 - 63 ?	56	57	58	59	60	61	62	63.
Outgoing Trk Grp #S 0 - 7 ?	0	2	4	6	7	5	3	1.
Outgoing Trk Grp #S 8 - 15 ?	10	12.						
Outgoing Trk Grp #S 16 - 23 ?	16	19	23.					
Outgoing Trk Grp #S 24 - 31 ?	24	26.						

Figure 6.3 Prompting Format for Outgoing Trunk Group Number Selection

Set MDR Options.
Remote Polling Device:

Call Duration (O-255 sec) ?	200							
Work Group #S 0 - 7 ?	0	3	7	2	4	5	1	6.
Work Group #S 8 - 15 ?	8	9	10	11	12	13	14	15.
Work Group #S 16 - 23 ?	16	17	18	19	20	21	22	23.
Work Group #S 24 - 31 ?	24	25	26	27	28	29	30	31.
Work Group #S 32 - 39 ?	32	33	34	35	36	37	38	39.
Work Group #S 40 - 47 ?	40	41	42	43	44	45	46	47.
Work Group #S 48 - 55 ?	48	49	50	51	52	53	54	55.
Work Group #S 56 - 63 ?	56	57	58	59	60	61	62	63.
Outgoing Trk Grp #S 0 - 7 ?	0	2	4	6	7	5	3	1.
Outgoing Trk Grp #S 8 - 15 ?	10	12.						
Outgoing Trk Grp #S 16 - 23 ?	16	19	23.					
Outgoing Trk Grp #S 24 - 31 ?								
Incoming Trk Grp #S 0 - 7 ?	0.							
Incoming Trk Grp #S 8 -15 ?	8	9	11	10.				
Incoming Trk Grp #S 16 - 23 ?	16	17.						
Incoming Trk Grp #S 24 -31 ?	24	26.						

Figure 6.4 Prompting Format for Incoming Trunk Group Number Selection

(c) The call must exist in two-way conversation for an interval of time greater than or equal to that specified as the minimum call duration for the given port. In the absence of answer supervision on outgoing trunks, calls are presumed to enter two-way conversation after the call answer time out interval in T4461, Record Code OT.

6.3.4 Dump Options Command

The DUMP OPTIONS command is used to display the current set of screening options retained in the data base for either the terminal (MDR software Port 0) or the Remote Polling Device (MDR software Port 1). If MDR software Port 0 is'equipped in data base, the screening options for the terminal will be displayed. If MDR software Port 1 is equipped in data base, then the screening options for the Remote Polling Device are displayed. To enter the Dump Options Command, the user types in "DUMP MDR OPTIONS" or "DU MDR OP" from the system maintenance terminal. A format example for the Dump Option Command is shown in Figure 6.5.

```
Dump MDR Options.
Remote Polling Device:
Call Duration (O-255 sec) > = 200
Work Group #S
00  01  02  03  04  05  06  07
08  09  10  11  12  13  14  15
16  17  18  19  20  21  22  23
24  25  26  27  28  29  30  31
32  33  34  35  36  37  38  39
40  41  42  43  44  45  46  47
48  49  50  51  52  53  54  55
56  57  58  59  60  61  62  63

Outgoing Trk Grp #S
00  01  02  03  04  05  06  07
08  09  10  11  16  17  24  26

incoming Trk Grp #S
00  02  08  10  12  13  16  19
23  25  28  30  31
```

Figure 6.5 Prompting Format for Dump Options Command

6.3.5 Output Formats - MDR Software Port 0

The MDR software port 0 is an ASCII port which may be connected to **a** hard copy terminal device, CRT, or outboard system that provides downstream processing of the output from this port. Two types of records appear on port 0: call records and time change records. A call-record header is output preceding the first call record and every sixty-fourth single-line call record or thirty-second double-line call record thereafter. Every time-change record also contains a header. See Figure 6.6 for sample call and time-change records with associated headers.

(a) **Call-Record Format**

Refer to Table 6.2 for the Call-Record Format.

(b) **Time-Change Record Format**

Refer to Table 6.3 for the Time-Change Record Format.

(c) **Termination of Records**

Call Records are terminated by a sequence constituted of CR (Carriage Return - OD hex), LF (Line Feed - OA hex) and nine ROs (rubout - 7F hex). Time-change records are terminated by two such CR, LF, RO, sequences.

Office ID: 000000 Date: 08-27-00 Time: 10:14:08

Originator # Grp C F F R	Terminator # Grp Type ....FRL.... Auth Code	Called Number...	Start Time	Call Length	A S	M R	S C	Account Code
TO63 02 3	**5208 00**	**861-5208**	10:14	00:00				

Incoming Trunk to Station

Office ID: 000000 Date: 08-27-00 Time: 14:21:40

Originator # Grp C F F R	Terminator # Grp Type ....FRL.... Auth Code	Called Number...	Start Time	Call Length	A S	M R	S C	Account Code
A000 3	**TO25 00** CO	897-2271	14:21	00:00	Y			

Console to Outgoing Trunk

Office ID: 000000 Date: 08-26-00 Time: 08:54:14

Originator # Grp C F F R	Terminator # Grp Type ....FRL.... Auth Code	Called Number...	Start Time	Call Length	A S	M R	S C	Account Code
TO63 02 3	A000	861	09:06	00:02				

Incoming Trunk to Console

Office ID: 000000 Date: 08-22-00 Time: 10:07:27

Originator # Grp C F F R	Terminator # Grp Type ....FRL.... Auth Code	Called Number...	Start Time	Call Length	A S	M R	S C	Account Code
4201 00 3	**TO24 00**	897-2271	09:09	00:00	Y			

Station to Outgoing Trunk

Figure 6.6 Port 0 Typical **Call and Time-Change Records**

Office ID: 000000 Date: 08-29-00 Time: 09:03:04

Originator	Terminator Called Number...	Start	Call	A	M	S	Account
# Grp	# Grp Type	Time	Length	S	R	C	Code
C F	....FRL....						
F R	Auth Code						
TO63 02 3	TO24 08 TIE 123-456-7890	08:07	00:00	Y			

Trunk to Trunk

Old Date	Old Time	New Date	New Time
08/28/00	11:22:09	02/28/00	11:47:09

Time Change

NOTES:

1. Abbreviations above are defined:
   CF = Call Forward
   FR = Facility Restriction Level
   AS = Answer
   MR = MERS
   SC = Speed Call

2. Line 2 is a data base programmable option.

Figure 6.6 Port 0 Typical Call and Time-Change Records (Continued)

Table 6.2 Port 0 Call-Record Format

COLUMN RANGE	INTERPRETATION
01-04	Originator of Call... -Line originators are identified by a 4-digit DN, or a "C" in column 1, followed by three digits if 3-digit numbering is used. -Trunk originators are identified by a "T" in column 1, followed by a 3-digit decimal trunk identity. The trunk identity may be decoded by dividing the trunk number by 128. The quotient is the PEC number and will always be 0. The value left in the dividend is the relative trunk number and will range between O-63. Example: TO62 = 62 ÷ 128 = 0 PEC 62 Trunk Number -Attendant originators are identified by an "A" in column 1 followed by a 3-digit decimal attendant identity (range: 000 - 001).
05-06	Always blank-filled
07-08	Originator's group number... -For line originators, this is the decimal work group number (range: 00-63). -For trunk originators, this is the trunk group number (range: 00-63). -For attendant originators, the field contains no information, and is therefore blank-filled.
09-l 0	Always blank-filled
11-14	Terminator of call... -Interpret in same manner as columns l-4.
15-16	Always blank-filled
17-18	Terminator's group number.. . -Interpret in same manner as columns 7-8.
19-20	Always blank-filled
21-24	Terminator's type: -For trunks, this field describes the type of trunk involved in the call. The field can take the following values:

Table 6.2 Port 0 Call-Record Format (Continued)

COLUMNRANGE	INTERPRETATION
	"CO" - central office trunk "RLT" - release link trunk "FX" - foreign exchange trunk "CLR" - combined line and recording trunk "WATS" - wide area telephone service trunk "TIE" - tie trunk "REC" - recorder/announcer trunk "CAMA" - centralized automatic message accounting trunk "PAGE" - paging trunk " " - (blanks) trunk type unknown
	NOTE: MERS and direct trunk access do not provide the terminator type of trunk.
	-For line or attendant terminators, the field contains no information and is therefore blank-filled.
25-26	Always blank-filled
27-46	Called number...
	-This is a right-justified string of variable length, blank-filled on the left, and containing dashes at appropriate points.
	-Dashes appear in columns 30, 34, 38, and 42 if the called number digit string is longer than 4, 7, 10, and 13 digits, respectively. In other words, leading dashes are replaced by blanks; only embedded dashes appear in the output.
Examples:	col 46 V 772-3372 3275 703-228-7131 01 l-21 6-762-4976
47-48	Always blank-filled
49-53	Start time of call...
	-Columns 49-50 contain start time hours in decimal. Time is in military format (range: 00-23).
	-Column 51 always contains a colon.
	-Columns 52-53 contain start time minutes in decimal (range: 00-59).
54-55	-Always blank-filled
56-60	Call duration...
	-Columns 56-57 contain duration hours in decimal (range: 00-99).
	-Column 58 contains a colon.

Table 6.2 Port 0 Call-Record Format (Continued)

COLUMN RANGE	INTERPRETATION
	-Columns 59-60 contain duration minutes in decimal (range: 00-59).
61-62	Always blank-filled
63	Answer supervision indicator...
	-Takes the value "Y" if answer supervision was received on the call.
	-Takes the value " " (blank) if answer supervision was not received.
64	Always blank-filled
65	Most Economical Route Selection (MERS) usage...
	-Takes the value "Y" if MERS was used on the call.
	-Takes the value " " (blank) if MERS was not used.
66-69	Always blank-filled
70-77	Account code...
	-A string of up to 8 digits, right-justified and blank-filled on the left.
	NOTE: The following is a description of the data base optional second line of the call record.
01-04	Always blank-filled
05	Call Forward Indicator.... (a) Takes the value "Y" if the call was forwarded. (b) Takes the value " " if the call was not forwarded.
06	Always blank-filled
07	Facility Restriction Level Indicator.... Takes a value in the range O-7 based on restriction level used in call.
08-I 0	Always blank-filled
11-17	FRL Authorization Code.... A string of 4-7 digits, left-justified with trailing blanks.

Table 6.3 Port 0 Time-Change Record Format

COLUMN RANGE	INTERPRETATION
01-03	Always blank-filled
04-11	Old date...
	-Columns 4-5 contain month in decimal (range: 01-12).
	-Column 6 contains a slash.
	-Columns 7-8 contain day of month in decimal (range: 01- 31)
	-Column 9 contains a slash.
	-Columns 10-11 contain the year in decimal (range: 00-99).
12-13	Always blank-filled
14-21	Old time...
	-Columns 14-15 contain hours in decimal (range: 00-23).
	-Column 16 contains a colon.
	-Columns 17-18 contain minutes in decimal (range: 00-59).
	-Column 19 contains a colon.
	-Columns 20-21 contain seconds in decimal (range: 00-59).
22-23	Always blank-filled
24-31	New date...
	-Columns 24-25 contain month in decimal (range: 01-12).
	-Column 26 contains a slash.
	-Columns 27-28 contain day of month in decimal (range: 01- 31).
	-Column 29 contains a slash.
	-Columns 30-31 contain the year in decimal (range: 00-99).
32-33	Always blank-filled
34-41	New time...
	-Columns 34-35 contain hours in decimal (range: 00-23).
	-Column 36 contains a colon.

Table 6.3 Port 0 Time-Change Record Format (Continued)

COLUMN RANGE	INTERPRETATION
	-Columns 37-38 contain minutes in decimal (range: 00-59). -Column 39 contains a colon. -Columns 40-41 contain seconds in decimal (range: 00-59).

6.3.6 Output Formats - MDR Software Port 1

The following paragraphs describe the output formats that are used to record call information for MDR output to MDR software Port 1.

Call-Record Block. A maximum of 11 entries (call records and time change records) of MDR data are accumulated in the memory buffer prior to being transmitted using FGBS-modified EBCDIC. The call-record block (Figure 6.7) contains the following specific data fields:

(a) Block Identifier. This field consists of four characters that identify the block as a call-record block. The identifier is 15 (βαβ (beta, alpha, beta, the number 15), or (DCDF) in hexadecimal.

(b) Sequential Block Count. This field consists of a four-character sequential count of call-record blocks.

(c) Call Record/Time-Change Record. This field is 49 bytes long and may be a call record or a time-change record.

Call Record. By means of the call record (Figure 6.8), information about calls in the system is recorded for later use. Each call record is 49 bytes long. Each byte is composed of FGBS-modified EBCDIC. The block contains the following specific data fields:

(a) Record Identifier. This field indicates the beginning of a new call record and the type of record. The identifier is 15 (alpha, the number 15), or (CF) is hexadecimal.

(b) Reserved for future use. The value is always 00.

(c) Type of Originator. This field describes the originator as a line, attendant, or trunk. The values are as follows:

07. Line
04. Trunk
01. Attendant

# BYTES	CONTENTS	
	BLOCK ID	
1	BLOCK ID	
1	SEQUENTIAL BLOCK COUNT, TENS	SEQUENTIAL BLOCK COUNT, UNITS
1	SEQUENTIAL BLOCK COUNT, THOUSANDS	StQUtN TIAL BLOCK COUNT, HUNDREDS
49		
49	CALL RECORD/TIME CHANGE RECORD	#2
49	CALL RECORD/TIME CHANGE RECORD	#3
49	CALL RECORD/TIME CHANGE RECORD	#4
49	CALL RECORD/TIME CHANGE RECORD	#5
49	CALL RECORD/TIME CHANGE RECORD	#6
49	CALL RECORD/TIME CHANGE RECORD	#7
49	CALL RECORD/TIME CHANGE RECORD	#8
49	CALL RECORD/TIME CHANGE RECORD	#9
49	CALL RECORD/TIME CHANGE RECORD	#10
49	CALL RECORD/TIME CHANGE RECORD	#11
1	I/O PROCESSING FLAG	
1	SPARE	
1	SPARE	
1	SPARE	

Figure 6.7 Port 1 Call-Record Block Format

BYTE	DESCRIPTION	
1	RECORD ID	
2	RESERVED FOR FUTURE USE	
3	TYPE OF ORIGINATOR	
4	ORIGINATING INFORMATION	
5	ORIGINATING INFORMATION	
6	ORIGINATING INFORMATION	
7	ORIGINATING INFORMATION	
a	WORK GROUP NUMBER	
	EQUIPMENT NUMBER	
g	PEC NUMBER	SOFTWARE ID
10	SOFTWARE ID;	SOFTWARE ID
11	TYPE OF DIALER	
12	SPECIAL FEATURES	
	CALLED NUMBER	
13	DIGIT0	DIGIT 1
14	DIGIT2	DIGIT 3
15	DIGIT4	DIGIT5
16	DIGIT6	DIGIT7
17	DIGIT 8	DIGIT9
18	DIGIT 10	DIGIT11
19	DIGIT12	DIGIT 13
20	DIGIT 14	DIGIT15

Figure 6.8 Port 1 Call-Record Format (Sheet 1 of 3)

	ACCOUNT	
21	DIGIT0	DIGIT 1
22	DIGIT 2	DIGIT 3
23	DIGIT4	DIGIT 5
24	DIGIT6	DIGIT 7
25	TYPE OF TERMINATOR	
26	TERMINATING INFORMATION	
27	TERMINATING INFORMATION	
28	TERMINATING INFORMATION	
29	TERMINATING INFORMATION	
30	WORK GROUP NUMBER	

Figure 6.8 Port 1 Call-Record Format (Sheet 2 of 3)

	EQUIPMENT NUMBER	
31	PECNUMBER	SOFTWARE ID
32	SOFTWARE ID	SOFTWARE ID
33	TERMINATING	ANSWER
	DATE OF ANSWER	
34	YEAR	YEAR
35	MONTH	MONTH
36	DAY	DAY
	TIME OF ANSWER	
37	HOUR	HOUR
38	MINUTE	MINUTE
39	SECONDS	SECONDS
	CALL DURATION	
40	HOUR	HOUR
41	MINUTE	MINUTE
42	SECONDS	SECONDS
43	MIDNIGHT PASSED	TIME CHANGES
44	SCCFLAG	CFWD
45	FACILITY RESTRICTION LVL # OF FRL AUTHORIZATION CODE DIGITS	
46	AUTHORIZATION CODE DO	D1
47	D2	D3
48	D4	D5
49	D6	D7

TOTAL = 49 BYTES/CALL RECORD

Figure 6.8 Port 1 Call-Record Format (Sheet 3 of 3)

(d) Originating Information. This field consists of eight characters that describe the type of originator in greater detail, as follows:

1. If the originator is a line, the first seven characters contain that line's directory number. The remaining character is a blank.

2. If the originator is a trunk (only incoming trunks can be originators), the first four characters contain the trunk group (00-63) and the trunk type. The valid trunk types are as follows:

81. DID
82. Tie (incoming)
83. Ringdown

The remaining characters are blanks.

3. If the originator is an attendant, the first two characters are the system console number (00-01). The remaining characters are blanks.

(e) Work Group Number. If the originator is a line, this field contains the two-character (00-63) work group number for that line. The work group number is data base programmable. If the originator is not a line, this field is filled with blanks.

(f) Equipment Number. This field consists of four characters that describe the location of the hardware associated with a line, trunk, or attendant. The first character represents the cabinet that holds the equipment (always 0 for SI). The next three characters contain the software identity of the hardware. The range of values is as follows:

Lines: 000-255
Trunks: 000-063
Attendants: 000-001

The software identity corresponds to the entry in the physical locating tables in the data base.

(g) Type of Dialer. This field consists of two characters that describe the actual dialer and/or type of dialing for the call. The values are as follows:

07. Line
04. Trunk
01. Attendant

(h) Special Features. This field consists of two characters that represent any special features used. The hex values are as follows:

08. Attendant hands off
10. Remote access
20. Trunk call queuing
40. Most Economical Route Selection (MERS)
80. Speed calling
00. No special feature used

NOTE: If more than two special features are used, the values are OR'ed. For example, a call involving remote access and trunk call queuing will show a hex value of 30.

(i) Called Number. This field consists of 16 characters that contain the dialed digits. If less than 16 digits were dialed, the remaining characters are filled with blanks. The valid range for each character is O-9, and blank. This field does not contain any access codes that were dialed to reach a trunk group in the originating PABX. Subsequently dialed tandem codes do appear.

(j) Account Code. This field consists of eight characters containing an account code that may have been dialed. If the account code is not eight characters long, the remaining characters are filled with blanks. If an account code is not used, the field is filled with blanks.

(k) Type of Terminator. This field consists of two characters that describe the terminator of the call. The values are as follows:

07. Line
04. Trunk
01. Attendant

(l) Terminating Information. This field consists of eight characters that describe the type of terminator in greater detail as follows:

If the terminator is a line, the first seven characters contain that line's directory number. The remaining character is blank.

If the terminator is a trunk (outgoing trunks only), the first four characters contain the trunk group number (00-63) and the trunk type. The valid trunk types are as follows:

00. Release link trunk (Centralized Attendant Service (CAS) branch Feature)
01. CO trunk
02. FX trunk
03. CCSA trunk
04. WATS trunk
05. Tie trunk (ringdown)
06. Tie trunk
07. MERS or CAMA trunk
0A. DTA (Direct Trunk Access)

The remaining characters are filled with blanks.

If the terminator is an attendant, the first two characters contain the system console number (00-01). The remaining characters are filled with blanks.

NOTE: MERS and direct trunk access do not provide the terminator type of trunk.

(m) Work Group Number. If the terminator is a line, this field contains the two-character work group number of that line (00-63). If the terminator is not a line, the field is filled with blanks.

(n) Equipment Number. This field consists of four characters that describe the 'location of the hardware associated with a line, trunk, or attendant. The first character represents the cabinet containing the equipment (always 0 for SI). The next three characters represent the software identity of the hardware. The range of values is as follows:

Lines: 000-255
Trunks: 000-063
Attendants: 000-001

(o) Terminating Answer. On outgoing trunk calls, this field consists of two characters that describe the type of answer generated for a particular call, as follows:

If answer supervision was received, the value is 02.

If answer supervision was not received, the value is 01. When this is the case, call duration timing begins at the time out value programmed in the call answer time out table in the office-dependent data base.

(P) Date of Answer. This field consists of six characters (two characters each: hour, minutes, seconds) that record the date when the call was answered.

(q) Time of Answer. This field consists of six characters that record the time when the call was answered. The value of the clock is in 24-hour format.

(r) Call Duration. This field consists of six characters that record the length of the call. The maximum value is 99 hours, 59 minutes, and 59 seconds.

(s) Midnights Passed. This field consists of one hexadecimal character (O-F) to record the number of times the call extended past midnight. This value does not return to zero, but remains at F if that number is reached.

(t) Time Changes. This field consists of one hexadecimal character (O-F) to record the number of time-change records generated while the call was in progress. This indicates that the call duration must be adjusted by the amount of time shown on the time-change record. This value does not return to zero but remains at F if that number is reached.

(u) SCC Flag. Indicates if an SCC was used (1 bit 0 = no, 1 = yes).

(v) Call Forward. Indicates if the call was forwarded (1 bit 0 = no, 1 = yes).

(w) Facility Restriction Level. Indicates Facility Restriction Level used in call (hex character O-7).

(x) Number of FRL Authorization Code Digits. Indicates how many FRL Authorization Code Digits are in the next field (0, or 4-7 hex).

(y) FRL Authorization Code. Indicates Authorization Code used to change Facility Restriction Level (4-7 binary code decimal digits).

Time-Change Record. A Time-Change Record (Figure 6.9) is made whenever the date or time in the system memory is adjusted. A time-change record is inserted with other call records in a call-record block in the proper time sequence. Records preceding a time-change record are not affected by the time change. The off-line processor is required to adjust any call records affected by a time change. The length of the time-change record is 49 bytes. The time-change record block consists of the following specific data fields:

(a) Record Identifier. This field consists of two characters that indicate the start of a new record and identify it as a time-change record. The identifier is (alpha, alpha).

(b) Previous Date. This field consists of six characters to record the date before it was changed.

(c) Previous Time. This field consists of six characters to record the time before it was changed.

(d) Adjusted Date. This field consists of six characters to record the new date.

(e) Adjusted Time. This field consists of six characters to record the new time. This is used in conjunction with the previous time to calculate the duration of calls in progress when the time change record was generated.

BYTE	DESCRIPTION	
1	RECORD ID	
	PREVIOUS DATE	
2	YEAR	YEAR
3	MONTH	MONTH
4	DAY	DAY
	PREVIOUS TIME	
5	HOUR	HOUR
6	MINUTE	MINUTE
7	SECONDS	SECONDS
	ADJUSTED DATE	
8	YEAR	YEAR
9	MONTH	MONTH
10	DAY	DAY
	PREVIOUS TIME	
11	HOUR	HOUR
12	MINUTE	MINUTE
13	SECONDS	SECONDS
	THE NEXT 36 BYTES/CALL RECORD	

TOTAL = 49 BYTES/CALL RECORD

Figure 6.9 Time-Change Record Format

6.3.7 Engineering Applications

(a) **Hardware.** No additional hardware is required for the SI system MDR option.

(b) **Output Device Speed Considerations.** When selecting the speed for the output device from port 0 and port 1, the amount of traffic being handled by the system must be considered. The MDR file is capable of buffering a fixed amount of call-record blocks, beyond which the information is completely lost for additional calls. Thus, the worst case peak traffic must be considered, along with the work group screening options and the trunk group screening options. These factors will determine the probability of losing a call-record block.

The MDR file is capable of buffering 300 call-record blocks. As each call record is output from the system, an additional call-record block can be generated and the data saved for output. If, however, 300 call-record blocks are already in use at the time another call-record block is generated, the data in that block will be lost for output to that port. The speed of the TTY or other output receiving device must therefore be chosen so that call-record blocks are not lost.

NOTE: When a call-record block is lost, no indication of the loss is provided nor is there any means of recovering the lost data.

6.3.8 Call Accounting for OMNI SI With PD-200 Data Switch

The accounting information for voice calls is processed by MDR software in the MDR processor for SI. X.25 data call records (Figure 6.10) are generated, maintained, updated, and processed by the data accounting software in the ADMP.

Depending on the switch configuration, the call record output to the terminal is provided as follows:

(a) Port0 = ASCII (record by record).

(b) Port 1 = Binary output in blocks of 11 call records to remote station via RS-232.

(c) Port 1 output, in the same format, can be transferred via an APM.

6.3.9 Data/Voice Switch Interface for MDR

Voice call records can be sent from MDR Port 1 through an APM to a remote terminal or computer for further processing. The format of MDR Port 1 output is shown in Figure 6.6.

BYTE	DESCRIPTION	
0	RECORD ID LENGTH INDICATOR	
1	CALLING LENGTH	CALLED LENGTH
	CALLED ADDRESS	
2	DIGIT 0	DIGIT 1
3	DIGIT 2	DIGIT 3
4	DIGIT 4	DIGIT 5
5	DIGIT 6	DIGIT 7
6	DIGIT 8	DIGIT 9
7	DIGIT 10	DIGIT 11
8	DIGIT 12	DIGIT 13
	CALLING ADDRESS	
9	DIGIT 0	DIGIT 1
10	DIGIT 2	DIGIT 3
11	DIGIT4	DIGIT 5
12	DIGIT 6	DIGIT 7
13	DIGIT 8	DIGIT 9
14	DIGIT 10	DIGIT 11
15	DIGIT 12	DIGIT 13
16	CALLED PORT	
17	CALLED PORT	
18	CALLING PORT	
19	CALLING PORT	
20	CALL REQUEST FACILITIES	
21	CALL ACCEPT FACILITIES	
22	CLEARING CAUSE	
23	CLEARING DIAGNOSTIC	
24	SOURCE TO DESTINATION PACKET COUNT	
25	SOURCE TO DESTINATION PACKET COUNT	
26	SOURCE TO DESTINATION PACKET COUNT	
27	DESTINATION TO SOURCE PACKET COUNT	
28	DESTINATION TO SOURCE PACKET COUNT	
29	DESTINATION TO SOURCE PACKET COUNT	
30	CALL IDENTIFIER	
31	CALL IDENTIFIER	WHOCLEAREDCALL
32	CLOSED USER GROUP	
33	CLOSED USER GROUP	
34	CLOSED USER GROUP	
	DATE OF ANSWER	
35	YEAR	YEAR
36	MONTH	MONTH
37	DAY	DAY
	TIME OF ANSWER	
38	HOUR	HOUR
39	MINUTE	MINUTE
40	SECONDS	SECONDS
	CALL DURATION	
41	HOUR	HOUR
42	MINUTE	MINUTE
43	SECONDS	SECONDS
44	#MIDNIGHTS	TIME CHANGES

Figure 6.10 X.25 Call Record Format

7.0 ACD PROCEDURES FOR STANDARD TELEPHONE

A standard telephone station instrument may be used as an Automatic Call Distribution (ACD) station. The procedures listed below will demonstrate how to use a standard telephone for ACD application.

7.1 Preparing for Service

STEP	OPERATION	RESULT
1	Go off-hook.	(a) Position is placed into the outgoing state. (b) Dial tone is heard.
2	Dial the staff access code.	Position is staffed, and confirmation tone is heard.

7.2 Emergency Service/Supervisory Assistance

STEP	OPERATION	RESULT
1	Perform a hookswitch flash.	(a) System returns distinctive dial tone. (b) Caller is placed on hold.
2	Dial emergency service or supervisory assistance access code.	(a) System sends appropriate message to supervisory instrument. (b) Agent hears confirmation tone and is returned to the caller.
3	Supervisor monitors call.	

7.3 Bad Line Service

STEP	OPERATION	RESULT
1	Perform a hookswitch flash.	(a) System returns distinctive dial tone. (b) Caller is placed on hold.
2	Dial bad line access code.	(a) System records the line information. (b) Agent hears confirmation tone and is returned to the caller.

7.4 PABX Features

STEP	OPERATION	RESULT
1	Perform a hookswitch flash.	(a) System returns distinctive dial tone. (b) Caller is placed on hold.
2	Dial PABX feature code desired.	
NOTE:	This procedure permits access to the features previously described in this section.	

7.5 Unstaffing the ACD

STEP	OPERATION	RESULT
1	Check that line is not active.	
2	Perform hookswitch flash.	System returns regular dial tone.
3	Dial the unstaffed access code.	
NOTE:	If distinctive dial tone is heard, an incoming call was put on hold due to a glare condition. You must perform a hookswitch flash to get the caller back.	
4	Dial the unstaffed access code.	(a) System returns confirmation tone. (b) Position is placed into the unstaffed position.
5	Go on-hook.	

7.6 Answering an Incoming ACD Call

STEP	OPERATION	RESULT
1	A call is placed to the agent position.	Alerting tone heard.
2	Answer the calling party with an appropriate phrase and provide the necessary assistance.	
3	When no further assistance is required, go on-hook.	(a) Caller is disconnected. (b) Position is placed into the work state.

STEP	OPERATION	RESULT
4	Perform the necessary off-line work.	
5	When no further work is required, go off-hook.	Position is staffed and ready.

7.7 Placing an ACD Call (Standard Telephone)

STEP	OPERATION	RESULT
1	Check that line is not active.	
2	Perform a hookswitch flash.	System returns regular dial tone.
NOTE:	If distinctive dial tone is heard, an incoming call was put on hold due to a glare condition. The agent must perform a hookswitch flash to get the caller back. Refer to paragraph 7.6.	
3	Key necessary digits.	
4	Handle the call.	
5	When call is complete, go on-hook.	Called party is disconnected.
6	To ready the position, (a) Go off-hook. (b) Perform a hookswitch flash.	 System returns regular dial tone. Position is placed in a ready condition.

7.8 Parking a Call

An agent parks a call by flashing and dialing a destination. The destination may be an agent group pilot number, or the line-l directory number of another agent. When the parked-to agent becomes available, the parked party will automatically be connected.

A park timer provides a predetermined length of time that a call may be parked. If the call times out, it is queued to the parked-to agent group delay queue. At this point, the next available agent in the group will receive the call.

7.9 Agent Call Hold/Consultation

The agent places a call on hold by flashing, then keying the station call hold access code. After receiving the feature confirmation tone, the agent is automatically placed into the outgoing state. The agent may then opt to go into a work state, key the pilot number of an agent group or key the directory number of an agent or station consultation. Additional consultation calls can be made by releasing from the previous call and going into the outgoing state. To return to the held call: the agent becomes available, the call alerting tone is received and the held call automatically terminates to the agent's line-l.

The agent call hold timer provides a predetermined length of time that the call is held. The first time a call times out, the agent hears a burst of tone and the call is requeued to that agent. If the call times out a second time, it is requeued to that agent group delay queue. At this point, the next available agent in the group receives the call.

8.0 CAS/ACD AGENT INSTRUMENT (PACET) PROCEDURES

The following provides the procedures for processing agent-related features.

8.1 Preparing the Agent Instrument for Service

8.1.1 For CAS Main Service

STEP	OPERATION	RESULT
1	Plug handset/headset into Agent Instrument.	LED off.
2	Depress Message Waiting pushbutton.	READY LED on.
3	Depress POS STAFF pushbutton.	POS STAFF LED on.
NOTE:	Message clear LED will be on when system first loads.	

8.1.2 For ACD Service

STEP	OPERATION	RESULT
1	Plug handset/headset into Agent Instrument.	All LEDs off.
2	Depress Message Waiting pushbutton.	READY LED on.
3	Depress POS STAFF (ACD) pushbutton.	POS STAFF (ACD) LED on.
4	Depress READY (ACD) pushbutton.	READY (ACD) LED on.
NOTE:	Message clear LED will be on when system first loads.	

8.2 Answering Incoming Line-l Call

8.2.1 As a CAS Main

STEP	OPERATION	RESULT
1	Incoming call received.	(a) Alerting tone heard.

STEP	OPERATION	RESULT
		(b) The left portion of display indicates calling source, e.g., "TOYS 252". (c) LINE 1 and RLS LEDs on.
2	Answer calling party and provide necessary assistance.	
3	Depress RLS pushbutton to release call.	(a) LINE 1 and RLS LEDs off. (b) Left portion of display is cleared. (c) If lit, the BAD LINE, SUPR ASST, or EMER LED off. (d) If line 2 is on hold, POS STAFF LED off.

8.2.2 As an ACD

STEP	OPERATION	RESULT
1	Incoming call received.	(a) Alerting tone heard. (b) Left portion of display indicates source, e.g., "TOYS 252". (c) LINE 1 and RLS LEDs on. (d) READY (ACD) LED off.
2	Answer calling party and provide necessary assistance.	
3	When no further assistance is required, depress RLS pushbutton to release call.	(a) LINE 1 LED off. (b) Left portion of display cleared. (c) If lit, BAD LINE, SUPR ASST (AGT), or EMER (AGT) LED goes off. (d) RLS LED off.
4	Perform necessary off-line work.	
5	Depress READY (ACD) pushbutton when work is complete.	READY (ACD) LED on.

a.3 Answering Incoming Line-2 Call (CAS Main/ACD)

STEP	OPERATION	RESULT
1	Call received on line 2.	The LINE 2 LED flashes at 120 IPM and ringing is heard.
2	Depress LINE 2 pushbutton.	(a) One of the following occurs: 1. If line 1 is active, request for line 2 is ignored. 2. If line 1 is on hold, LINE 2 LED on. 3. If line 1 is staffed/ready, POS STAFF (CAS) or READY (ACD) LED turns off and LINE 2 LED is on. (b) LINE 2 and RLS LEDs on.
3	When call is completed, depress the RLS pushbutton to release from call.	(a) LINE 2 and RLS LEDs off. (b) Agent Instrument returned to initial state of staffed ready.

a.4 Placing Line-2 Call (CAS Main/ACD)

STEP	OPERATION	RESULT
1	Check that line 1 is either on hold or inactive.	
2	Depress LINE 2 pushbutton.	(a) If the instrument is in POS STAFF (CAS) or READY (ACD) state, it is removed from that state. (b) LINE 2 and RLS LEDs on. (c) Dial tone heard.
3	Dial desired number or depress appropriate repertory dial pushbutton.	(a) All associated LEDs on. (b) The right portion of the display indicates the dialed number "XXXX". (c) Ringback tone heard.
4	Depress RLS pushbutton to release call.	(a) RLS LED on during pushbutton depression interval. (b) LINE 2 and RLS LEDs off.

8.5 Extending Call to Station (CAS Main)

STEP	OPERATION	RESULT
1	With call on line 1, depress FLASH pushbutton.	(a) FLASH LED on. (b) Calling party put on hold. (c) Dial tone heard from branch location.

NOTE: If flash is part of repertory dial number being used, this step is not necessary.

STEP	OPERATION	RESULT
2	Depress appropriate repertory dial pushbutton.	(a) Repertory dial LED on. (b) Repertory dial message displayed on right portion of display. (c) Ringback tone heard.

NOTE: The number may be dialed manually if a repertory dial is not programmed for the number.

STEP	OPERATION	RESULT
3	Depress RLS pushbutton.	(a) LINE 1 LED off. (b) RLS LED off.

8.6 Three-Party Conference

8.6.1 Establishing Three-Party Connection

With call on line 1, add line 2 to the connection by performing the following:

STEP	OPERATION	RESULT
1	Place line 1 on hold.	LINE 1 LED winks.
2	Place line 2 call.	LINE 2 LED on.
3	Depress LINE 1 pushbutton.	LINE 1 LED on steady signaling that three-party connection is established.

8.6.2 Releasing Three-Party Connection

To release call on line 1, and place line 2 on hold:

STEP	OPERATION	RESULT
1	Depress LINE 2 pushbutton.	LINE 2 LED flashes.
2	Depress RLS pushbutton.	LINE 1 LED off.

To release a call on line 2, and place line 1 on hold:

STEP	OPERATION	RESULT
1	Depress LINE 1 pushbutton.	LINE 1 LED flashes.
2	Depress RLS pushbutton.	LINE 2 LED off.

To release calls on lines 1 and 2:

STEP	OPERATION	RESULT
1	Depress RLS pushbutton.	LINE 1 and LINE LEDs off.

a.7 PABX Features on an ACD Call

STEP	OPERATION	RESULT
1	Depress FLASH pushbutton.	(a) FLASH LED on. (b) Dial tone heard.
2	Depress the appropriate repertory dial pushbutton, or the system code for the desired feature.	

8.8 Call Hold (CAS Main/ACD)

To place a call on hold while connected to parties on line 1 or line 2:

STEP	OPERATION	RESULT
1	Depress LINE pushbutton with a call.	LINE LED winks.
2	Depress LINE pushbutton again to return to the call on'hold.	LINE LED on steady.

8.9 Bad Service Line Reporting

STEP	OPERATION	RESULT
1	Transmission difficulty is encountered on line 1. To report, depress BAD steadily.	BAD LINE LED winks until system records line information: then BAD LINE LED on LINE pushbutton.

8.10 Supervisory Assistance Features

8.10.1 Emergency Assistance Request

STEP	OPERATION	RESULT
1	Depress EMER (AGT) pushbutton with a call on line 1.	(a) EMER (AGT) LED flashes at 120 IPM. (b) TLWMON (SUPV) LED on the supervisory instrument flashes at 120 IPM. (c) Emergency ringer sounds at supervisory instrument.
2	Depress TONE MUTE (SUPV) pushbutton on supervisory instrument.	(a) Emergency ringer stops. (b) "EMERGENCY AGENT XX"

NOTE: If ringer silence is not desired before talking or when monitoring the requesting agent, do not perform Step 2, but go to Step 3.

STEP	OPERATION	RESULT
3	Depress TLWMON (SUPV) pushbutton on supervisory instrument.	(a) TLWMON (SUPV) LED on. (b) Supervisory instrument is connected to requesting Agent Instrument over supervisory line 2, and the supervisory LINE 2 LED on. (c) Supervisor can talk with/monitor requesting agent.

NOTE: If Step 2 operation is not performed, the emergency message stops, and the requesting agent ID number appears in supervisory instrument display.

STEP	OPERATION	RESULT
4	Depress RLS pushbutton on supervisory instrument to release call.	TALWMON (SUPV), LINE 2, and RLS LEDs off.

8.10.2 Requesting Supervisory Assistance

STEP	OPERATION	RESULT
1	Depress SUPR ASST pushbutton.	SUPR ASST LED winks.

8.10.3 Responding to Supervisory Assistance Request

STEP	OPERATION	RESULT
1	Depress SUPR ASST (AGT) pushbutton with a call on line 1.	(a) TLWMON (SUPV) LED on supervisory instrument flashes at 60 IPM. (b) Assistance ringer sounds at supervisory instrument.

NOTE: If ringer silence is not desired before talking or monitoring the requesting agent, do not perform Step 2, but go to Step 3.

STEP	OPERATION	RESULT
2	Depress TONE MUTE (SUPV) pushbutton on supervisory instrument.	(a) Assistance ringer stops. (b) “ASSIST AGENT XX”

STEP	OPERATION	RESULT
3	Depress TLWMON (SUPV) pushbutton.	(a) TLWMON (SUPV) LED on. (b) Supervisor instrument is connected to requesting Agent Instrument over supervisor line 2 and supervisor LINE 2 LED on. (c) Supervisor can talk with/monitor requesting agent. (d) RLS LED on.

NOTE: If Step 2 operation is not performed, the assistance ringer stops, and assistance message and requesting agent ID number appear in the supervisory instrument display.

STEP	OPERATION	RESULT
4	Depress RLS pushbutton to release call.	TLWMON (SUPV), LINE 2, (SUPV) and RLS LEDs off.

8.10.4 Supervisor-Initiated Monitor

STEP	OPERATION	RESULT
1	Depress TLWMON (SUPV) pushbutton. Seizes line 1 or line 2.	(a) TLWMON (SUPV) LED on supervisory instrument winks. (b) DIAL AGENT ID appears in display. (c) TLWMON (SUPV) LED on.
2	Dial a two-digit agent identification.	Supervisor can talk with/monitor agent at Agent Instrument if it is staffed/ready.
3	Depress the RLS pushbutton to release call.	Returned to normal.

8.41 Unstaffing Agent Instrument

8.11.1 As a CAS Main

STEP	OPERATION	RESULT
1	Depress POS STAFF (CAS) pushbutton.	POS STAFF (CAS) LED off.

NOTES:

1. Before removing the headset/handset from the Agent Instrument (if desired for safekeeping), the Agent Instrument must be in the idle mode or a warning tone (alarm) will sound. An Agent Instrument is in the idle mode under the following conditions:

   (a) unstaffed
   (b) not on hold
   (c) not ringing
   (d) lines 1 and 2 are both inactive

   In addition, if the Agent Instrument is a supervisor position, there must not be any emergency or supervisory assistance request pending.

2. The Agent Instrument should be left POWERED ON.

8.11.2 As an ACD Agent

STEP	OPERATION	RESULT
1	Depress READY pushbutton.	READY LED off
2	Depress POS STAFF (ACD) pushbutton.	POS STAFF (ACD) LED off.

NOTE: Same conditions as in the note in paragraph 8.11.1

THIS PAGE INTENTIONALLY LEFT BLANK.

9.0 KEY ENTRY DISPLAY UNIT

9.1 General

The Key Entry Display Unit (KEDU), shown in Figure 9.1, is an optional accessory necessary for hotel or health care use to provide additional features including, but not limited to, message waiting, message metering, and room status.

In the Hotel/Health Care configuration the FGBS OMNI SI PABX will accommodate a maximum of two Attendant Consoles, four KEDUs, and a maximum of one hard copy printer per KEDU to provide records of all transactions. The Axiom printer is shown in Figure 9.2.

The KEDU is a desk-mounted console containing data and control entry pushbuttons, control function LEDs, status and clock displays.

9.2 Controls, Indicators, and Displays

Various controls, indicators, and displays located on the KEDU (Figure 9.2) are listed below with brief functional descriptions.

SECURITY LOCK. Prevents unauthorized use of the KEDU.

ALM RESET. (Alarm Reset) Displays the current integrity of memory and allows reset.

WAKE UP. Allows the entry of an automatic wake-up call for a specific room and time, up to 24 hours in advance.

MESG METER. (Message Meter) Allows the entry of a room number for recording outgoing calls. Recording can be pegged per completed call or on a measured rate of service based on the trunk group. This information is programmed into the data base for the specific switch and location.

ROOM RSTR. (Room Restriction) Allows the entry of a room number to be restricted from Direct Outside Dialing (DOD).

TIME. Allows the display or update of month, day, hour, minute, and year.

MESG WTG. (Message Waiting) Allows the entry of a room number by flashing a lamp on the room telephone to advise a guest of a message waiting at the desk.

ROOM STAT. (Room Status) Allows the status of a room to be displayed as follows: available or unavailable, occupied or unoccupied; if maid service is needed, in progress, or completed.

DONT DIST. (Do Not Disturb) Allows a room number to be entered to block that room from all calls. An administrative station or the Attendant Console' can activate, deactivate, or override the DO NOT DISTURB feature.

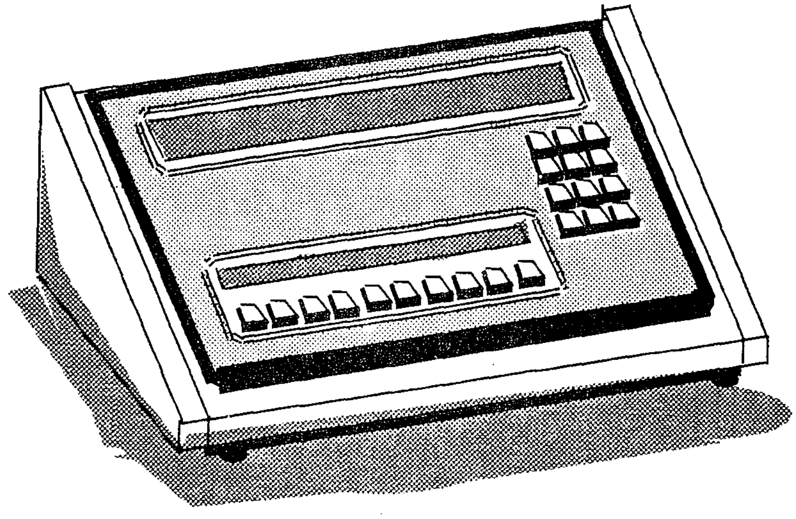


Figure 9.1 Key Entry Display Unit

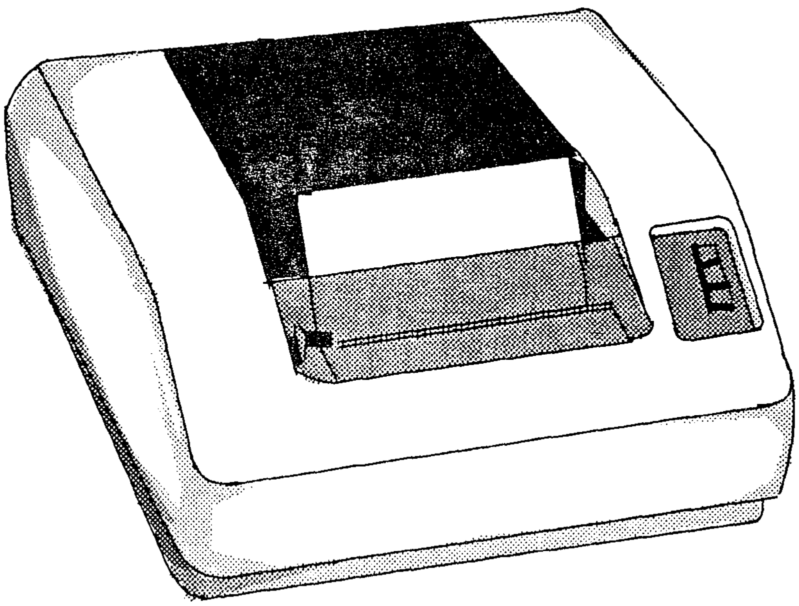


Figure 9.2 Axiom Printer Used with (or without) KEDU in Hotel Health Care Applications

CLEAR. Allows the removal of data or time information from the KEDU memory.

KEYPAD. Allows the entry of the appropriate numerical data required for programming and executing the various functions.

ENTER/XCUTE. (Enter/Execute) Allows the removal of data if depressed after the CLEAR pushbutton, or allows entry of new data after it is entered via the KEYPAD, and executes the new data when depressed a second time for the following features:

WAKE UP	MESG WTG
MESG METER	ROOM STAT
ROOM RSTR	DONT DIST

MONTH-DAY. When the TIME function is requested, the LED lights and the display indicates the month (1 through 12) and day (1 through 31).

HOUR-MINUTE. When the TIME or WAKE-UP function is requested, the LED lights and the display indicates real or wake-up times, respectively.

ROOM NUMBER. When DONT DIST, MESG METER, MESG WTG, ROOM RSTR, ROOM STAT, or WAKE-UP functions are being executed, the LED lights and the display indicates the room number (3 or 4 digits).

DATA-RESPONSE CODE. When DONT DIST, MESG METER, MESG WTG, ROOM RSTR, or ROOM STAT functions are requested, the LED tights, and the display indicates the respective data and/or response code.

9‘3 Preparing the KEDU for Service

9.3.1 Real-Time Clock Update

The real-time clock must be updated manually for system start-up, time changes, and a new year. When more than one KEDU is used, only the KEDU designated as master can change the real time.

STEP	OPERATION	RESULT
1	Depress TIME button.	(a) TIME LED on. (b) MONTH-DAY and HOUR-MINUTE are displayed.
2	Depress ENTER/XCUTE button.	All display fields off.
3	Enter appropriate month, day, hour, and minute. Month and day must be two digits. Hour and minute must be in military time. (See Table 4.1 for time of day, month, day, and time code conversion information.)	The appropriate month, day, hour, and minute appear in their respective display fields.

STEP	OPERATION	RESULT
4	Depress ENTER/XCUTE button.	(a) Response code field displays "A" for valid entry. (b) Response code field displays "2" for an invalid entry. (c) If a printer is provided, a print-out is generated that gives the transaction code, the previous date and time, the new date and time, and the new device identity number. See Table 9.1 for typical print-outs.
5	To change YEAR only, repeat Steps 1 and 2 only.	Same as Steps 1 and 2.
6	Enter OOOOOOXX, where XX are the last two digits of the present year.	(a) The MONTH-DAY display-field displays 0000. (b) The HOUR-MINUTE display field displays OOXX, where XX are the last two digits of the present year.
7	Repeat Step 4.	Same as Step 4.

NOTES:

1. When minutes are being changed, use another time source (such as a wall clock) as a reference and defer operation of the ENTER/XCUTE pushbutton until the time displayed coincides with that shown by the reference source.

2. If the KEDU is left in the TIME mode, the current time and date are automatically displayed.

Table 9.1 Typical Print-outs

DATE TIME TRANSACTION CODE
01/01 02:05 MESG WTG
5606 4
ROOM/STATION NUMBER — DEVICE IDENTITY NUMBER'

TYPICAL MESSAGE-WAITING ACTIVATION PRINT-OUT.*

DATE TIME TRANSACTION CODE
01/01 02:11 ENT WAKE-UP
5206 02:12 4
ROOM/STATION NUMBER — WAKE-UP TIME — DEVICE IDENTITY NUMBER •

TYPICAL WAKE-UP ACTIVATION PRINT-OUT.*

DATE TIME TRANSACTION CODE
01/01 02:06 CLR MESG WTG
5206 4
ROOM/STATION NUMBER — DEVICE IDENTITY NUMBER *

TYPICAL MESSAGE-WAITING CANCELLATION PRINT-OUT.*

DATE TIME TRANSACTION CODE
01/01 02:13 CLR WAKE-UP
5206 02:12 4
ROOM/STATION NUMBER — WAKE-UP TIME — DEVICE IDENTITY NUMBER •

TYPICAL WAKE-UP CANCELLATION PRINT-OUT.*

DATE TIME TRANSACTION CODE
01/01 02:07 DO NOT DIST
5206 4
4
ROOM/STATION NUMBER — DEVICE IDENTITY NUMBER •

TYPICAL DO-NOT-DISTURB ACTIVATION PRINT-OUT.*

DATE TIME TRANSACTION CODE
01/01 02:14 RESTRICT
5209 4
ROOM/STATION NUMBER — DEVICE IDENTITY NUMBER •

TYPICAL ROOM-RESTRICTION ACTIVATION PRINT-OUT.*

DATE TIME TRANSACTION CODE
01/01 02:08 CLR DO NOT DIST
5206 4
4
ROOM/STATION NUMBER — DEVICE IDENTITY NUMBER •

TYPICAL DO-NOT-DISTURB CANCELLATION PRINT-OUT.*

DATE TIME TRANSACTION CODE
01/01 02:17 CLR RESTRICT
5209 4
ROOM/STATION NUMBER — DEVICE IDENTITY NUMBER •

TYPICAL ROOM-RESTRICTION CANCELLATION PRINT-OUT.*

* SEE NOTES AT END OF TABLE.

Table 9.1 Typical Print-outs (Continued)

TRANSACTION CODE DATE TIME

```
CHG TIME O L D 01/01/00 02:26
         NEW   01/02/00 lo:oo 0
```

DATE TIME DEVICE IDENTITY NUMBER •

TYPICAL CHANGE OF KEDU REAL-TIME CLOCK PRINT-OUT.*

DATE TIME TRANSACTION CODE

```
01/02   07:03   TRY 2
5209 BUSY
```

ROOM/STATION NUMBER STATION RESPONSE

TYPICAL WAKE-UP ATTEMPT TO A BUSY STATION PRINT-OUT.

DATE TIME TRANSACTION CODE

```
01/02   06:58   ENT WAKE-UP
5209    07:00   0
```

ROOM/STATION NUMBER WAKE-UP TIME DEVICE IDENTITY NUMBER *

TYPICAL KEDU WAKE-UP ACTIVATION PRINT-OUT.*

DATE TIME TRANSACTION CODE

```
01/02   07:06   TRY 3
5209 ANS
```

ROOM/STATION NUMBER STATION RESPONSE

TYPICAL ANSWERED WAKE-UP ATTEMPT PRINT-OUT.

DATE TIME TRANSACTION CODE

```
01/02   07:00   TRY 1
5209 RUNG
```

ROOM/STATION NUMBER STATION RESPONSE

TYPICAL SYSTEM WAKE-UP ATTEMPT PRINT-OUT.

DATE TIME TRANSACTION CODE

```
04/09   10:06   ENT WAKE-UP
5209    07:00   0
```

4 WAKE-UP TIME ROOM/STATION NUMBER DEVICE IDENTITY NUMBER *

TYPICAL KEDU WAKE-UP ENTRY PRINT-OUT.*

* SEE NOTES AT END OF TABLE.

Table 9.1 Typical Print-outs (Continued)

```
DATE     TIME      TRANSACTION CODE
04/09    10:11     CLR WAKE-UP
5209     07:00     0
4
ROOM/STATION   WAKE-UP   DEVICE IDENTITY
NUMBER         TIME      NUMBER •
```

TYPICAL KEDU WAKE-UP CANCELLATION PRINT-OUT.*

```
DATE     TIME      TRANSACTION CODE
04/09    10:28     MESG WTG
5209     0
ROOM/STATION   DEVICE IDENTITY
NUMBER         NUMBER •
```

TYPICAL KEDU MESSAGE-WAITING ACTIVATION PRINT-OUT.*

```
DATE     TIME      TRANSACTION CODE
04/09    10:15     CLR METER
5209     055       0
ROOM/STATION   MESSAGE       DEVICE IDENTITY
NUMBER         METER COUNT   NUMBER •
```

TYPICAL KEDU MESSAGE-METER CLEARANCE PRINT-OUT.*

```
DATE     TIME      TRANSACTION CODE
04/09    10:30     CLR MESG WTG
5209     0
ROOM/STATION   DEVICE IDENTITY
NUMBER         NUMBER •
```

TYPICAL KEDU MESSAGE-WAITING CANCELLATION PRINT-OUT.*

```
DATE     TIME      TRANSACTION CODE
04/09    10:20     CHG METER
5209 OLD 055   NEW 057        0
ROOM/STATION   MESSAGE       DEVICE IDENTITY
NUMBER         METERCOUNT    NUMBER •
```

TYPICAL KEDU MESSAGE-METERCHANGE PRINT-OUT.*

```
DATE     TIME      TRANSACTION CODE
04/09    10:32     RESTRICT
5210     0
ROOM/STATION   DEVICE IDENTITY
NUMBER         NUMBER •
```

TYPICAL KEDU ROOM-RESTRICTION ACTIVATION PRINT-OUT.*

* SEE NOTES AT END OF TABLE.

Table 9.1 Typical Print-outs (Continued)

DATE / TIME / TRANSACTION CODE

04/09 10:35 CLR RESTRICT
5210 0

ROOM/STATION NUMBER / DEVICE IDENTITY NUMBER •

TYPICAL KEDU ROOM-RESTRICTION CANCELLATION PRINT-OUT.*

DATE / TIME / TRANSACTION CODE

04/09 11:41 CHG STATUS
5207 OLD 000 NEW 113 0

ROOM/STATION NUMBER / STATUS CODE / DEVICE IDENTITY NUMBER'

TYPICAL KEDU CHANGE-OF-STATUS PRINT-OUT.*

DATE / TIME / TRANSACTION CODE

04/09 13:46 DO NOT DIST
5208 0

ROOM/STATION NUMBER / DEVICE IDENTITY NUMBER *

TYPICAL KEDU DO-NOT-DISTURB ACTIVATION PRINT-OUT.*

DATE / TIME / TRANSACTION CODE

04109 11:45 CLR STATUS
5207 OLD 113 NEW 000 0

ROOM/STATION NUMBER / STATUS CODE / DEVICE IDENTITY NUMBER •

TYPICALCLEARANCE-OF-STATUS PRINT-OUT.*

DATE / TIME / TRANSACTION CODE

04/09 13:48 CLR DO NOT DIST
5208 0

ROOM/STATION NUMBER / DEVICE IDENTITY NUMBER *

TYPICAL KEDU DO-NOT-DISTURB CANCELLATION PRINT-OUT.*

DATE / TIME / TRANSACTION CODE

04/09 11:52 PENDING WAKE-UPS

ROOM	TIME	ROOM	TIME
5105	08:00	5204	06:30
5512	07:00	5806	07:00
1004	08:30	1515	08:00

ALL PENDING WAKE-UPS DUMPED

FINAL TRANSACTION CODE

TYPICAL WAKE-UP AUDIT PRINT-OUT.

* SEE NOTES AT END OF TABLE.

Table 9.1 Typical Print-outs (Continued)

DATE TIME TRANSACTION CODE

04/19 14:34 NON-ZERO METERS

ROOM	VAL	ROOM	VAL	ROOM	VAL
5204	050	5608	002	5809	100
1221	255	1406	050	1513	007

ALL NON-ZERO METERS DUMPED

FINAL TRANSACTION CODE

TYPICAL MESSAGE-METER AUDIT PRINT-OUT.

DATE TIME TRANSACTION CODE

04/19 14:19 RESTRICTED ROOMS

5304 5508 5915 1212 1406

1624

ALL RESTRICTED ROOMS DUMPED

FINAL TRANSACTION CODE

TYPICAL ROOM-RESTRICTION AUDIT PRINT-OUT.

DATE TIME TRANSACTION CODE

04/19 14:24 MESSAGES WAITING

5305 5607 5924 1111 1313

1409

ALL MESSAGES WAITING DUMPED

FINAL TRANSACTION CODE

TYPICAL MESSAGE-WAITING AUDIT PRINT-OUT.

DATE TIME TRANSACTION CODE

04/09 11:52 ROOM STATUS

ROOM	OAM	ROOM	OAM	ROOM	OAM
5101	000	5102	000	5103	000
5104	000	5105	000	5106	000
5107	000	5108	000	5109	000
5110	000	5111	000	5112	000
5113	000	5114	000	5115	000
5116	000	5117	000	5118	000
5119	000	5120	000	5121	000
5122	000	5123	000	5124	000
5201	000	5202	000	5203	000
5204	000	5205	000	5206	000
1613	000	1614	000	1615	000
1616	000	1617	000	1618	000
1619	000	1620	000	1621	000
1622	000	1623	000	1624	000

ALL ROOM STATUSES DUMPED

FINAL TRANSACTION CODE

TYPICAL ROOM-STATUS AUDIT PRINT-OUT.

DATE TIME TRANSACTION CODE

04119 14:21 ACTIVE DO NOT DISTURBS

5214 5408 5808 5810 1012

1313 1506 1607

ALL ACTIVE DO NOT DISTURBS DUMPED

FINALTRANSACTION CODE

TYPICAL DO-NOT-DISTURB AUDIT PRINT-OUT.

Table 9.1 Typical Print-outs (Continued)

DATE TIME TRANSACTION CODE

04/09 11:02 DO NOT DIST
5210 6

ROOM/STATION NUMBER DEVICE IDENTITY NUMBER •

TYPICAL STATION DO-NOT-DISTURB ACTIVATION PRINT-OUT.*

DATE TIME TRANSACTION CODE

05/10 07:02 TRY 2
5208 ANS

ROOM/STATION NUMBER STATION RESPONSE

TYPICAL ANSWER OF WAKE-UP ATTEMPT PRINT-OUT.

DATE TIME TRANSACTION CODE

04109 11:05 CLR DO NOT DIST
5210 6

ROOM/STATION NUMBER DEVICE IDENTITY NUMBER *

TYPICAL STATION DO-NOT-DISTURB CANCELLATION PRINT-OUT.*

DATE TIME TRANSACTION CODE

04119 14:18 RESTRICT
5508 6

ROOM/STATION NUMBER DEVICE IDENTITY NUMBER *

TYPICAL STATION-RESTRICTION ACTIVATION PRINT-OUT.*

DATE TIME TRANSACTION CODE

04/19 14:16 ENT WAKE-UP
1004 08:30 6

ROOM/STATION NUMBER WAKE-UP TIME DEVICE IDENTITY NUMBER'

TYPICAL STATION WAKE-UP ACTIVATION PRINT-OUT.*

DATE TIME TRANSACTION CODE

05/10 07:00 TRY 1
5207 RUNG

ROOM/STATION NUMBER STATION RESPONSE

TYPICAL WAKE-UP ATTEMPT PRINT-OUT.

DATE TIME TRANSACTION CODE

04/09 13:46 CLR RESTRICT
5208 6

ROOM/STATION NUMBER DEVICE IDENTITY NUMBER •

TYPICAL STATION-RESTRICTION CANCELLATION PRINT-OUT.*

* SEE NOTES AT END OF TABLE.

Table 9.1 Typical Print-outs (Continued)

NOTES:

Device Identity Number 0 is KEDU 0.

Device identity Number 1 is KEDU 1.

Device Identity Number 2 is KEDU 2.

Device Identity Number 3 is KEDU 3.

Device Identity Number 4 is Attendant Console.

Device Identity Number 5 is Administrative Phone.

Device Identity Number 6 is Room Phone.

Device Identity Number 7 is System-Initiated.

Device Identity Number 8 is Maintenance TTY.

Device Identity Number 9 is MDR.

9.4 KEDU Operating Procedures

The following paragraphs provide the instructions for processing KEDU-related features in the PABX. Table 9.2 provides the possible response codes and message meter readouts that can be received when performing the procedures. Table 4.1 provides a timetable for converting standard times into KEDU-related times.

9.4.1 Room Status Control and Display

This feature allows the KEDU to display the status of a room as follows:

(a) The room is available (1) or not available (0).

(b) The room is occupied (1) or not occupied (0).

(c) Maid service is required (3), maid service is in progress (2), maid service is complete (1), or maid service is unknown (0).

NOTE: This information is generally entered by the maid on duty from the room station.

STEP	OPERATION	RESULT
	Display Room Status	
1	Depress ROOM STAT button.	(a) ROOM STAT LED on. (b) KEDU display fields off. (c) ROOM NUMBER LED on. (d) DATA LED on.
2	Enter the desired four-digit room number. If a three-digit number, enter a # as the fourth digit.	(a) Room number is displayed in ROOM NUMBER display field. If it is a three-digit number, it will display "F" on the right-hand side of the display field; the room number will be displayed on the left-hand side of the display field. (b) Response code “A” is displayed in the response code field.

Table 9.2 KEDU Response Code Identification

FEATURE	FUNCTION	HOUR/MINUTE DATA	RESPONSE CODE	DEFINITION
Wake-Up	Enter room number	FFFF or time in memory	A	Valid response.
		Blank	1	Nonexistent room.
	Enter time	Time entered via keyset	A	Valid response.
		'Time entered via keyset	2	Invalid time.
		Time entered via keyset	3	Another time already in memory.
	Clear time from memory	FFFF	A	Valid response.
		FFFF	5	Room not on wake-up list.
Message Metering	Enter room number	Count in memory	A	Valid response.
		Blank	1	Nonexistent room.
	Enter meter count	Count entered via keyset	A	Valid response (O-255).
		Count entered via keyset	3	New count overwrites old count.
		Count entered via keyset.	2	invalid count (greater than 255).
	Clear meters to zero	000	A	System has cleared meter count.
		Blank	1	Nonexistent room.
		060	5	Meter count is already cleared.
Room Restriction	Enter room number	AAAA	A	Room restricted.
		BBBB	A	Room not restricted.
		Blank	1	Nonexistent room.
	Enter restriction	AAAA	A	Room restricted.
		AAAA	3	Room already restricted
	Clear restriction	BBBB	A	Restriction off.

Table 9.2 KEDU Response Code Identification (Continued)

FEATURE	FUNCTION	HOUR/MINUTE DATA	RESPONSE CODE	DEFINITION
		Blank	1	Nonexistent room.
		BBBB	5	Room not restricted.
Time	Enter date and time	Month, day, hour, & minutes	A	System accepts entry.
		Month, day, hour, &minutes	2	Invalid entry.
Message Waiting	'Enter room number	AAAA	A	Message waiting activated.
		BBBB	A	Message waiting not activated.
		BBBB	1	Nonexistent room.
Alarm indicator	After a power failure		F	Message metering and wake-up tape data bad.
			A	Message metering memory bad, wake-up tape good,
	With no power failure		A	Message metering and wake-up tape data good.
			F	System diagnosed wake-up and/or message metering data as bad.
Room Status	Enter room number	Current room status	A	Valid entry.
		Blank	2	Invalid entry.
	Clear status	000	A	System has cleared.
Do Not Disturb	Enter room number	AAAA	A	Room restriction activated.
		BBBB	A	Room restriction not activated.
	Clear do not disturb	BBBB	A	Room restriction deactivated.

MOTE: A response of D after any function keying indicates that the function is not allowed from the KEDU feature.

STEP	OPERATION	RESULT
		(c) DATA field displays a three-digit code, as follows: 1. Left-most digit (field H) is either 0 for an unoccupied room or 1 for an occupied room. 2. Center digit (field I) is either 0 for an unavailable room or 1 for an available room. 3. Right-most digit (field J) gives information on the status of maid service for that room. 0= Status of the room is unknown (no date). 1 = maid service is complete. 2 = maid service is in progress. 3 = maid service is required.
3	The next function performed clears the KEDU.	
	Enter or Change Room Status	
4	Depress ROOM STAT button.	(a) ROOM STAT LED on. (b) KEDU display fields off. (c) ROOM NUMBER LED on. (d) DATA LED on.
5	Enter room number	(a) ROOM NUMBER field displays the room number. (b) Response code field displays “A”: (c) DATA field displays the current status.
6	Depress ENTER/XCUTE button.	(a) DATA display field off. (b) Response code display field off. (c) Room number field displays the room number.
7	Enter new room status.	Fields H through J display the new room status.
NOTE:	Three values must be entered, even if only one value is to be changed.	
8	Depress ENTER/XCUTE button.	(a) DATA field displays new room status. (b) Response code field displays “A”. (c) Response code field displays “2” if invalid data is entered. (d) If printer provided, see Table 9.1 for typical printer displays.

STEP	OPERATION	RESULT
	Clearing Room Status Information	
9	Depress ROOM STAT button.	(a) ROOM STAT LED on. (b) KEDU display fields out. (c) ROOM NUMBER LED on. (d) DATA LED on.
10	Enter room number.	Same as Step 5.
11	Depress CLEAR button.	(a) CLEAR LED on. (b) Display fields H through K off. (c) Response code display field off.
12	Depress ENTER/XCUTE button.	(a) Response code "A" is displayed in the response code field. (b) DATA field displays 000.

9.4.2 Maid Service

Using a room telephone, the maid or supervisory personnel can indicate to the attendant or a clerk whether maid service is needed, in progress, or completed in the room.

STEP	OPERATION	RESULT
1	Take station off-hook.	Dial tone heard.
2	Enter room status change access code.	Confirmation tone heard.
3	Place station on-hook.	

NOTE: If you have this feature with Property Management System (PMS) equipped, you must enter the ID (1-6 digits) after the room status change access code before hearing the confirmation tone.

9.4.3 Wake-Up Call

When a guest requests a wake-up call, the system can be programmed to call the room automatically at the specified time (not to exceed 24 hours in advance). A tone will be heard or, optionally, a recorded announcement will be heard when the room answers. If the station is busy or does not answer, the system will:

1. Try a number of times at intervals before canceling the wake-up call with the number of times and the time intervals programmed in the data base.

2. Connect to the console or another station (data base programmable).

STEP	OPERATION	RESULT
	Enter Request for Wake-Up	
1	Depress WAKE-UP button.	(a) WAKE-UP LED on. (b) KEDU display fields off. (c) ROOM NUMBER LED on. (d) HOUR-MINUTE LED on.
2	Enter room number.	(a) Room number displayed in ROOM NUMBER display field. (b) FFFF displayed in the HOUR-MINUTE display field. (c) Response code field displays "A" if the room number is valid. (d) Response code field displays "1" if the room number is invalid.

NOTES:

1. Where room numbers are of three- and four-digit format, four digits must be entered. If a room number is of three-digit format, enter a # after the three digits.

2. If a time is displayed instead of FFFF, the time must be canceled before entering another wake-up time.

STEP	OPERATION	RESULT
3	Depress ENTER/XCUTE button.	(a) HOUR-MINUTE field off. (b) Response code field off.
4	Enter wake-up time.	Wake-up time is displayed in the HOUR-MINUTE field.
5	Depress ENTER/XCUTE button.	(a) Response field displays "A" if no previous wake-up time for this room number was entered. OR (b) Response field displays "3" if a previous wake-up time was entered for this room number. (c) If printer is provided, see Table 9.1 for typical printer displays.
	Cancellation of Previously Entered Wake-Up Call	
6	Depress WAKE UP button.	(a) WAKE-UP LED on. (b) KEDU display fields off.

STEP	OPERATION	RESULT
		(c) ROOM NUMBER LED on. (d) HOUR-MINUTE LED on.
7	Enter room number.	(a) Room number is displayed in the ROOM NUMBER field. (b) Data previously entered (wake-up time) shown on display field. (c) ROOM NUMBER LED on. (d) HOUR-MINUTE LED on. (e) Response code field displays “A”.
8	Depress CLEAR button.	HOUR-MINUTE field off.
9	Depress ENTER/XCUTE button.	(a) HOUR-MINUTE field displays FFFF. (b) The response code field displays “A”. (c) If printer is provided, see Table 9.1 for typical printer displays.
10	The next function performed clears the KEDU display.	

9.4.4 Message Waiting

The message waiting feature of the KEDU allows the attendant or administrator to flash the LED on a room telephone indicating that a message is waiting at the message center.

STEP	OPERATION	RESULT
	Initiate Message Waiting	
1	Depress MESG WTG button.	(a) MESG WTG LED on. (b) KEDU display fields off. (c) ROOM NUMBER LED on. (d) DATA LED on.
2	Enter room number. (If room number is three digits, enter a # after the third digit.)	(a) Room number is displayed in the ROOM NUMBER field. (b) DATA field displays message waiting status “BBBB” (not activated) or "AAAA" (activated). (c) Response code field displays “A”.
NOTE:	If “AAAA” is displayed, message waiting is already activated.	
3	If “BBBB” is displayed, depress ENTER/XCUTE button.	“BBBB” and response code “A” off.

STEP	OPERATION	RESULT
4	Depress ENTEFVXCUTE button.	(a) DATA field displays "AAAA" (message waiting activated). (b) Response code field displays "A". (c) If printer provided, see Table 9.1 for typical printer displays.
5	The next function performed clears the KEDU.	
	Cancel Message Waiting	
6	Depress MESG WTG button.	(a) MESG WTG LED on. (b) KEDU display fields off. (c) ROOM NUMBER LED on. (d) DATA LED on.
7	Enter room number. (If room number is three digits, enter a # after the third digit.)	(a) Room number is displayed in the ROOM NUMBER field. (b) DATA display field displays "AAAA" (message-waiting activated). (c) The response code field displays "A".
8	Depress CLEAR button.	DATA and response code display fields off.
9	Depress ENTER/XCUTE button.	(a) DATA display field displays "BBBB" (message waiting deactivated). (b) The response code field displays "A". (c) If printer is provided, see Table 9.1 for typical printer displays.
10	The next function performed clears the KEDU.	

NOTE: When the room occupant responds to the message-waiting indication using the message center access code, the flashing LED on the room telephone automatically goes off.

9.4.5 Message Metering

The Message-Metering feature allows the entry of a room number for recording outgoing calls. Recording can be by peg-per-completed-call or on a measured rate of service based on trunk group. This information is programmed into the data base for the specific switch and location.

STEP	OPERATION	RESULT
	Display Message Meter Status	
1	Depress MESG METER button.	(a) MESG METER LED on. (b) KEDU display fields off. (c) ROOM NUMBER LED on. (d) DATA LED on.
2	Enter room number. (Must be 4 digits. If 3 digits, enter a # after the third digit.)	(a) Room number is displayed in the ROOM NUMBER field. (b) DATA field displays the meter count (if any). (c) Response code field displays “A”.
3	The next function performed clears the KEDU.	
	Entering New Count in Message Meter	
4	Depress MESG METER button.	(a) MESG METER LED on. (b) KEDU displays off. (c) ROOM NUMBER LED on. (d) DATA LED on.
5	Enter room number.	(a) Room number is displayed in the ROOM NUMBER field. (b) DATA field displays the meter count or 000. (c) Response code field displays “A”.
6	Depress ENTER/XCUTE button.	(a) DATA display field off. (b) Response code display field off
7	Enter meter count (000 to 255).	DATA field displays the meter count.
8	Depress ENTER/XCUTE button.	(a) Response code field displays “A” if a valid meter count was entered. (b) Response code field displays “2” if an invalid meter count was entered. (c) if printer is provided, see Table 9.1 for typical printer displays.
9	The next function performed clears the KEDU.	
	Clearing Message Meters	
10	Depress MESG METER button.	(a) MESG METER LED on. (b) KEDU display fields off. (c) ROOM NUMBER LED on. (d) DATA LED on.

STEP	OPERATION	RESULT
11	Enter room number.	(a) Room number is displayed in the NUMBER field. (b) DATA field displays the meter count. (c) Response code field displays "A".
12	Depress CLEAR button.	(a) DATA field off. (b) Response code "A" off.
13	Depress ENTER/XCUTE button.	(a) DATA field displays "A". (b) The response code field displays "A". (c) If printer is provided, see Table 9.1 for typical printer displays.
14	The next function performed clears the KEDU.	

9.4.6 Audit Requests

An audit may be requested from a KEDU that has been assigned to a specific printer. The audit will list all stations that have information in order by room and floor, with appropriate title and column. The following reports may be requested depending upon data base programming:

Do Not Disturb	Room Restriction
Message Meters	Room Status
Message Waiting	Wake-up Calls

STEP	OPERATION	RESULT
1	Depress button associated with the feature to be audited.	LED associated with the feature to be audited goes on.
NOTE:	Wait about 20 seconds for print-out, if printer is provided.	
	initiate Audit Request	
2	Enter *** 0.	(a) ROOM NUMBER field displays "EEEO". (b) Response code field displays one of the following: 1. If the audit is initiated, "A" is displayed. 2. If the audit function is denied, "D" is displayed. 3. If the request failed, "F" is displayed.

STEP	OPERATION	RESULT
		(c) Printer will print the appropriate title and column heading followed by data.
	Abort Audit Request	
3	Enter *** 1.	(a) ROOM NUMBER field displays "EEE1". (b) Response code field displays one of the following: 1. If the audit is stopped, "A" is displayed. 2. If the abort audit function is denied, "D" is displayed. 3. If the audit is not in progress, "F" is displayed.
	Message Meter Audit with Memory Clear	
4	Enter *** 2.	(a) ROOM NUMBER field displays "EEE2". (b) Response code field displays one of the following: 1. If the audit is stopped, "A" is displayed. 2. If the abort audit function is denied, "D" is displayed. 3. If the audit is not in progress, "F" is displayed.

9.4.7 Do Not Disturb, KEDU-Provided

This feature allows the blocking of all calls to a room when requested by the room occupant. Calls to the do-not-disturb station are routed to reorder tone (120 IPM), a recorded announcement, or the attendant.

NOTE: An administrative station or the Attendant Console can activate or deactivate the do-not-disturb feature. The Attendant Console can also override this feature to reach the do-not-disturb party. Procedures are given in Sections: Do Not Disturb, Station-Initiated, Do Not Disturb, Attendant Console-Initiated, and Override a Do-Not-Disturb Line.

STEP	OPERATION	RESULT
	Initiate Do Not Disturb	
1	Depress DONT DIST button.	(a) DONT DIST LED on. (b) KEDU display fields off. (c) ROOM NUMBER and DATA LEDs on.
2	Enter room number. (If room number is three digits, enter a # after the third digit.)	(a) ROOM NUMBER field displays the room number. (b) DATA field displays "AAAA" (activated) or "BBBB" (not activated). (c) Response code field displays "A".
3	If BBBB is displayed, depress ENTEFVXCUTE button.	DATA and response code display fields off.
4	Depress ENTER/XCUTE button.	(a) DATA field displays "AAAA" (do-not-disturb activated). (b) Response display field displays "A".
	Clear Do Not Disturb	
5	Depress DONT DIST button.	(a) DONT DIST LED on. (b) KEDU display fields off. (c) ROOM NUMBER and DATA LEDs on.
6	Enter room number. (If room number is three digits, enter a # after the third digit.)	(a) ROOM NUMBER field displays the room number. (b) DATA field displays "AAAA" (do-not-disturb activated). (c) Response code field displays "A".
7	Depress CLEAR button.	DATA and response code display fields off.
8	Depress ENTEFVXCUTE button.	(a) DATA field displays "BBBB" (do-not-disturb deactivated). (b) Response code field displays "A".
9	The next function performed clears the KEDU display.	

NOTE: Automatic wake-up calls are not blocked by the activation of this feature. An access code also permits the attendant to reach the do-not-disturb party.

9.4.8 Room Restriction, KEDU-Provided

The KEDU allows individual room station users (non-administrative phones) to be restricted from Direct Outside Dialing (DOD).

NOTE: The Attendant Console and administrative telephone are also capable of providing the restriction feature. Procedures for these are covered in Sections: Room Restriction, Attendant Console-Initiated and Room Restriction, Administrative Telephone-Initiated.

STEP	OPERATION	RESULT
	Initiating Room Restriction	
1	Depress ROOM RSTR button.	(a) ROOM RSTR LED on. (b) KEDU display fields off. (c) ROOM NUMBER LED on. (d) DATA LED on.
2	Enter room number. (Must be 4 digits. If 3 digits, key in # after the third digit.)	(a) ROOM NUMBER field displays the room number. (b) DATA field displays “BBBB” (room restriction not activated). (c) Response code field displays “A”.
3	Depress ENTER/XCUTE button.	Data and response display fields off.
4	Depress ENTER/XCUTE button.	(a) Data field displays “AAAA” (room restriction activated). (b) Response code field displays “A”. (c) If printer is provided, see Table 9.2 for typical printer displays.
5	The next function performed clears the KEDU.	
	Cancellation of Room Restriction	
6	Depress ROOM RSTR button.	(a) ROOM RSTR button on. (b) KEDU display off. (c) ROOM NUMBER LED on. (d) DATA LED on.
7	Enter room number.	(a) ROOM NUMBER field displays the room number. (b) Data field displays “AAAA” (room restriction activated). (c) Response code field displays “A”.

STEP	OPERATION	RESULT
8	Depress CLEAR button.	(a) Data and response code display fields off. (b) CLEAR LED on.
9	Depress ENTER/XCUTE button.	(a) Data field displays "BBBB" (room restricted deactivated). (b) Response code field displays "A".
10	The next function performed clears the KEDU display.	

NOTE: Calls to the attendant, services, and administrative stations can still be made from the restricted station.

THIS PAGE INTENTIONALLY LEFT BLANK.

10.0 REFERENCES

The following reference documents complement/supplement the information provided in this Practice:

DOCUMENT	NUMBER	ISSUE	DESCRIPTION
FGBS Practices:	TL-130400-1001	1	Data Base
	TL-130000-1001	1	Features
	TL-130500-1001	1	System Configuration
	TL-130300-1001	1	installation
	TL-130200-1001	1	Maintenance
	TL-100000-1001	1	ADMP User's Guide
Hardware Ordering Guide:	FM-41 460	1	Ordering Guide
	FM-41 460-A	1	Ordering Sheets
Data Base Sheets:	FM-41 461	3/85	Software Programming Data Sheets (Instructions for completing the software programming data sheets are contained in Practice 278-921-030.)
Customer Instructions:	Cl-278-248	1	Executive Features Insert
	Cl-278-294	2	Busy Lamp Display Unit
	Cl-278-401	1	Attendant Manual Hotel/Motel-Healthcare Features Insert
	Cl-278-403	1	Administrative Station Hotel/Motel-Healthcare Features Insert
	Cl-278-409	1	OMNI Series Station User's Guide
	Cl-278-41 0	1	OMNI Series Attendant Manual
	Cl-278-41 1	1	Maid Service Features Insert
	Cl-278-41 2	1	OMNI Series Generic Station User's Guide (No Access Codes Included)
	Cl-473-335	1	Analog Featurephone User's Guide (FeatureComm III and IV) (This document is obsolete.)
	Cl-473-336	1	Analog Featurephone Programming Manual (FeatureComm III and IV) (This document is obsolete.)
	Cl-473-365	1	FeatureComm V/VI Handbook for CD-l 00 Data
	Cl-473-366	1	FeatureComm V/VI User's Manual for CD-1 00 Data
	Cl-473-397	1	OMNI FeatureComm Handbook (Voice Features Only)
	Cl-473-398	1	OMNI FeatureComm User's Manual (Voice Features Only)
	Cl-47351 9	1	FeatureComm Quick Reference Guide

THIS PAGE INTENTIONALLY LEFT BLANK.

INDEX

A

B

C

D

E

K

L

M

N

O

P

Q

R

S

T

U

Index

OMNI SI®

Purpose

This Technical Practice has been prepared for FGBS Employees who operate and maintain the equipment manufactured and sold by Fujitsu GTE Business Systems, Inc. The information in this Practice is subject to change and may not be suitable in **all** situations. Fujitsu GTE Business Systems, Inc. acknowledges that a customer's special requirements or practices may take precedence over those supplied in this Practice if a conflict develops during installation or ongoing operation. Fujitsu GTE Business Systems, Inc. hereby disclaims any responsibility or liability for any consequential or inconsequential damages that may result from the use of this practice.

OMNI SI® SVR 5210 Technical Practices

This FGBS Practice is part of a series of Practices for the FGBS OMNI SI, System Version Release 52.1 .O. The series includes the following:

TL-130000-1001	System Description/Features
TL-130100-1001	Operation
TL-130200-1001	Maintenance
TL-130300-1001	Installation
TL-130400-1001	Data Base
TL-130500-1001	System Configuration
TL-130600-1001	5120 to 5210 Upgrade
TL-130700-1001	Index
TL-100000-1001	PD-200 Data System

USER RESPONSE/REQUEST FORM

Send To: Fujitsu GTE Business Systems
Publications Manager, Dept. 443
2411 West 14th Street
Tempe, AZ 85281-9904

Date Submitted: ______________

- Document Number: TL-130700-1001
- Document Title: OMNI SI Index
- Issue Number: 1 Date: August 1987

☐ General Comments. Enter Comments concerning overall organization, presentation, or content of this document in the space below.

☐ Suggested Revisions. Enter specific additions deletions or changes to this document in the space below, or provide a marked up copy of the original document.

Paragraph, Figure, or Table Number   Comment

- Revision Priority ☐ 1: Service Affecting Problem
  ☐ 2: Potentially Service Affecting Problem
  ☐ 3: Administrative, Documentation, and Other Maintenance Problems
- Please attach a copy of your marked up pages.
- May we contact you concerning your comments? ☐ YES ☐ NO
- May we send you information on other available publications? ☐ YES ☐ NO
- Please enter your:

Name: ______________
Title: ______________
Company/Department: ______________
Address: ______________
Area Code/Tel. No./Ext.: ______________

Thank you for your cooperation and comments.

INDEX

A

B

C

D

E

F

G

H

I

K

L

M

N

P

Q

R

S

T

U

V

W

X

Z

NUMERICAL

System Configuration

OMNI SI®

Purpose This Technical Practice has been prepared for FGBS Employees who operate and maintain the equipment manufactured and sold by Fujitsu GTE Business Systems, Inc. The information in this practice is subject to change and may not be suitable in all situations. Fujitsu GTE Business Systems, Inc. acknowledges that a customer's special requirements or practices may take precedence over those supplied in this practice if a conflict develops during installation or ongoing operation. Fujitsu GTE Business Systems, Inc. hereby disclaims any responsibility or liability for any consequential or inconsequential damages that may result from the use of this practice.

OMNI SI® SVR 5210 Technical Practices

This FGBS practice is part of a series of practices for the FGBS OMNI SI, System Version Release 5.2.1 .O. The series includes the following:

TL-130000-1001	System Description/Features
TL-130100-1001	Operation
TL-130200-1001	Maintenance
TL-130300-1001	Installation
TL-130400-1001	Data Base
TL-130500-1001	System Configuration
TL-130600-1001	5120 to 5210 Upgrade
TL-100000-1001	ADMP User’s Guide
TL-130700-1001	Index

THIS PAGE IS INTENTIONALLY LEFT BLANK.

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

TABLE OF CONTENTS

THIS PAGE IS INTENTIONALLY LEFT BLANK

LIST OF FIGURES

LIST OF FIGURES

LIST OF FIGURES

LIST OF FIGURES

LIST OF FIGURES

LIST OF FIGURES

LIST OF TABLES

LIST OF TABLES

LIST OF TABLES

THIS PAGE IS INTENTIONALLY LEFT BLANK.

GENERAL

1.0 This technical practice describes the Fujitsu GTE Business Systems' OMNI SI Digital PABX (Private Automatic Branch Exchange) manufactured by Fujitsu GTE Business System's Inc SVR (System Version Release) 5210 incorporates packet switching data into system. This practice describes both voice and data:

- System hardware components
  - general description
  - operation overview
  - interaction
- Trunks, lines, and system capacities
- Ordering procedures
- Software subroutines

The system is an electronic, microprocessor based, PCM (Pulse Code Modulation) switching system that supports up to 256 lines or 64 trunks. It provides:

- High traffic capability
- Fully stored program control
- Full compatibility with all current industry standard systems, including
  - North American central offices
  - switched service networks
- The transfer of data by either
  - circuit switching
    or
  - packet switching (120 data ports)

The SVR (System Version Release) number provides a coded description that identifies the equipment system, hardware, data base, and generic patches. Table 1.1 showes the SVR number breakdown.

Table 1 .1 System Version Release 5210

5 = System	**2 = Version**	**1 = Release**	**0 = Point Release**
= NEW SYSTEM	= NEW HARDWARE	= NEW DATA BASE	= GENERIC PATCHES

NOTE: The SVR number provides descriptive information that identifies the equipment.

THIS PAGE IS INTENTIONALLY LEFT BLANK.

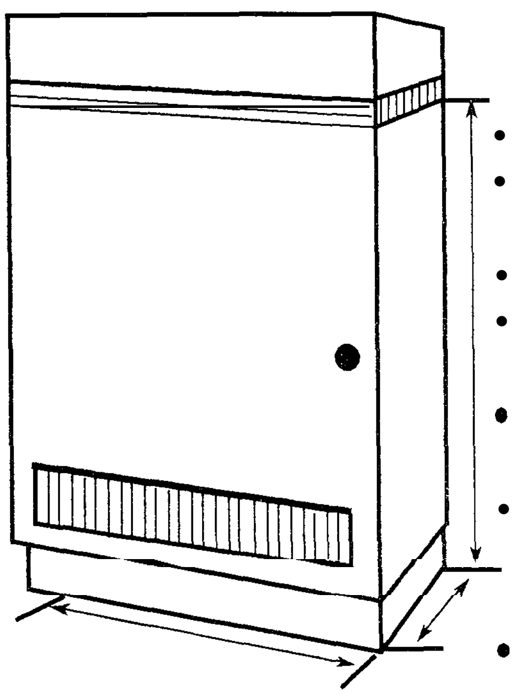
EQUIPMENT CABINET

2.0 The equipment cabinet (Figure 2.1) contains the following:

- System computer and memory
- Common control equipment
- Digital time-switch network
- Line, trunk, and feature circuits
- Optional CD-100 or PD-200 data switching hardware

The cabinet and internal chassis are constructed of high-grade steel. Universal cabinet doors can be removed for maintenance and provide for left- or right-hand door hinge mounting.

Cutouts and louvers provide cooling for the equipment cabinet. The cabinet openings, card file design, and high-efficiency power supply allow for convection cooling without fans. An access hole routes the cabinet cable to external equipment. Optional casters can be attached to the cabinet so that it can be moved during installation and maintenance.



- Single cabinet
- **24.44** in. L x 30 in. W **x 45.37** in. H
- Weight - 425 pounds
- Left or right-hand door hinge mounting
- Floor loading capacity-82 pounds per square foot
- Access holes in bottom of cabinet for system cable routing
- Cutouts and louvers for cooling

Figure 2.1 Equipment Cabinet

Equipment Cabinet Placement

2.1 A detailed description of cabinet placement and installation within the site can be found in TL-130300-1001.

Equipment Cabinet Cabling

2.2 Cable is routed through an access hole in the rear of the cabinet, near the cabinet base. The cable connects to the CDF (Combined Distribution Frame) at sites using typical central office methods. TL-130300-1001 provides information for cabinet and system cabling.

Cabinet Configuration

2.3 This paragraph provides configuration information for the system.

Contents of Lower Files

2.3.1 Located at the bottom of the equipment cabinet (Figure 2.2) is the following hardware:

- Power monitor transfer file, which contains
  - power fail transfer card
  - charging card
  - power supervisor card
  circuitry for alarm signals, circuitry for monitoring system status, and circuitry for monitoring power supplies of the equipment file
- Two power supplies
- Optional battery pack
- Ring generator
- Space for the floppy and hard disk drive assemblies

The remaining hardware consists of two equipment files. These contain the PCBs (Printed Circuit Boards) and the associated backplane assemblies that support the system's lines, trunks, features, and optional data hardware.

- Equipment files
  - Get Started File
  - Expansion File
  - PMT (Power Monitor Transfer) file

- Power Supplies
  - 1 for Get Started and PMT Files
  - 1 for Expansion File (optional)

- Battery Pack (optional)
  -available in 2-, 8-, or 14-hour holdup times; maintains memory integrity when AC power is lost.

- Ring Generator standard 20 watt

- Floppy Disk Drive
  -8 megabyte storage capacity
  -5.25 inch
  -device for loading system software and data base backup for both voice and data systems

- Hard Disk Drives
  -10 megabyte storage capacity
  -5.25 inch
  -device that contains loader generic software and data base software for both voice and data systems

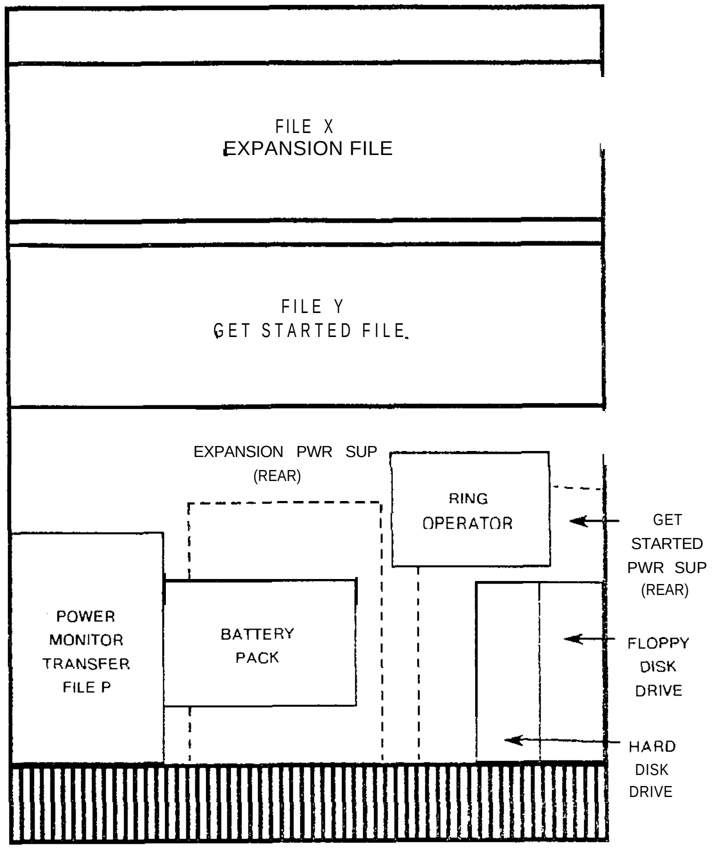


Figure 2.2 Equipment Cabinet Configuration

Get Started File

2.3.2 The Get Started File includes the following:

- Equipment that provides common system monitoring and control
- Digital time-switching network for all lines, trunks, and features used in the system
- Line, trunk, and feature cards as welt as their interface control

Expansion File

2.3.3 Figure 2.3 shows the Get Started and Expansion Files.

- The two files provide a total of: 256 lines, 64 trunks, 120 low speed data ports, 30 high speed data ports, or some combination thereof.
- The Expansion File includes the following:
  - equipment that provides common system monitoring and control for peripherals in its file
  - line, trunk, and feature cards as well as their interface control
  - the T1-span digital trunks and interface
- Interconnect circuitry provides communication from the Expansion File to the Get Started File.
- If an Expansion File is added to a system that has already been installed,. the system must be powered down before the Expansion File is added. Figure 2.3 shows the system configuration and Table 2.1 lists possible system configurations.

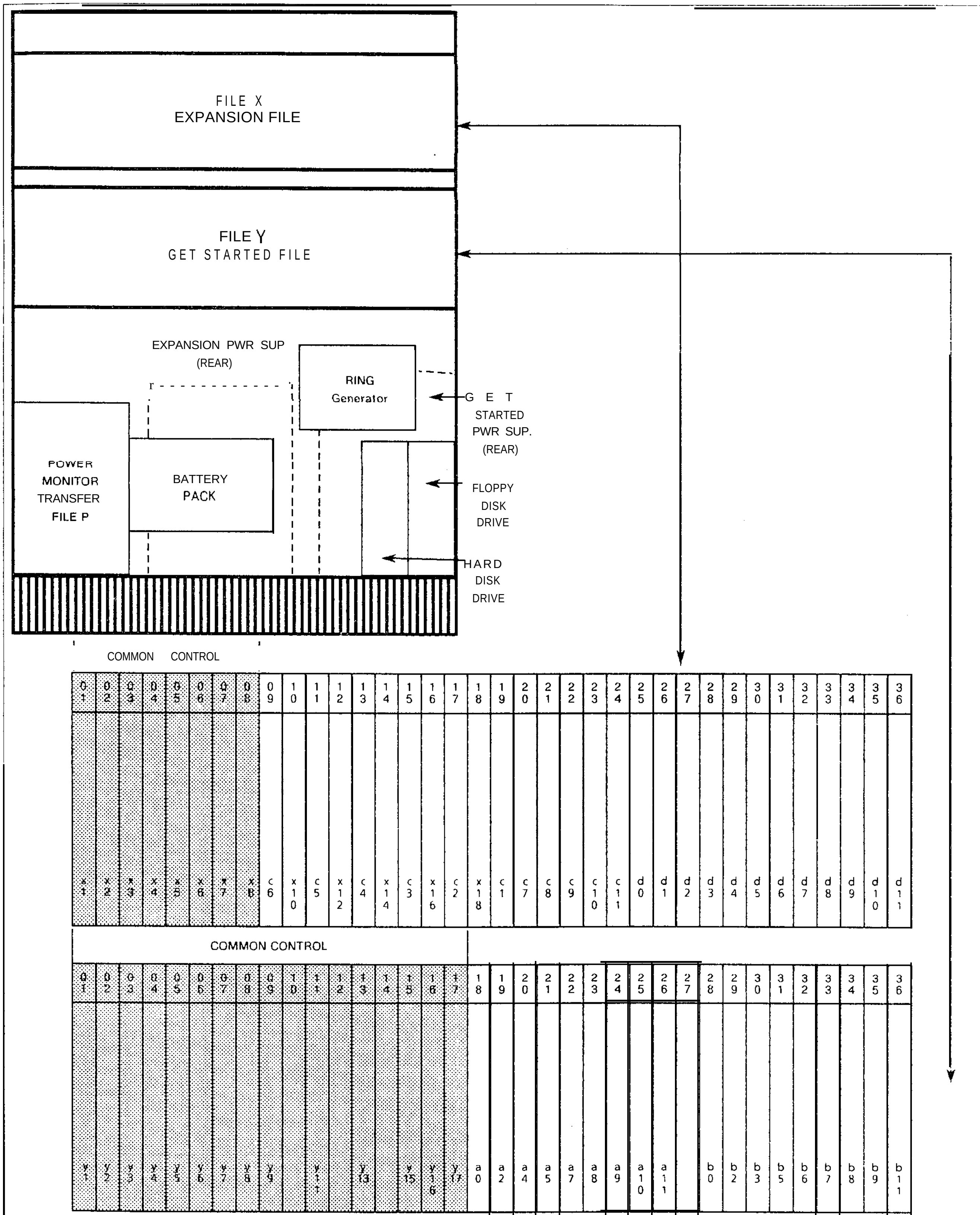


Figure 2.3 OMNI SI

Table 2.1 System Trade-Offs

Lines (Cards)	Trunks (Cards)
256 (32) **	32 (8)
248 (31)	32 (8)
240 (30)	36 (9)
224 (28)	40 (10)
216 (26)	40 (10)
192 (24)	48 (12)
176 (22)	52 (13)
160 (20)	56 (14)
144 (18)	60 (15)
136 (17)	60 (15)
128 (16)	64 (16) **
120 (15)	8 (2)
104 (13)	12 (3)
88 (11)	16 (4)
72 (9)	20 (5)
56 (7)	24 (6)
40 (5)	28 (7)
24 (3)	32 (8)
8 (1)	36 (9)
0 (0)	40 (10)
** = software limit	

Backplane Assemblies

2.3.4 The equipment cabinet contains the backplane assemblies for the Get Started, Expansion, and Power Monitor Transfer Files.

Get Started Backplane Assembly

Get Started backplane assembly supports the common control circuits and time-switch network for all the lines, trunks, and features used in the system. It also supports a limited number of lines, trunks, and data ports. The wire wrapped backplane is installed as the Y file in the cabinet and allows file equipment communications in the Y file.

Expansion Backplane Assembly

The Expansion File wire wrapped backplane is installed as the X file in the cabinet. This allows file equipment communications in the X file.

Power Backplane Assembly

The power backplane assembly contains circuitry for monitoring power supplies of equipment files, system status LEDs, and alarm signals.

Message Exchange

Communications between the files is provided for by an inter-file cable connection.

THIS PAGE IS INTENTIONALLY LEFT BLANK.

PRINTED CIRCUIT BOARDS (VOICE)

3.0 This section describes card slot locations and functions for system voice applications. It also provides the mnemonic, part number, type, and general function of the cards that can be installed in the system.

- Common control cards
- Tl-Span cards (digital trunks)
- Universal cards (trunks, lines and features)
- Power cards

Card Slot Locations

3.1 The equipment cabinet configuration provides for two equipment files. The Get Started File is located in file position Y. The Expansion File is located in file position X near the top of the cabinet. The X and Y files together are called a PEC (Peripheral Equipment Complex). In the OMNI SI, this PEC is always called PEC 0. The common control equipment occupies the first 17 card slots of the Get Started File. The line and trunk interface cards occupy the last 19 card slots of the Get Started File and most of the card slots in the Expansion File.

Most of the equipment installed into the equipment files consists of plug-in PCBs (Printed Circuit Boards) which are front-mounted into the card slots. An exception is the FB-20996-A Recorder Announcer Buildout Resistor (RABR) card, which mounts on the rear of the power transfer backplane assembly.

The width of the cards may require allocation of more than one card slot area for installing some cards. A card may occupy one or two card slot areas (the AIOD card needs three slots).

Card slot locations (Figure 3.1) are broken down as follows:

- The physical card slots within the Get Started and Expansion Files are identified by the numbers at the bottom of the files (01-36).
- The Get Started File (Y) is divided into software groups A and B.
- The Expansion File (X) is designed into software card groups, C and D.
- The hardware and software card slot designations are located at the bottom of the card slot and represent the slot number within the group (O-11).
- Card slots 01 through 17 of the Get Started File (file Y) and 01 through 08 of the Expansion File (file X) are reserved for common control cards.
- Card slots 09 through 19 of the Expansion File (file A or X) are reserved for T1-span (digital trunk) cards.
  - T1-span cards are double width, and the slots provided for them are designed to accommodate double width cards.

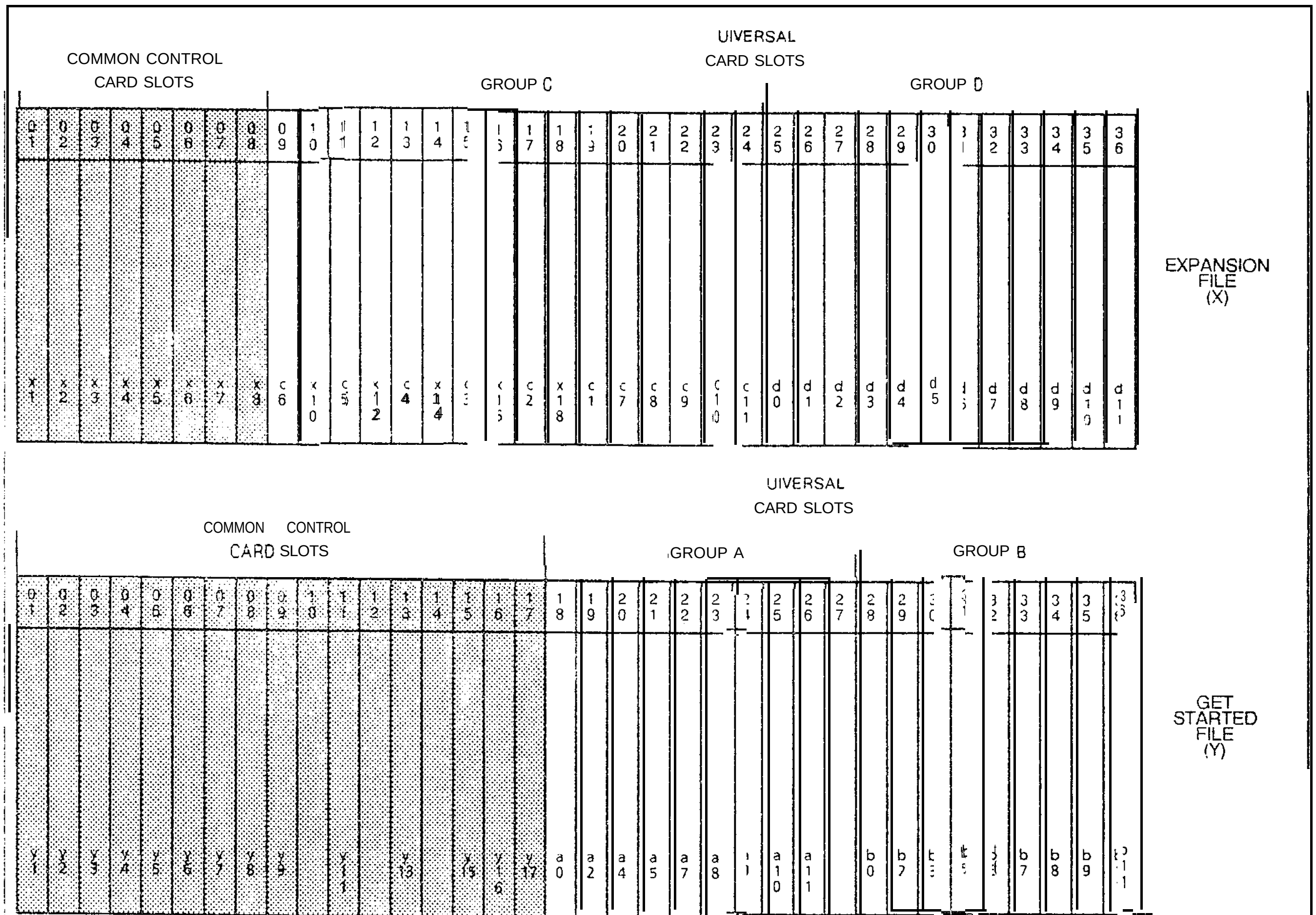


Figure 3.1 **Card Slot Locations**

- if a T1-span is not needed, then trunk, line, or feature cards can go here. it is recommended to put double width cards here.

- Card slots 18 through 36 of the Get Started File (file Y) and 09 through 36 of the Expansion File (file X) are reserved for trunk, line, or station cards. Slot 27 in the Get Started File (Y) is not available for use.

- Card slot A0 of the Get Started File (file Y) can only be used by a conference card or a dual tone multi-frequency card; this is because of the backplane configuration.

NOTE: if the PD-200 data option is equipped. card slot A0 can house a data card used to extend or terminate an LPB (Local Packet Bus).

PCMUSs (Pulse Code Modulation Universal Slots)

3.2 (Figure 3.2) Circuits that provide connections to lines are housed on line cards (eight line circuits per card) and circuits that provide connections to trunks are housed on trunk cards (four trunk circuits per card). Line cards provide connectivity between peripheral devices and the system. Trunk cards provide connectivity between the system and the Telco. Feature cards provide special capabilities to the system e. g., DTMF to rotary dial conversion, paging, connection to a music source, etc. The system has 40 card slots that can support line, trunk, or feature cards (data cards will be discussed separately). These card slots are referred to as PCMUSs. The 40 PCMUSs in the system are divided into six PCMUS groups. The physical relationships of the files, the PCMUS groups, and the card slots are shown in Figure **3.4.**

NOTES:

- Card slot A0 is not counted in the 40 PCMUS total.
- Slot A0 is reserved for certain cards; including this slot provides a total of 41 PCMUSs.

Several types of PCBs can be installed in a PCMUS. Each occupies one or two card slots. The number of line, trunk, or feature cards that can be mounted in a given PCMUS group is physically limited by the number of slots in that group and the number of slots that a given card occupies.

This paragragph provides the part number, mnemonic, name, function, and location of the PCBs that are installed within the PABX.

T1

Slot	EXPANSION FILE X	GET STARTED FILE Y	POWER FILE
01	CHM85	M1MB	PSUPPY
02		MPG16	
04	MPB85	CP85E	PFT
05	PCMFS	MPB85	
06	PCMFS	NSDC	BC5R
07	PCMI	FMSD	
08	PCMI	TPI2	
09	PCMUS C6	EPCMN	
10	FDC		
11	PCMUS C5	INCKS	
12	LCM		
13	PCMUS C4	PCMTS	
14	SIL		
15	PCMUS C3	CHM85	
16	T1S	PCMFS	
17	PCMUS C2	PCMI	
19	PCMUS C1	PCMUS A2	
20	PCMUS C7	PCMUS A4	
22	PCMUS C9	PCMUS A7	
23	PCMUS C10	PCMUS A8	
25	PCMUS D0	PCMUS A10	
26	PCMUS D1	PCMUS A11	
27	PCMUS D2		
28	PCMUS D	PCMUS B0	
30	PCMUS D5	PCMUS B3	
31	PCMUS D6	PCMUS B5	
32	PCMUS D7	PCMUS B6	
33	PCMUS D8	PCMUS B7	
34	PCMUS D9	PCMUS B8	
36	PCMUS D11	PCMUS B11	

Figure 3.2 Frame Image Card Locations for a Fully Configured System

Pulse Code Modulation Slot Groups

3.3 The PCMUSs of the system are divided into groups. The Get Started File accommodates two groups, groups A and B. The Expansion File differs in that its two groups, C and D, are each subdivided into two groups. Card slots C1 through C6 and C7 through C11 are the subgroups of C group. Card slots D0 through D5 and D6 through D11 are the subgroups of D group (see Figure 3.3).

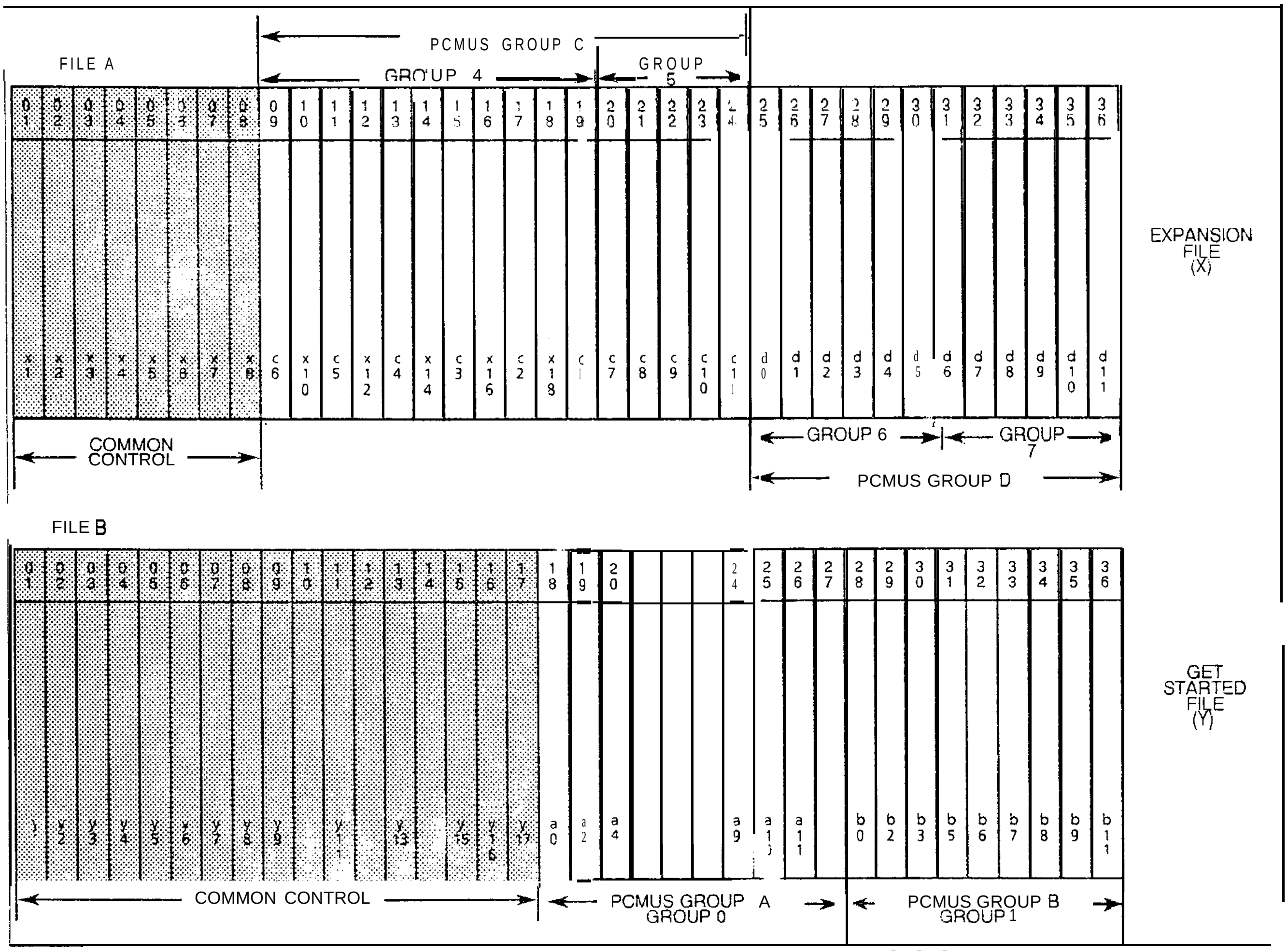


Figure 3.3 Pulse Code Modulation **Group Division**

Common Control Printed Circuit Boards

3.4 This paragraph describes the PCBs that are installed within the common control portion of the files. Common control cards are responsible for call control and call processing. Information concerning whether the PCB is needed to support a system option or a capability requirement is also included. Figure 3.4 shows the dedicated card slots used for common control. Table 3.1 lists the FB (Functional Board) number, mnemonic, PCMUS card function, and slot location of the common control cards.

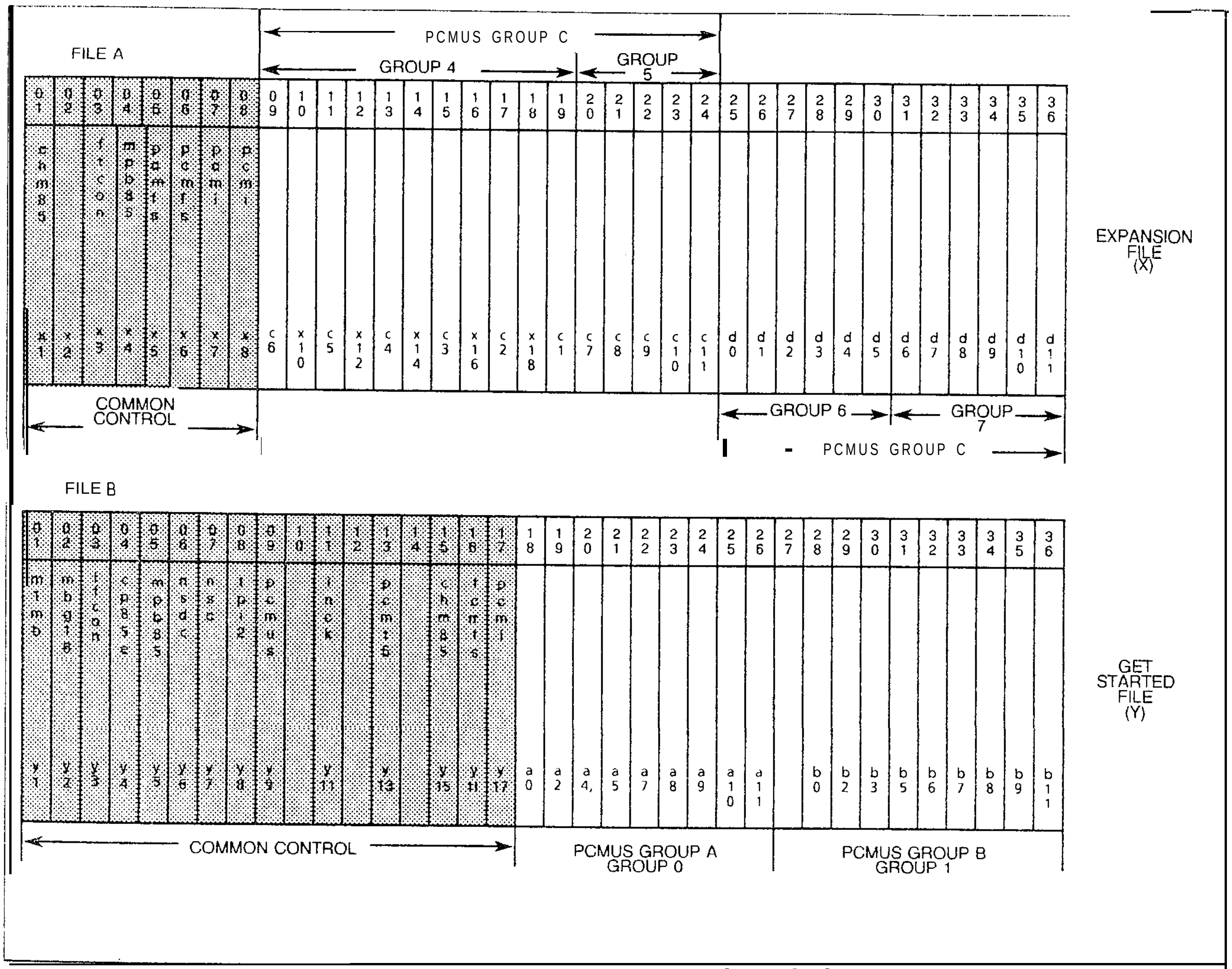


Figure 3.4 **Common Control Card Slots**

Table 3.1 Common Control Cards

FB#	MNE-MONIC	PCMUS CARD FUNCTION	SLOT LOCATION
FB-17218	CHM85	Channel Memory-8085	X/01 and Y/15
FB-17314	M1MB	1 Meg X 8 Bit Memory Card	Y/01
FB-17213-80	MPG1 6	Memory Paging- 3085	Y/02
FB-17224	IFCON	Interfile Connector	X/03 and Y/03
FB-17288	CP85E	Central Processor Unit-8085 Enhanced	Y/04
FB-17215	MPB85	Multiprocessor Buffer-8085	X/04 and Y/05
FB-20992	NSDC	Narrow Serial Device Controller	Yi06
FB-17220-BO	FMSD	File Management System	Y/07
FB-17188	TP12	Test Panel Interface-Version 2	Y/08
FB-17217	EPCMN	Expanded PCM Network	Y/09
FB-20922 or FB-20771-l	INCK	Network Clock	Y/11
FB-17189	PCMFS	PCM Frame Synchronization	X/05, X/06 & X/1 6
FB-17187	PCMI	PCM Interface	X/07, X/08 & Y/1 7
FB-20974	PCMTS	Pulse Code Modulation Tone Source	Y/13

CHM85
FB-17218-A

3.4.1 Each file in the system requires a CHM85 (Channel Memory) card. Channel memory holds the identity of the equipment interface cards and the circuit on each card being used in a call.

Each CHM85 card contains a RAM which stores the hardware IDs of the line, trunk, and feature circuitry that have been assigned channels within the system. Ninety-six time slots (four 24-channel groups) are available per CHM85 card:

- The CHM85 card stores hardware IDs. Slot X/01 is for PCM groups 4, 5, 6 and 7; 96 time slots are stored in this card.
- Slot Y/l 5 is for PCM groups 0 and 1; 48 time slots are stored in this card.

When time slots are assigned to the interfaces, the channel memory controls the time slot related sampling signals to the interface cards. Channel memory is addressed via the CPU address bus. Intercard communications data is transferred between the channel memory and the CPU via the CPU data bus.

Each CHM85 card functions as a PCM (Pulse Code Modulation) highway between its PCM groups and the PCM time-switch network EPCMN (Expanded PCM Network) card for data transfer.

MEMORY CARD

M1MB FB-17314-A 3.4.2 The CPU 1 Meg x 8 bit Memory card provides 1 Meg by 8 bits of system RAM. It is accessed by the CP85E (FB-17288-A). The system generic and data base software are loaded onto this card during system initialization. The software is stored within sixteen 64K pages

M512B FB-17193-A 512K

4 INST PAGES (0 - 3)

- Eight instruction pages: I0 through I7
- Eight data pages: D0 through D7

4 DATA PAGES (D0 - D3)

The 16 pages of memory are addressed via the MPG16 (paging card) and the address/control buses that originate at the microprocessor. Data transfers are made across the 8-bit data out/data in buses within the microprocessor complex. Data transfer between the CPU and the Ml MB is routed via the MPG 16.

MPG16 FB-17213-BOA 3.4.3 The Memory Paging card performs the following functions:

- Buffers data, address, read, write, strobe, and memory reference signals between the CPU and the M1 MB.
- Buffers the address, control and data out buses between the CP85E and the M1MB memory.
- Refreshes the CPU RAM.
- Refreshes Ml MB memory.
- Increases the addressable memory space from 64K to 1 Meg. This is done by using a paging method, based on processor instruction analysis, to select one of 16 physical pages (of 64K bytes) fcr processor access.
- Performs memory paging of up to 16 pages within the M1MB memory (1 page = 64K x 8 bits).
- Permits direct memory access by peripherals (Type 200 Test Set).

IFCON FB-17224-A 3.4.4 The Interfile Connector card is required only when the Expansion File is installed in the field. It provides a circuit-to-circuit connection and interfaces the microprocessor buses between the Get Started and Expansion File backplanes.

CP85E
FB-17288-A

3.4.5 The Central Processor Unit card contains an INTEL 8085 microprocessor and associated clock, control, interrupt, and I/O (Input/Output) interface circuits. The 8085 microprocessor provides both normal switching and MDR functions. The card controls all operations within the system including:

- Path connect/disconnect within the network
- Data transfer to/from system maintenance terminals
- Data transfer to/from system disk drive
- Communications to/from system peripherals

The 8085 makes decisions based on the various inputs including an 8-level system interrupt structure. It controls the operations such as peripheral interface scanning, data sampling routine, and message transfers. This card contains a 512 word by d-bit ROM (read-only memory), which contains the system bootstrap loader and CPU self-test software. The card also contains the system watchdog timer circuitry as well as circuitry for addressing data and control signals. Communication with other subsystems is through a 16-bit address bus, an 8-bit data bus, and control signal lines.

MPB85
FB-17215-A

3.4.6 Each file in the system requires a Multi-Processor Buffer card. This MPB85 card enables the equipment contained in the Get Started File or the Expansion File. Once enabled, the equipment receives control and sensing signals from the CPU and the PCBs within the PCM universal slots:

- Circuitry on MPB85 in the Get Started File performs control/sensing functions for groups A and B.

- Circuitry on MPB85 in the Expansion File performs control/sensing functions for groups C and D.

MPB85 provides interfacing circuitry to the T1 span circuitry, which can be equipped in group C of the Expansion File.

NSDC
FB-20992-A

3.4.7 The Narrow Serial Device Controller card is optional. The NSDC provides two independent serial interface ports that are cable-connected to modems and/or terminals external to the equipment cabinets. The NSDC is addressed via the CPU address bus, and data is transferred between the CPU and NSDC via the CPU data bus. Data transfers can be programmed as I/O or interrupt-driven. Each port can be configured for current loop or EIA (W-232-C) operation. Full-duplex, synchronous, and asynchronous communication is allowed. Typically, the NSDC card is connected via a cable to the system maintenance terminal.

FMSD
FB-17220-B0A

3.4.8 The File Management System Data card provides an intelligent serial data interface between the floppy disk drive, the hard disk drive, the PD-200's ADMP, and the CPU. Data and control signals are cable-connected between the floppy disk drive and FMSD. The FMSD is addressed via the CPU address bus. Data is transferred between the CPU and the FMSD via the CPU data bus. This card provides DMA (Direct Memory Access) and interrupt control to the CPU.

The FMSD contains the following major circuitry:

- Floppy disk control interface logic
- SASI bus interface logic
- DMA interface logic
- 8085 microprocessor, on-board memories, and associated clock,

TPI2
FB-17188-A

3.4.9 The Test Panel Interface card permits communication between the system CPU and the optional Type 200 Test Set. Serial, address, data, and control signals are cable-connected between the TP12 and the test set. The TP12 is addressed via the CPU address bus. Data is transferred between the CPU and the T PI2 via the CPU data, address, and control buses. A manual switch on the test set generates an interrupt signal that is sent to the CPU via the TPI2 when the test set is activated.

TL-1 30200-l 004 describes the test set operation.

EPCMN
FB-17217-A

3.440 The Expandable Pulse Code Modulation Network card is a time-switch network that provides connections for the communications paths. The EPCMN contains five separate 256 x 8-bit RAM memories: Control A Memory, Control B Memory, Information Memory, Pad Memory, and Interconnect Memory.

- Information memory temporarily stores voice, data, and tone PCM samples until a time switch is made.

- Control memory temporarily store time slot interchange data, used to address the information memory to initiate the time slot interchange.

- Pad memory controls the dB level of attenuation applied to the PCM output from the time-switch network, determines whether information memory is to be enabled during this time slot, and also controls speaker B memory used in a three-way conversation.

INCKS
FB-20922-A

3.4.11 The Synchronizable Intermediate Network Clock card (Figure 3.5) provides the basic timing pulses to the system network and PCM associated circuitry. The INCKS card permits the system to communicate over T1 trunks. In this case, the system is slaved to the far end of the span. The INCKS card also contains the network time slot counter.

The INCKS card receives control inputs from, and outputs clock pulses to, the EPCMN card.

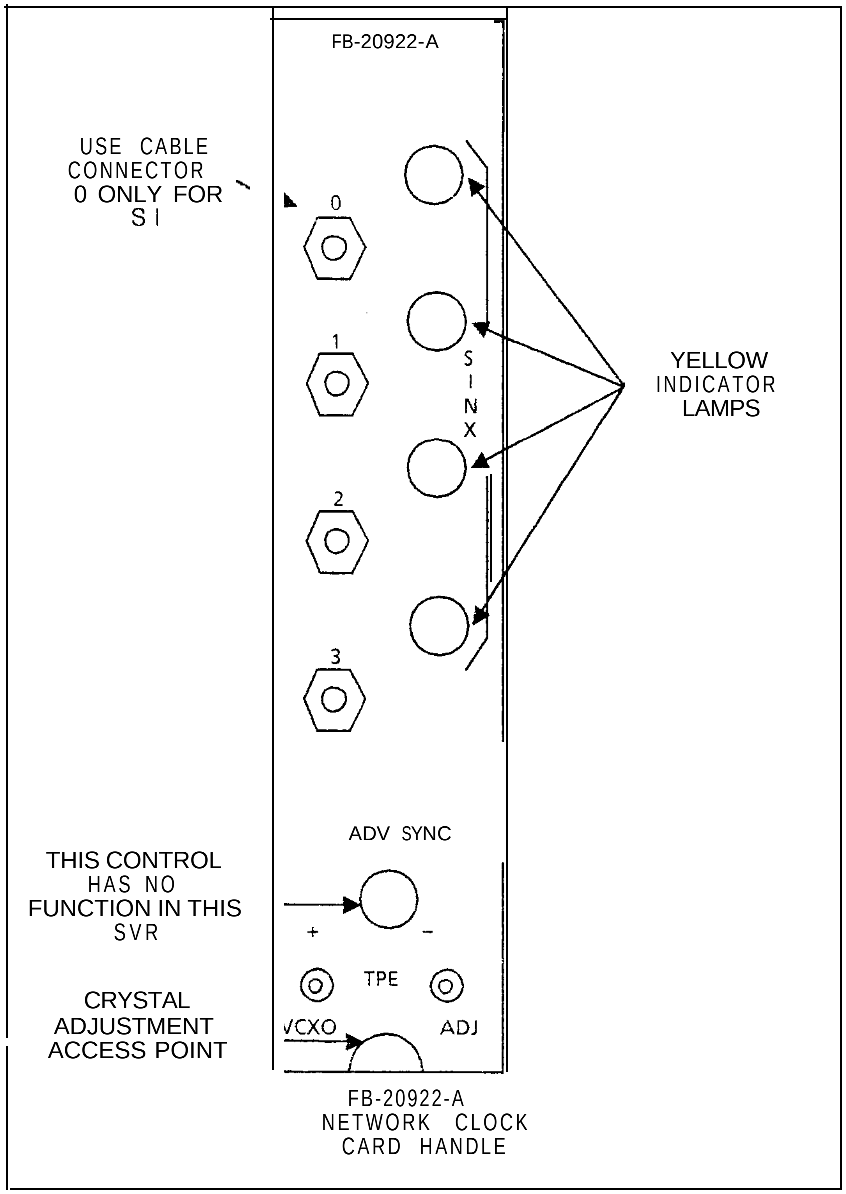


Figure 3.5 INCKS Card Handle View

INCK
FB-20771 -IA

3.4.12 The Intermediate Network Clock card provides all the functions of the INCKS card, except that T1 synchronization is not possible. This card is used if no Tl functions are used in the system.

NOTE: Use FB-20922 network clock when network timing is derived from a T1 span (slave operation).

PCMFS
FB-17189-A

3.4.13 Each 48 channels in the system requires one Pulse Code Modulation Frame Synchronization card. The PCMFS provides frame synchronization for the PCM line and trunk cards.

Each PCMFS card receives hardware IDs from its respective CHM85 card and decodes each hardware ID into a discrete circuit selection, enabling the transfer of PCM to/from that circuit.

- PCMFS in X/05 provides circuit selection for groups 4 and 5.
- PCMFS in X/O6 provides circuit selection for groups 6 and 7.
- PCMFS in Y/l 6 provides circuit selection for groups 0 and 1.

NOTE: Card location in slot X/06 is required when using group D.

One PCMFS card is located in file Y. This card accommodates the 48 channels available in the file. In file X, groups C and D each supply 48 channels: therefore if both groups are in use, two cards are required.

PCMI
FB-17187-A

3.4.14 The Pulse Code Modulation Interface card performs serial-to-parallel and parallel-to-serial conversion of system PCM signals.

An 8-bit parallel PCM signal from the time-switch is converted to a serial PCM signal (24 channels in a 125-microsecond frame) for transmission to digital line cards. The card also processes signals in the reverse direction. The PCMI contains two complete converter circuits which can handle a group of 24 channels each.

- PCMI in X/07 performs the conversion processes for groups 4 and 5.
- PCMI in X/08 performs the conversion processes for groups 6 and 7.
- PCMI in Y/17 performs the conversion processes for groups 0 and 1.

NOTE: The card located in slot 07 is required only when using Group D.

PCMTS
FB-20974-A

3.4.15 The Pulse Code Modulation Tone Source card stores samples of the various system tones within a ROM (Read-Only Memory). Tones read from ROM are transferred to the information memories for use throughout the system. This provides the dial tone, ringback tone, and other alerting tones used in the system.

The PCMTS receives control inputs from the Network Clock INCKS or INCK card. The PCM tone sources are stored in ROM on the PCMTS and, when selected, are applied as data inputs to information memory on the EPCMN card.

FD-1070-AY
(full height)
Hard Disk Assembly

3.4.16 Location: File D, Slots 16-25. The hard disk assembly consists of a 5-1/4 inch, 10 megabyte rigid disk that utilizes Winchester technology, a disk drive controller card, and the disk mount and associated cable assemblies. The hard disk assembly contains loader, generic, and data base software. The software and data base on this disk are loaded into the CPU via the FMSD card during system initialization and reload procedures. PD-200 software and data base are loaded into the PD-200 devices via the FMSD card and the ADMP card during PD-200 initialization/reload procedures.

FD-1070-BD
(half height)

3.4.17 Provides the same function as the FD-1070-AY hard disk assembly.

FD-1070-BA
Floppy Disk Assembly

3.4.18 Location: File D, Slots 26-31. This 5-1/4 inch, 1-megabyte floppy disk assembly is used to SYSGEN the hard disk during startup. It also provides back-up for the system.

Digital Trunking Printed Circuit Boards

3.5 This paragraph provides information about PCBs that are installed within the T1 span digital trunk preferred location. In this system, only one T1 span, which has a maximum of 24 channels, can be installed. Only file X can be used for this type of configuration and only certain slots within that file can be used by these cards. Not all T1 span cards require two card slots. Card slots C6, C5, and C4 may be used even if a T1 span is implemented. However, 4 channels are lost for every T1 span placed. Each card used for a T1 span requires two slots and is referred to as a double height card. Figure 3.6 shows the cards used by the T1 span. Table 3.2 lists the dedicated card slots used for the T1 span.

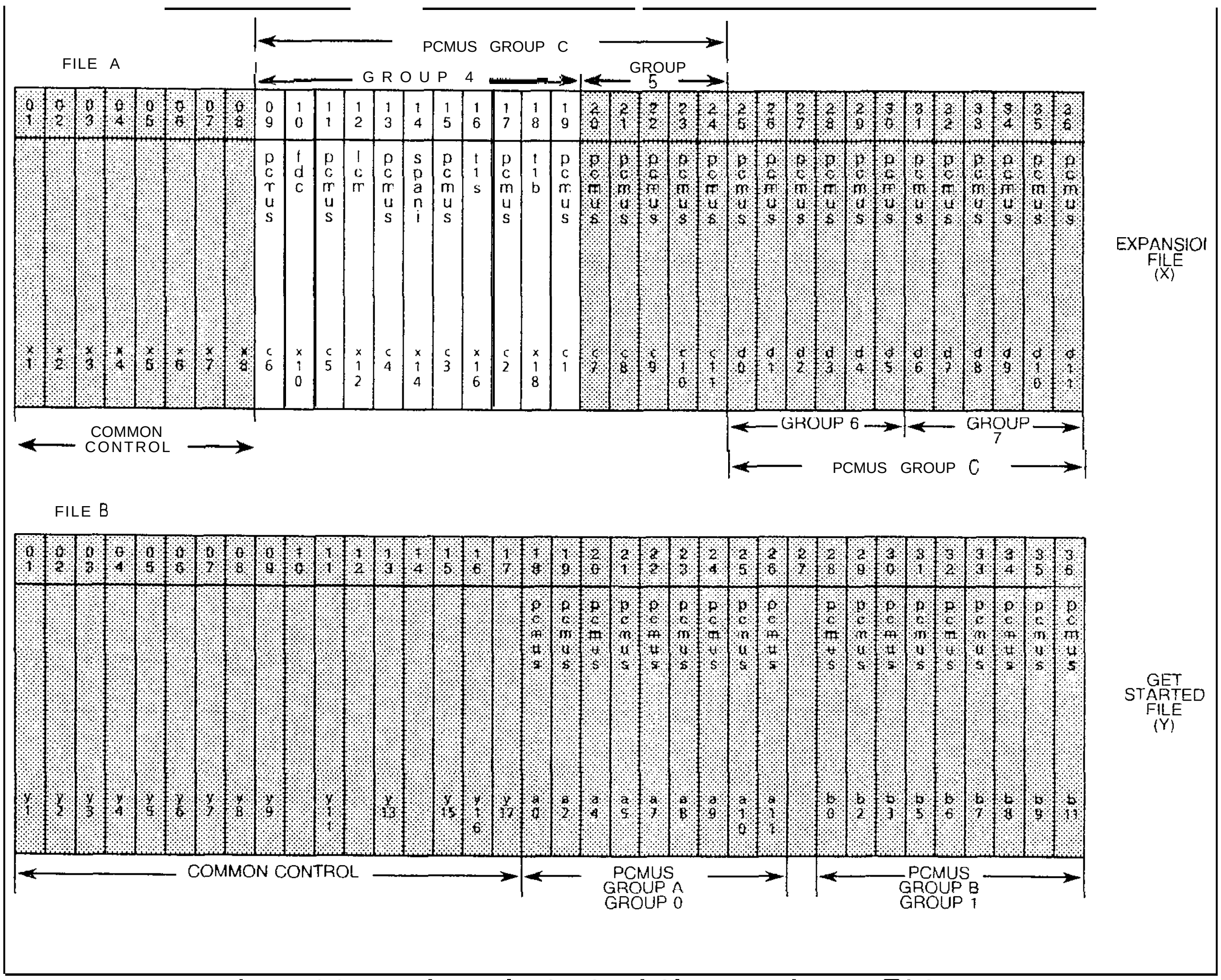


Figure 3.6 Universal PC5 Card Slots Used By A T1 Span

Table 3.2 T1 Span Cards

Card Number Mnemonic and Name	Card Slot
FB-15278-A, FDC, Frame Detector Card	X/10
FB-15280-A, LCM, Line Compensator Card	x/12
FB-17277-A or FB-15277-1, SIL, Span Interface Card	X/14
FB-20718-l A, T1 S, T1 -Type Supervisory Card	X/16
FB-17192-A, T1 B2, T1 Buffer Card	X/18

FDC
FB-15278-A

3.5.1 The Frame Detector card is required for the T1 span and provides the following functions:

- Monitors for errors in framing synchronization patterns
- Generates a framing alarm signal to the Tl-Type Supervisory card when three or more bit-pattern errors are found out of five incoming synchronization bits examined
- Signals the span interface when a new frame of voice samples is to arrive
- Signals the arrival of bit 2 (the second most important bit) to the Span Interface card
- Generates the supervisory frame signal that decodes channel A and B signaling

LCM
FB-15280-A

3.5.2 The Line Compensator card is required for the T1 span. The LCM provides buffering to compensate for propagation delays due to temperature changes over the span. The card can store two PCM frames that support the compensation process.

The LCM card receives a serial unipolar bit stream from the SIL (Span Interface cardj. This stream is converted to an 8-bit voice sample that is forwarded to the Tl-Type Buffer card in parallel format.

SIL
FB-17277-A

FB-15277-G

3.5.3 The Span Interface card (Figure 3.7) is required for the T1. span. The SIL receives the incoming bipolar signai and converts it to a unipolar bit stream which is then sent to the LCM. The SIL also works in reverse to prepare a bipolar signal for transmission over the T1 span. A strapping field is provided on the card for application configuration at installation (see TL-130200-1001). The SIL provides the looping ability to test the framing synchronization of the digital cards.

NOTE: To synchronize the PABX digital network timing to the T1 -span timing, use FB-15277-l.

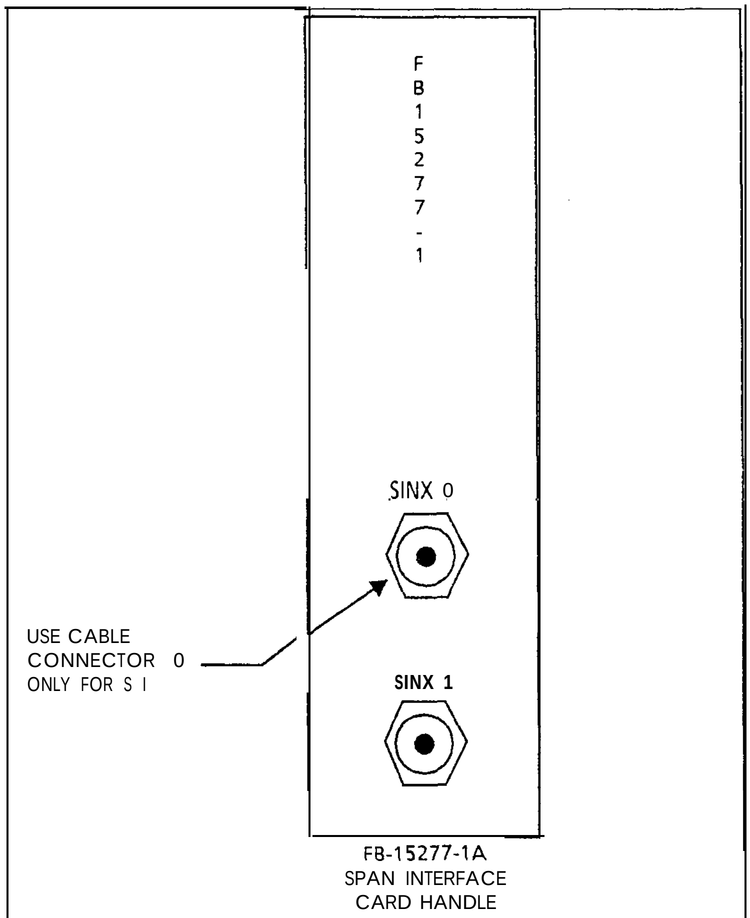


Figure 3.7 SIL Card Handle View

T1S
FB-20718-1A

3.5.4 The Tl-Type Supervisory card is required for the T1 span. It provides a supervisory signal interface between the system and the T1 span. The buffers on the T1S card retain the status of sense and control points.

The T1S card has a program board which can be strapped to decode FX trunk signals or E&M trunk signals. Strapping is also provided to change from D2 to D3 signaling formats, as well as to provide a variable framing alarm delay time. Strapping option procedures are described in TL-130300-1001.

The T1S card handle, shown in Figure 3.8, contains the following lamps and switches:

- Local alarm (LOC) lamp tha. lights when the framing of the incoming bipolar signal is lost (indicating a misframe: loss of framing synchronization).

- Remote alarm (REM) lamp that lights when the second bit of an incoming bipolar stream is inhibited for 1.32 to 1.44 seconds.
- System alarm (SYS) lamp that lights when any alarm condition exists, including when the system is fully frame-synchronized but in a loop mode.
- Remote power failure alarm (RPF) lamp that lights when a power failure occurs in the office-terminating shelf.
- Alarm cutoff (ACC) lamp that lights when the alarm cutoff switch and the loop test switch are activated.
- Two-position alarm cutoff (ACO) switch; UP is the activated position.
- Loop switch (LP) lamp that indicates when the loop test and alarm cutoff switches are activated.
- Two-position loop (LPT) switch that must be activated along with the alarm cutoff switch to start the loop test, UP is the activated position.

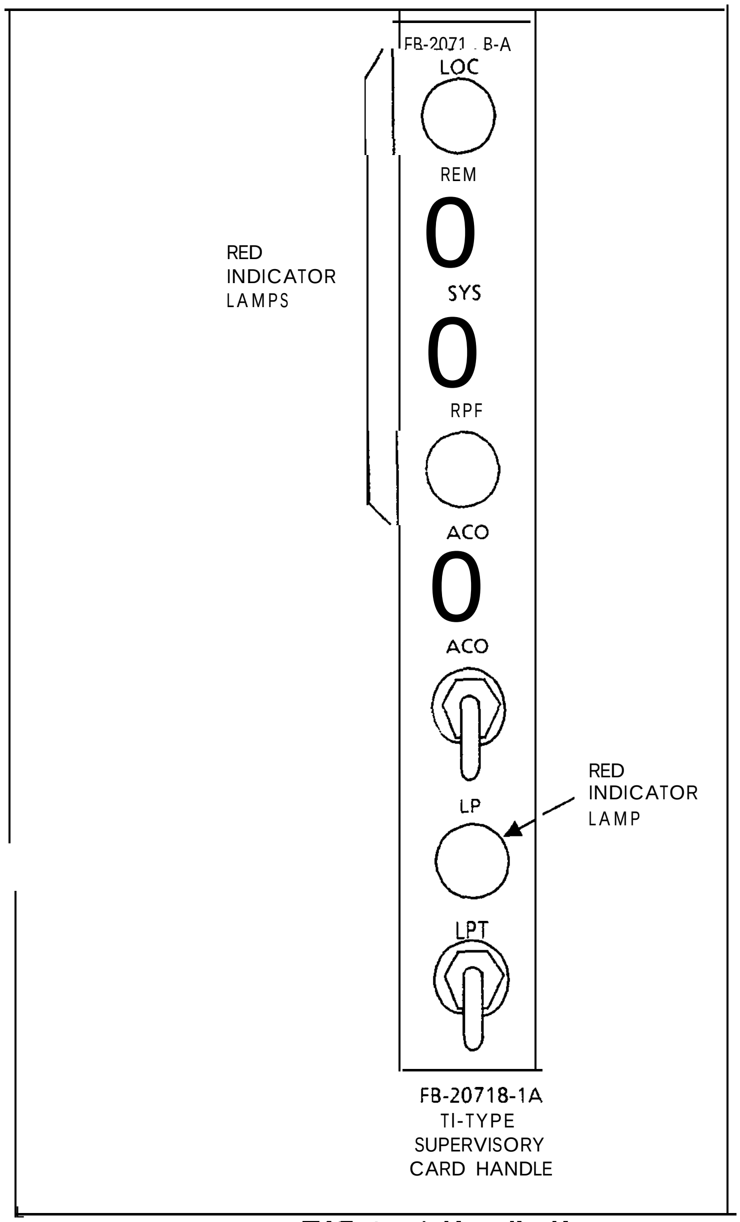


Figure 3.8 T1S Card Handle View

T1B2
FB-17192-A

3.5.5 The T1 Buffer card is required for the T1 span. It provides a buffer between the incoming PCM data from the Line Compensator card and the digital time-switch network in the system. It also buffers the outgoing PCM data from the digital time-switch network to the Span Interface card. It will synchronize and align the 24 PCM channels between the digital network and the Tl digital trunk interface.

Universal PCBs

3.6 This paragraph provides information about the PCBs installed in the remaining card slots (slots not used by common control cards). Slots used by these cards (which support lines, trunks, or features) are called PCMUSs (Pulse Code Modulation Universal Slots). PCMUSs are not reserved for a certain type of card, but can be used by any card needed to support the system configuration. Each card may occupy one, two, or three card slots. The number of line, trunk, or feature cards that can be mounted in a given PCMUS group is physically limited by the number of slots in that group and the number of slots occupied by a given card.

NOTE: When a T1 span is not used or not fully used, universal cards can be put into the T1 span slots. If a T1 span of less than maximum configuration is used, do not place cards that use time slots into these slots. Time slots are explained in paragraph 4.3.1.

Figure 3.9 shows the cards slots that are used by line, trunk, or feature cards. Table 3.3 lists the cards, and Table 3.4 lists the number of slots needed for each card.

NOTES:

- For information on maximum number of card types per system, see section 10.0.

- PD-200 data cards are described in section 7.0.

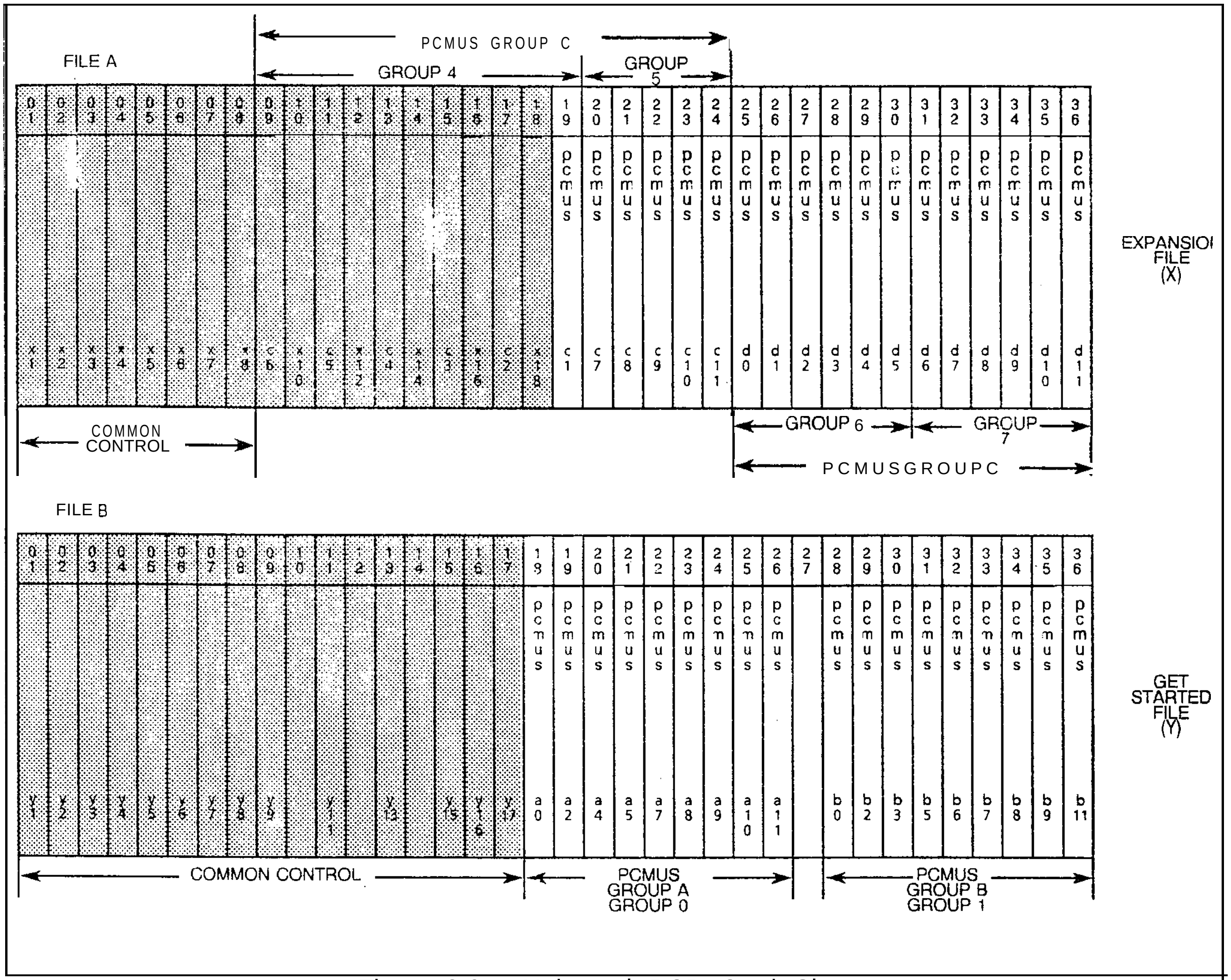


Figure 3.9 Universal PCB Card Slots

Table 3.3 Universal Printed Circuit Boards

FB#	MNE-MONIC	PCMUS CARD FUNCTION	CIRCUITS PER CARD
FB-17276-A	AIOD	Automatic Identification of Outward Dialing	(2 E&M, 2 simplex circuits)
FB-17208-A	ATT12	Attendant Interface	(2 circuits)
FB-17225-A	CIP	Control Interface Processor	(8 circuits)
FB-17236-A	DVCIP	Data Voice Control Interface Processor	(4 circuits; (4 Data/Voice)
FB-1721 O-A	PADIC	Public Address and Dictation	(2/1 circuits)
FB-51279-A	PCONF	PCM Dual Tone Multi-frequency Receiver	(1 circuits)
FB-17202-BOA	PCOT	Two-way Trunk to CO	(4 circuits)
FB-17203-A	PDTMF	PCM	(4 circuits)
FB-‘7201 -A	PEMT	Two-wire E&M Trunk	(4 circuits)
FB-51267-A	PFWTA	PCM Four-wire E&M Trunk	(4 circuits)
FB-51280-A	PILT	PCM Incoming Loop Trunk	(4 circuits)
FB-17254-A	PLCC	PCM Line Circuit Card	(8 circuits)
FB-17250-A	POPS	PCM Off-Premises Station Line	(8 circuits)
TR-100119-I	PMI	PMS Interface Card	
FB-17280-A	PPTR	PCM Progress Tone Recognizer	(4 circuits)
FB-17251 -A	PRLT	PCM Release Link Trunk	(4 circuits)
FB-17209-A	SIDML	SI Dual Modem and Current Loop	(2 circuits)
FB-17235-A	VCIP	Voice Control Interface Processor	(8 circuits)

Table 3.4 Universal Printed Circuit Boards Physical Mounting Requirements

PART NUMBER	CARD NAMES	CARD SLOTS REQUIRED	CARD WIDTH (INCHES)
FB-17201 -A	PEMT	2	1.2
FB-17202-A	PCOT	2	1.2
FB-17203-A	PDTMF	1	.6
FB-17208-A	ATTI2	2	1.2*
FB-17209-A	SIDML	2	1.6
FB-1721 O-A	PADIC	2	1.6
FB-17225-A	CIP	1	.6
FB-17235-A	VCIP	1	.6
FB-17236-A	DVCIP	1	.6
FB-17250-A	POPS	1	1.2
FB-17251 -A	PRLT	2	1.2
FB-17254-A	PLCC	1	.6
FB-17276-A	AIOD	3	1.8
FB-17280-A	TC/PPTR	1	.6
FB-51267-A	PFWTA	2	1.2
FB-51279-A	PCONF	1	.6
FB-51280-A	PILT	2	1.2

[**NOTE: *-** two slots required for heat output

AIOD
FB-17276

3.6.1 The Automatic identification of Outward Dialing card provides automatic number identification of the calling station on outgoing calls.

The system identifies the station number and sends the number over a dedicated channel to the central office AIOD equipment in the connecting central office. This card will supply four stations and four trunk numbers encoded in two of five formats (0, 1, 2, 4, 7) over a choice of four channels, two simplex loop channels and two E&M channels.

ATT12
FB - 17208

3.6.2 The Attendant Interface card is required for the system to interface with an Attendant Console or BLDU (Busy Lamp Display Unit). This card consists of two ports; each port supports either one Attendant Console or one to four BLDUs.

The three possible configurations of these ports are as follows:

- Two Attendant Consoles, or
- Four BLDUs
  - Each BLDU could have three more BLDUs "daisy chained" off it (the features of the daisy chained BLDUs would be the same as the main BLDU).
  - The system supports a maximum of 4 BLDUs, in a daisy-chained configuration, per card
- One Attendant Console and one BLDU (daisy chaining can be used here as well).

NOTES:

- In a two-console application, it is preferred to have one Attendant Console per card in case of card failure.
- A PLCC line interface card circuit is aiso required to provide an analog voice path to an Attendant Console. It is recommended to put the ATT12 and PLCC cards next to one another. The ATTI2 card connects the Attendant Console or BLDU to the digital data link that connects via the MPB85 card to the CPU. The transfer of data between this card and an Attendant Console or BLDU is performed in a full-duplex, 1,200 baud, seriai, current loop mode. The transfer of data between this card and the MPB85 card is in parallel by eight-bit format under control of the CPU.

CIP
FB-17225-A

3.6.3 The CIP (Control Interface Processor) card provides the system interface to an Analog Integrated Featurephone. One card supports up to eight Analog Integrated Featurephones.

Note: A line interface card circuit must also be used to provide an analog voice path for an Analog Integrated Featurephone.

CIP to Analog Integrated Featurephone communications is a digital data link: system to CIP communications is done by memory mapping, using the MPB85 card. The transfer of digital data between this card and an Analog Integrated Featurephone is performed in duplex, using MPRTs (Mini-Packet Receiver/Transmitters) (FGBS's proprietary data link) and the associated LBP (Link Bus Protocol). The transfer of data between this card and the MPB85 card is in parailel by eight-bit format under control of the CPU.

DVCIP
FB-17236-A

3.6.4 The Data Voice Control Interface Processor card is required for the system to interface with a data/voice DFP (Digital Featurephone). This card supports interfacing for up to four DFPs having both data and voice transmission capability. Each card has eight circuits. One circuit supplies the voice connection while another circuit supplies the data connection. Each DFP connects to the system via an individual twisted pair wire. DVCIP-to-DFP communications is via a digital data link; system to DVCIP communications is done by memory mapping, using the MPB85. The transfer of digital data between this card and a DFP is performed in duplex, using MPRTs (Mini-Packet Receiver/Transmitters).

The card separates the transparent voice and data mini-packets received by the phone, converts them back to PCM form, and presents them to the PCM bus on four even channels (0, 2, 4, and 6). For voice, the data is transmitted on circuits 1, 3, 5, and 7. The transfer of data between this card and the MPB85 card is in parallel by eight-bit format under control of the CPU.

NOTES:

- The DVCIP card is used to support circuit switched data which is discussed in section 5.0.
- PD-200 data cards are discussed in section 7.0 of this document.

PADIC
FB-17210-A

3.6.5 The Public Address and Dictation card is required for the system to interface with customer-provided dictation or paging equipment. This card supports two dictation trunks, one paging circuit with four outputs, and one input from a music source for the four paging outputs.

Each dictation access circuit allows station users with the appropriate class of service to access and control the specific dictation equipment with either dial pulse or tone signaling methods. Alternate dictation equipment, which is directly connected to the standard line circuit card, is also available for DTMF only signaling. The user uses the station instrument dial to enter control codes which start and stop the recording process, rewind the tape a short distance, and then play back a portion of the recorded message. This allows erasure and recording for corrections and rewind capabilities for review of the entire message.

The page access portion of the PADIC card has four individual audio circuits with low-impedance outputs and dry-circuit contacts for control of the associated external paging system equipments. Data base programming allows the user to select 11 different zone combinations, each of which can consist of one of the four circuits or any combination of the four circuits. A single music input is provided and four front panel switches allow selection of the circuits to which background music will be applied. The background music is automatically disconnected from the circuit accessed for paging.

With the page access circuit installed, the code-calling feature is available. Only data base programming is required to establish the desired code sequences, duration, etc.

This card contains a CODEC (Coder/Decoder) for analog/digital and digital/analog conversion. Turn-on and turn-off control signals are received from the system digital data link. Sampling control signaling is controlled by channel memory via the PCMFS card.

PCONF
FB-51279-A

3.6.6 The PCM Conference card is required if the system includes eight-party conference feature. This card contains a CODEC for analog-to-digital and digital-to-analog conversion. Sampling signals from the PCMFS card allow this card to process PCM inputs representing eight time slot related end-to-end connections. A summing circuit on the card combines and amplifies all connected stations in a manner that allows each party to hear all other parties in the conference except the subscriber. This card does not connect to any external peripheral equipment. This card is also used in support of the silent monitor and station monitor features. There are a maximun of eight cards allowed for both features.

NOTE: Slot A0 can only contain a PCONF or PDTMF voice card. For data cards allowed in this slot, see section 7.0.

PCOT
FB-17202-BCA

3.6.7 The Two-Way Trunk to CO card is required for the system to interface with loop dial trunks or release link trunks. This card supports four trunk circuits with trunk impedance levels that are nominally 600 ohms. Either the loop-start or ground-start mode of operation is selected by data base programming. Special strapping is not required for either operational mode. This card detects incoming trunk signaling and provides outgoing trunk signaling control.

For example, if the power failure transfer option is incorporated in the system, one PCOT card (four trunk circuits) can be wired for power failure transfer. During normal operation, the CO trunks dedicated to power failure transfer operate like all other CO trunks.

This card contains a CODEC for analog-to-digital and digital-to-analog conversion. Scan and seize control signals are received from, and request signals are transferred to, the PEC CPU via the digital data link. Sampling control signaling is controlled from channel memory via the PCMFS card.

PDTMF
FB-17203-A

3.6.8 The PCM Dual Tone Multi Frequency Receiver card is required for the system to process dialed DTMF digits from a DTMF telephone and the Attendant Consoles. This card receives and decodes the 2-out-of-7 DTMF tones from the station's line interface card via the PCM input bus and outputs the decoded digits in hexadecimal format to the PEC CPU via the digital data link. This card contains four DTMF receiver circuits.

NOTE: Slot A0 can only contain a PCONF or PDTMF voice card. For data cards allowed in this slot, see section 7.0.

PEMT
FB-17201 -A

3.6.9 The Two-Wire E&M Trunk card is required fcr the system to interface with two-wire E&M trunks. This card supports interfacing to four two-wire E&M trunk circuits. Connection to the trunk consists of one tip and ring analog path plus an E signaling and M signaling connection.

The Two-Wire E&M Trunk card is typically used for Tie trunks. The card provides a two-way operation between systems. These trunk circuits can also be used as one-way-only interfaces and operate in either a dial or ringdown mode of operation. They can be used as CO trunks if required. External signaling circuits are required at both ends.

Each circuit provides for tandem, trunk-to-trunk, and CCSA (Common Control Switching Arrangement) applications. One E&M trunk circuit on this card is also required for each recorder announcer used in the system.

This card contains a CODEC for A/D (Analog-to-Digital) and D/A (Digital-to-Analog) conversion. Scan and seize control signals are received from, and request signals are transferred to, the CPU via the digital data link. E&M signaling is also interfaced through the digital data link. Sampling control signaling is controlled from channel memory via the PCMFS card.

PLCC FB-17254-A

3.6.12 The PCM Line Circuit card, which has eight tip and ring circuits, is required for system to interface with any of the following:

- Attendant Console
- Music-on-Hold
- Agent Instruments
- Station Instrument
- Stand-alone Featurephone
- Integrated Featurephone (analog only)

The card contains a CODEC for A/D and D/A conversion. Scan control signals are received from, and request control signals are transferred to, the CPU via the digital data link. Sampling control signaling is controlled by channel memory via the PCMFS card.

NOTE: IFPs (Integrated Featurephones) must be on a separate PLCC and cannot be mixed With other types of users (attendant, station instrument, etc.). All other line types can be mixed on the same card (attendant, station instruments, etc.).

POPS FB-17250-A

3.6.13 The PCM Off-Premises Station Line Circuit card allows the system to interface with distant or off-premises station instruments operating with up to 1,300 ohm loops (this includes the typical 200-ohm instrument resistance). This card supports interfacing with eight station instruments. This card contains a CODEC for A/D and D/A conversion. Scan control signals are received from, and request control signals are transferred to, the PEC CPU via the digital data link. Sampling control signaling is controlled by channel memory via the PCMFS card.

This card meets the FCC Class C requirements for off-premises extensions to PABX systems, allowing the system planner to implement less costly distant or off-premises extensions by eliminating the requirements for midpoint signaling and voice repeaters for loops up to 1,300 ohms.

TR-100119-1
PMS interface Card
PMI

3.6.14 The Property Management System Interface card connects the system to the PMS. This card communicates with the PMS using its PMS port and with the OMNI PABX using its KEDU port on the SIDML card, FB-17209-A. The PMI card provides for an interface to the system hotel printer via its printer port and the SIDML card. The PMI card fits into any universal card slot and its external connections are made to the CDF via outrigger cable adapters.

PPTR
FB-17280-A

3.6.15 The PCM Progress Tone Recognizer card is optional for stations that are allowed to access the MERS (Most Economical Route Selection) SCC (Special Common Carrier) option. The PPTR card eliminates the requirement for a programmed pause when accessing the SCC. Although this card is optional, it is recommended for the SCC access through the MERS option to prevent long delays in accessing the SCC.

PRLT
FB-17251 -A

3.6.16 The PCM Release Link Trunk card is required in the branch system for CAS Main system applications. The card supports four (RLTs) Release Link Trunks. A PRLT is used as a momentary access between the CAS Branch system and the CAS Main system. After a CAS Main attendant extends a line or trunk, the PRLT is dropped, making the particular PRLT circuit on this card available for use again.

The RLT circuit basically makes a system trunk look like a CO loop trunk. It provides a - 48 VDC battery output for loop operation, and also provides signaling, dial tone, etc. RLTs also allow an operator at the CAS Main location to direct calls within a branch system that require operator assistance. Primarily these calls are incoming CO trunk calls but can also be timed recalls, transfers, dial 0, etc., all of which provide the appearance of a staffed branch system.

RLTs may also be implemented in the system serving as the CAS Main system if centralized service features are to be provided as well.

Using a standard two-wire loop facility, each RLT circuit from the branch is connected to a two-way loop trunk circuit at the CAS Main location where data base programming establishes the CAS control capabilities.

SIDML
FB-17209-A

3.6.17 The SI Dual Modem and Current Loop card has two circuits and is required for the system to interface with the following:

- Agent Instrument
- Printer
- KEDU (Key Entry Display Unit)

NOTE: A Line Interface card circuit is also required to interface with an Agent Instrument. One or two line interface circuits are used for each Agent Instrument, and two circuits are always required for the supervisor's Agent Instrument.

The SIDML card connects the Agent Instrument to the digital data link that connects through the MPB85 card to the PEC CPU. The transfer of digital data between this card and an Agent Instrument is performed in a full-duplex, 1,200-baud, serial, current loop mode. The transfer of data between this card and the MPB85 card is in parallel by eight-bit format under control of the CPU.

VCIP
FB-17235-A

3.6.18 The Voice Control Interface Processor card is required for the system to interface with a voice-only Digital Featurephone. This card supports interfacing to eight Digital Featurephones via individual twisted pair wiring. VCIP to Digital Featurephone communications is via a digital data link; system to VCIP communications is done by memory mapping, using the MPB85 card. The transfer of digital data between this card and a Digital Featurephone is performed in duplex, using MPRTs (Mini-Packet Receiver/Transmitters). The transfer of data between this card and the MPB85 card is in parallel by eight-bit format under control of the CPU.

The card contains a 6502 microprocessor with 40K bytes of RAM and 15K bytes of ROM.

Power File Printed Circuit Boards

3.7 This paragraph contains descriptive information about PCBs installed within the Power File (Figure 3.10). Table 3.5 lists the power cards and the dedicated card slots.

Table 3.5 Power Cards

Card Number Mnemonic and Name	Card Slot
FB-17197-A, PSUPY, Power Supervisory Card	C/01
FB-51051 -A, PTF, Power Fail Transfer	C/04
FB-17204-A, BC5R, Battery Charger 5-Volt Regulator	C/06
FB-20996-A, RABR, Recorder Announcer Buildout Resistor (Rear Mount)	C/08

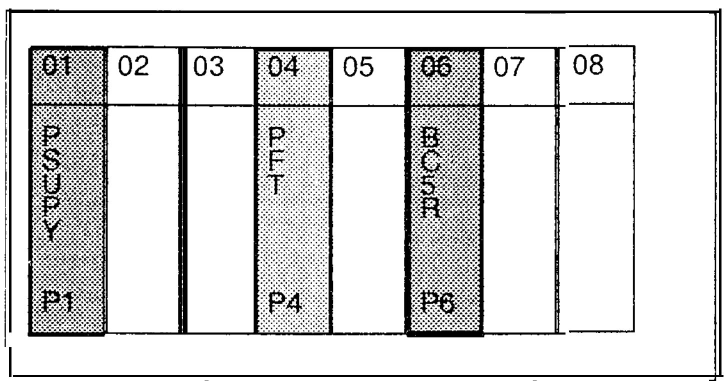


Figure 3.10 PSUPY Card

PSUPY FB-17197-A

3.7.1 The Power Supervisory card monitors the power supply outputs from both the Get Started and the Expansion Files. The card provides the following:

- LEDs (light emitting diodes) that indicate
  - fuse alarms
  - ringing voltage alarms
  - initialization status
  - status of + 5 VDC, + /-12 VDC, and -48 VDC potentials for both files
- Housing for
  - CPU85E RESET button for memory loading routine preparation
  - Type 200 Test Set enable/disable TOGGLE switch

PFT
FB-51051 -A

3.7.2 The Power Fail Transfer card is required when provisions for power failure trunks are required. This card connects up to seven dedicated line/trunk circuits directly to the DDD network on power outages.

During normal operation, the switch on the front panel of the PFT card should be in the upward or horizontal position. Setting the switch in the downward position results in a forced transfer of the seven trunks on the corresponding relays on the PFT card. This causes LED-l, located on the card, to light. This forced transfer is only used for testing purposes.

BC5R
FB-17204-A

3.7.3 The Battery Charger 5-Volt Regulator card provides + 5 volt DC to the Ml MB card. Under normal operating conditions (when AC power is present), it charges the 5 volt back-up batteries. When power faiis, this card will provide + 5 volt DC from the back-up battery pack to the MPG16, Ml MB. and the CP85E. This action maintains the system memory and the CP85E wake-up clock.

RABR
FB-20996-A

3.7.4 The Recorder Announcer Buildout Resistor is required for a music-on-hold or recorder announcer connection. Located on the rear of the power file backplane in slot 8, it provides tip and ring resistors to the lines connected to the recorder announcer. This prevents system users who are attached to the announcer from talking to each other.

The card also contains resistors and capacitor circuitry for connecting music-on-hold.

SYSTEM HARDWARE 4.0 This section describes the OMNI SI system hardware.

Common Control Hardware **4.1** The system uses an INTEL 8085 microprocessor as the master system control device. The INTEL 8085 and other associated circuitry are mounted on a PCB called the CPU (Central Processor Unit) FR 17288-A, Figure 4.1.

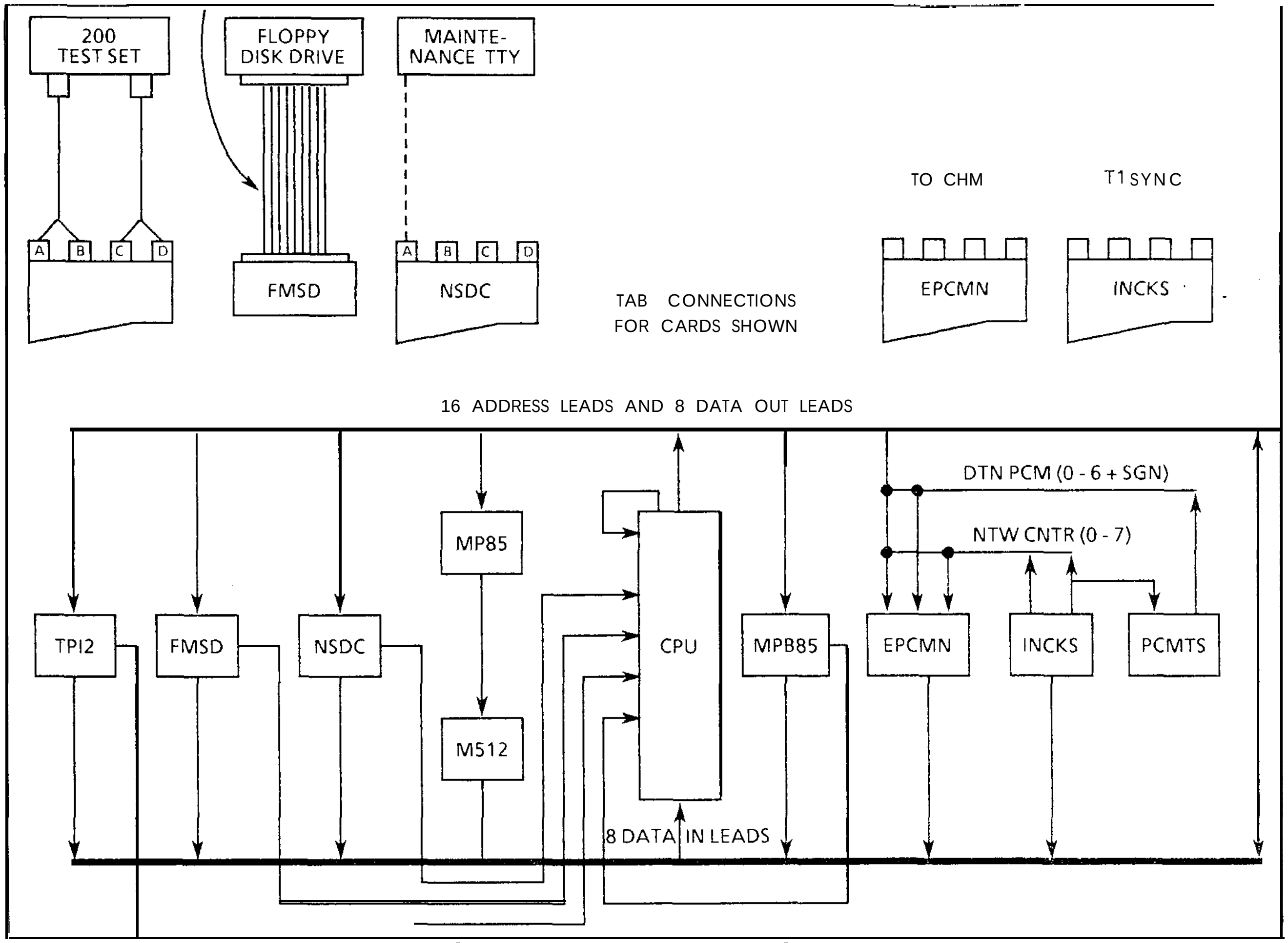


Figure 4.1 CPU Connections

The CPU has 16 parallel leads known collectively as the ADDRESS bus, which are used for addressing memory locations and PCB (Input/Output) locations. Since the Intel 8085 is an eight-bit microprocessor, the CPU DATA IN and DATA OUT buses that communicate with the memory and I/O consist of eight parallel leads each. The CPU, via the ADDRESS bus, can identify any memory storage location (eight-bit word) and instruct the memory to load the contents of that location into the CPU DATA IN bus so the CPU can read the data and respond to it accordingly. The CPU also generates new information that must be stored and uses the ADDRESS bus to instruct the memory where to store the information being received from the CPU DATA OUT bus.

In addition, there are also status and control signal leads, and priority interrupt leads complementing the CPU.

The system is equipped with an Ml MB memory card. The 1 MEG memory is divided into Instruction Pages and Data Pages O-7. The MPG16 paging card, under software control, determines which page should be active and sets it accordingly.

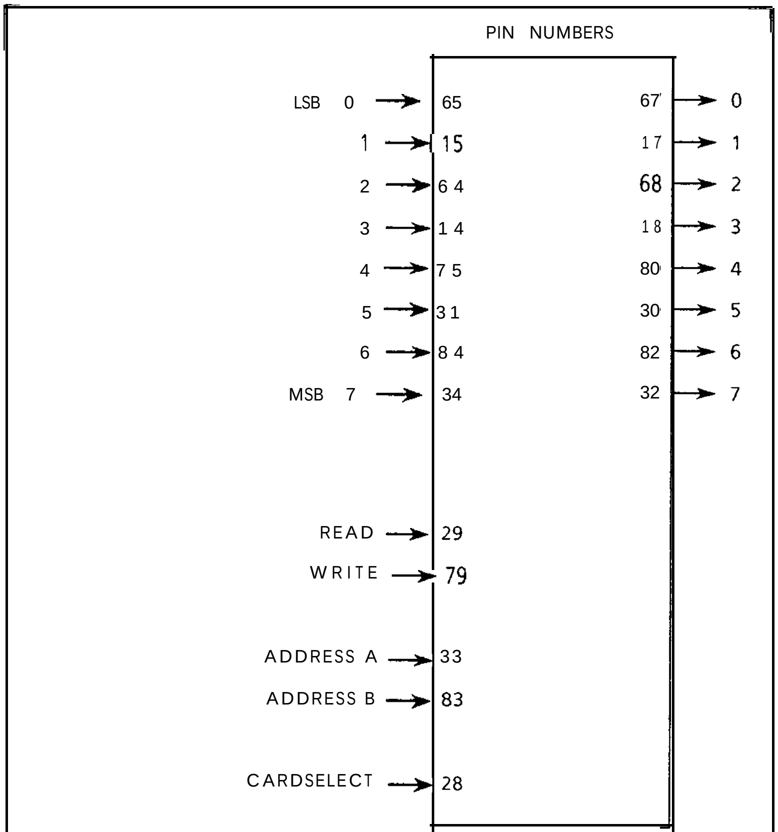
The CPU also uses the adddress and data buses to communicate with other PCBs. The CPU connects to the MPB85 card to perform common control functions (such as PCM Universal Slot (PCMUS) circuit sense and control functions), and it connects to the CHM85 card to control the time-switch network. The CPU connects to all of the common control cards except the Pulse Code Modulation Tone Source (PCMTS) card and INCKS card associated with the time-switch network.

The CPU has priority interrupt leads, which are used by some cards to gain access to the CPU when required. For instance, the Narrow Serial Device Controller (NSDC) card is more efficiently serviced on a priority interrupt basis. The priorities are arranged so that the CPU attends to the most urgent requests first.

Peripheral Equipment Interface

4.2 The OMNI SI system has 40 PCMUSs (PCM Universal Slots) which provide connectorized cable access to the outside world. There is an extra PCMUS (AO) available for circuits not requiring external connections (PDTMF or PCONF). One MPB85 card controls the Get Started File, while a second MPB85 card controls the Expansion File. Both MPB85s are directly connected to the CPU address, data out, and data in buses. Only one MPB85 will be considered in this discussion.

Common buses connect the MPB85 card to all of the PCMUSs in each group, and to the T1 card slots (in the Expansion File only). These are the read or sense bus (8 leads), the write or control bus (8 leads), the circuit address bus (2 leads), and a read lead and a write lead. The MPB85 card can control individual PCMUS positions because of a separate select lead connection to each PCMUS in the file (Figures 4.2 and 4.3). The MPB85 card can select a circuit address (0-3), enable the read lead and then use one PCMUS select lead to initiate a read command to the PCB installed in that PCMUS. Any data appearing on the read/sense bus leads will be from that PCMUS circuit card. The two address leads allow four different circuits to be mixed on each PCMUS circuit card.



'igure 4.2 Control and Sensing Connections of PCN Universal Slot

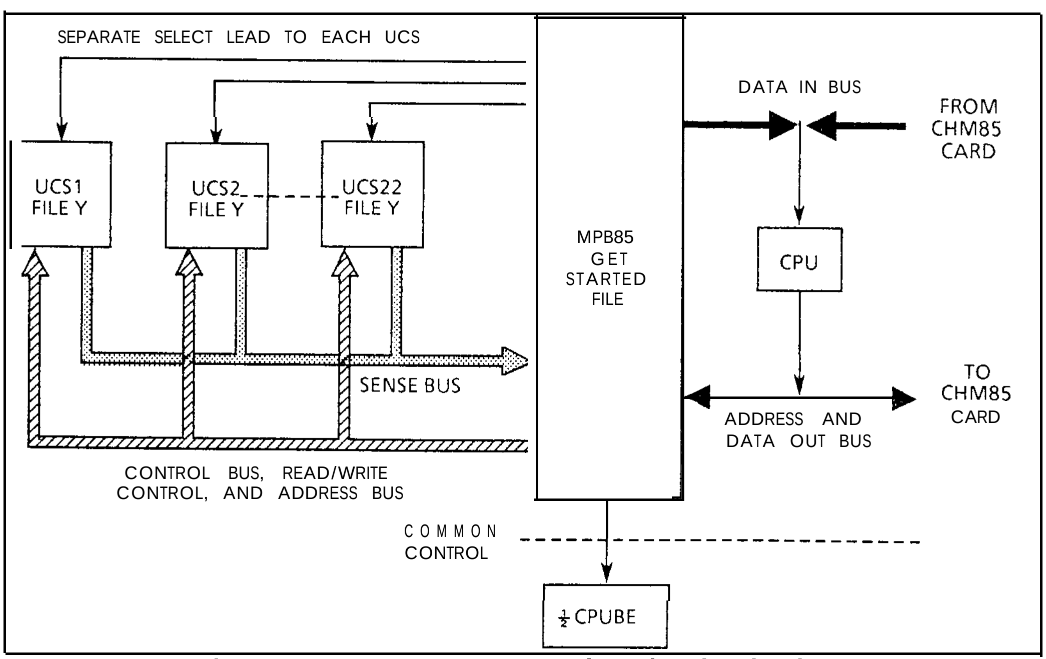


Figure 4.3 Common Control Related Circuits

For example, a loop trunk circuit card has four identical circuits. The MPB85 card addresses each individually. When one of the trunk circuits detects an incoming seizure (ringing), it changes the status of a gate (lead 6) on its sense bus. When the MPB85 card requests a read of that circuit, the sense/read bus (lead 6) reflects this change of status. The MPB85 card accepts this information as a change in status of sense bit 6 of an 8-bit digital word, passes this information on to the CPU, which compares it to the previous sense word received from that trunk circuit. When the CPU receives a second sense word confirming the incoming seizure from this trunk circuit, it sends the request-for-service event message which alerts the time-switch network to prepare for another conversation (assign a time slot).

When a PCMUS contains a line circuit card, the address select function is not required. Since only on-hook/off-hook sense information is required, one sense/read word representes all eight circuits, .

Time -Switch Network

4.3 The time-switch network of the system establishes the connection between stations and passes data between them. The time-switch network uses five memories and a one-to-one correspondence between memory locations (called a time slot) Figure 4.4.

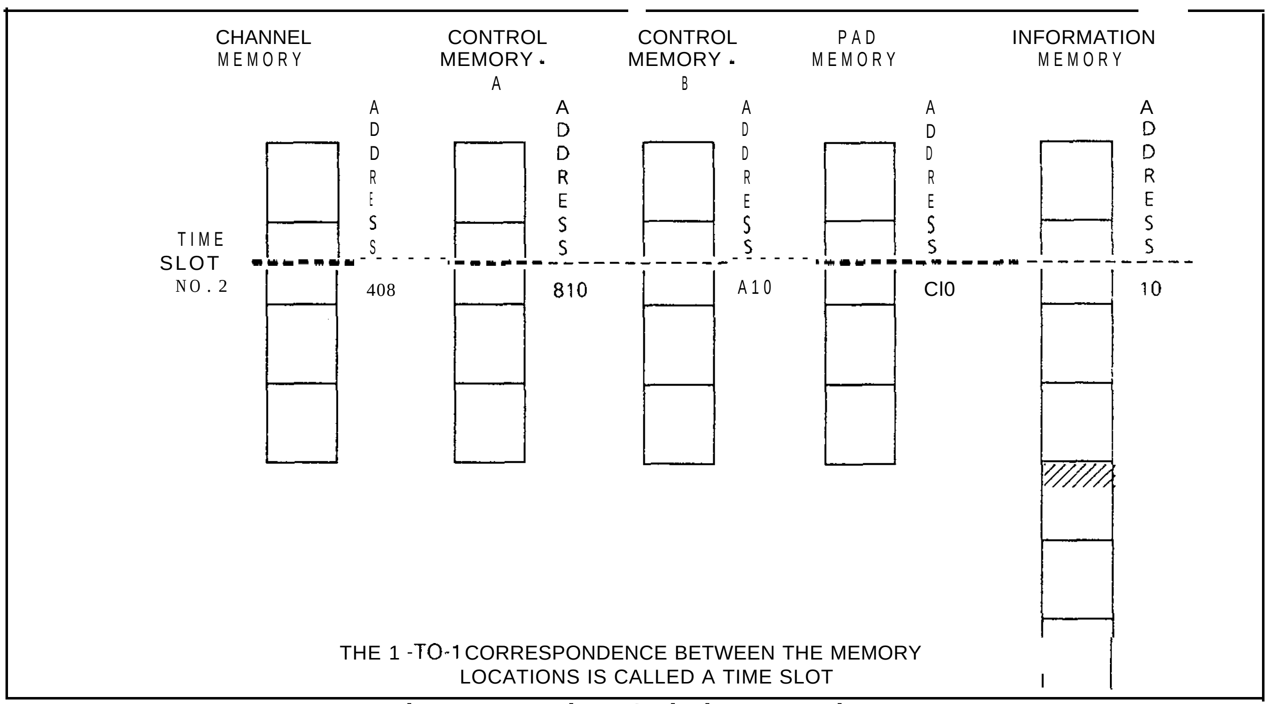


Figure 4.4 Time-Switch Network

Call Processing

4.3.1 A basic call, which uses information memory, channel memory, and control memory (A), is processed as follows:

1. The system scans every line for an off-hook condition, and finding one (assume phone number 83), reports it to the CPU.

2. The system writes the location number of that phone into an unused location in the channel memory (time slot); assume time slot number 3 (Figure 4.5). At the same time, the system also reserves time slot number 3 in control memory.

3. The system collects the digits dialed and determines who is being called (assume phone number 55)

   a. Time slot number 1 is assigned to phone number 55, and

   b. Time slot number 1 is reserved in control memory.

   c. Connection is completed by writing the time slot number of the other phone into the two reserved locations in the control memory (Figure 4.5).

   NOTE: The memories involved are channel memory, control memories A and B, and pad memory.

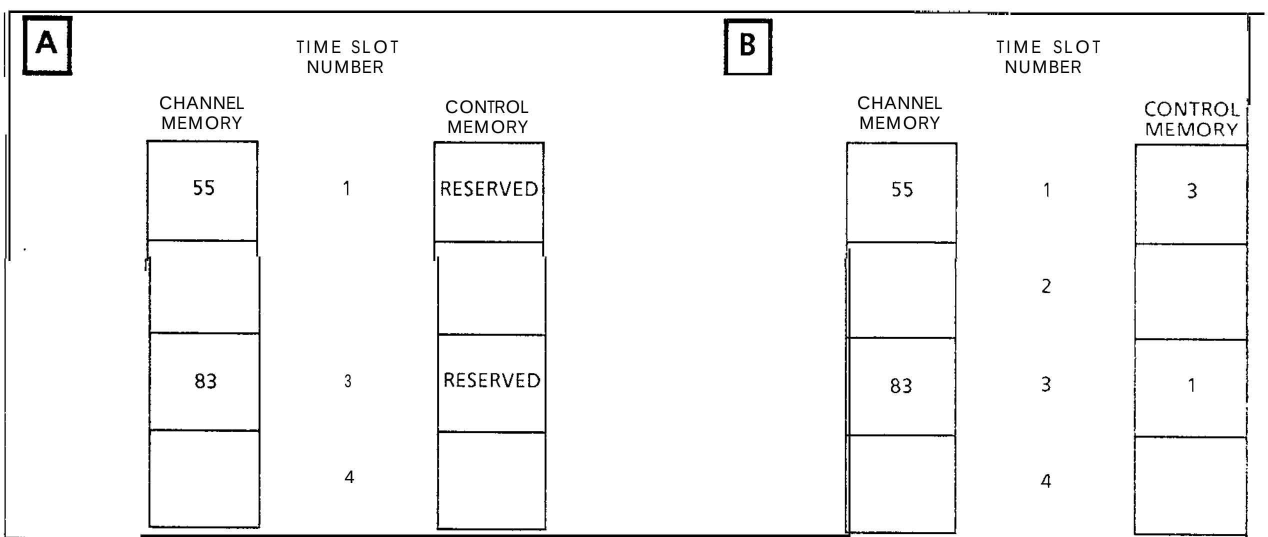


Figure 4.5 Time-Switch Memory Setup

NOTE: For further definition of time slots, see section 10.0 of this document.

Call Processing Data Transfer

4.3.2 Data is passed between the connected users (stations 83 and 55) in the following manner:

1. The network scans through each time slot in the information memory.

2. For every time slot that has a written phone location, the network goes to that location, gets a data sample, and writes the sample into the same time slot in the information memory

3. The network, after doing the data collection for phone 83, looks in the same time slot in the control memory where the time slot number is for the other phone that phone 83 is talking to.

4. The network goes to that time slot in the information memory to get the data previously taken from the other phone (55), which is then sent to phone 83.

To summarize the data transfer: phone 83 is in time slot 3, and time slot 3 in control memory contains a 1. Therefore, a data sample should be taken from time slot 1 information memory and sent to phone 83. Similarly, phone 55 is in time slot 1, and time slot 1 in control memory contains a 3; therefore, a data sample should be taken from time slot 3 information memory and sent to phone 55.

NOTE: The previous discussion used a “black box” approach in which memory was accessed by time slot numbers; however, the actual system accesses memory by memory address. The memory addresses contain the information that directs the connection established by the time-switch network. All of these memories are interrelated with a time slot/channel number. A complete memory cross-reference is shown in Tables 4.1 and **4.2.**

Table 4.1 Memory Cross-Reference Get Started File
Channel Memory to EPCMN Network Memory Cross-Reference

	Group 0 Slots AO, A2, A4, A5, A7, A9, A10, All				Group 1 Slots BO, B2, B3, B5, B6, B7 B8, B9, B11			
	Channel Memory	EPCMN Network Memory			Channel Memory	EPCMN Network Memory		
	04. . .	CMA 08. . .	CMB OA. . .	PAD oc. . .	04. . .	CMA 08. . .	CMB OA. . .	PAD oc. . .
CH 00	. .oo		. . .00		. . .01		. . .02	
CH 01	. . .04		. . .08		. .05		. .OA	
CH 02	. . .08		. . .10		. .09		. . .12	
CH 03	. . .OC		. . .18		. . .0D		. . .1A	
CH 04	. . .10		. . .20		. . .11		. . .22	
CH05	. . .14		. . .28		. .15		. .2A	
CH06	. . .18		. . .30		. .19		. . .32	
CH 07	. . .1C		. . .38		. .1D		. . .3A	
CH 08	. . .20		. . .40		. . .21		. .42	
CH 09	. . .24		. . .48		. .25		. . .4A	
CH 10	. ..28		. . .50		. 29		. . .52	
CH 11	. . .2C		. . .58		. 2D		. .5A	
CH12	. ..30		. . .60		. . .31		. . .62	
CH13	. ..34		. . .68		. . .35		. .6A	
CH 14	. . .38		. . .70		. . .39		. . .72	
CH 15	. . .3C		. . .78		. .3D		. . .7A	
CH16	. ..40		. . .80		. .41		. . .82	
CH17	. ..44		. . .88		. .45		. .8A	
CH18	. ..48		. . .90		.. 49		. .92	
CH 19	. . .4C		. . .98		. . .4D		. . .9A	
CH 20	. . .50		. . .A0		. . .51		. . .A2	
CH 21	. .54		. .A8		. . .55		. AA	
CH 22	. . .58		. . .BO		. . .59		. .B2	
CH 23	. .5C		. . .B8		. .5D		. .BA	

Table 4.2 Memory Cross-Reference Expansion File
Channel Memory to EPCMN Network Memory Cross-Reference

	Group 4 Slots Cl - C6				Group 5 Slots C7 - Cl 1			
	Channel Memory	EPCMN	Network	Memory	Channel Memory	EPCMN	Network	Memory
	02.. .	CMA 08. . .	CMB OA. . .	PAD OC. . .	. 02. . .	CMA 08. . .	CMB OA. . .	PAD oc. . .
CH 00	. .OO		. . .01		. . .01		. . .03	
CH 01	. . .04		. . .09		. . .05		. . .0B	
CH 02	. .08		. . .11	. .	. 0 9		. . .13	
CH 03	. . .0C		. . .19		. . 0D		. . .1B	
CH 04	. . .10		. . .21		. . .11		. . .23	
CH 05	. . .14		. . .29		. . .15		. . .2B	
CH 06	. .18		. . .31		. . .19		. . .33	
CH07	. . .1C		. . .39		. . .1D		. . .3B	
CH 08	. .20		. . .41		. . .21		. . .43	
CH 09	. . .24		. . .49		. . .25		. . .4B	
CH 10 .	..28		. . .51		. . .29		. . .53	
CH 11 .	. .2C		. . .59		. . .2D		. . .5B	
CH12 .	..30		. . .61		. . .31		. . .63	
CH13 .	..34		. . .69		. . .35		. . .6B	
CH 14	. . .38		. . .71		. . .39		. . .73	
CH15 .	. .3C		. . .79		. . .3D		. . .7B	
CH16 .	..40		. . .81		. . .41		. . .83	
CH17 .	..44		. . .89		. . .45		. . .8B	
CH 18 .	..48		. . .91		. . .49		. . .93	
CH 19 .	. .4C		. . .99		. . .4D		. . .9B	
CH 20	. . .50		. . .AI		. . .51		. . .A3	
CH 21	. . .54		. .A9		. .55		. . .AB	
CH 22	. .58		. . .B1		. . .59		. . .B3	
CH 23	. . .5C		. . .B9		. . .5D		. . .BB	

Table 4.2 Memory Cross-Reference Expansion File
Channel Memory to EPCMN Network Memory Cross-Reference (Continued)

	Group 6 Slots DO - D5				Group 7 Slots D6 - D11			
	Channel Memory	EPCMN Network Memory			Channel Memory	EPCMN Network Memory		
	02. . .	CMA 08. . .	CMB OA. . .	PAD oc. . .	02. . .	CMA 08. .	CMB OA. .	PAD oc. . .
CH 00	. .02		. . .05		. . .03		. 07	
CH 01	. .06		. . .0D		. . .07		. .0F	
CH 02	. . .OA		. . .15		. . .0B		. .17	
CH 03	. . .OE		. . .1D		. . .0F		. . .1F	
CH04	. ..12		. . .25		. . .13		. .27	
CH 05	. . .16		. . .2D		. . .17		. . .2F	
CH 06	. .1A		. .35		. . .1B		. .37	
CH 07	. . .1E		. . .3D		. . .1F		. .3F	
CH 08	. . .22		. . .45		. . .23		. .47	
CH 09	. . .26		. . .4D		. . .27		. 4F	
CH 10	. . .2A		. . .55		. . .2B		. 57	
CH 11	. . .2E		. . .5D		. .2F		. .5F	
CH12	. ..32		. .65		. . .33		. .67	
CH13	. ..36		. .6D		. . .37		. .6F	
CH 14	. . .3A		. . .75		. .3B		. . .77	
CH 15	. . .3E		. . .7D		. . .3F		. 7F	
CH 16	. . .42		. . .85		. . .43		. 87	
CH17	. ..46		. . .8D		. . 47		. 8F	
CH18	. ..4A		. . .95		. . .4B		. . .97	
CH 19	. . .4E		. .9D		. .4F		. .9F	
CH 20	. . .52		. . A5		. 53		. A7	
CH 21	. . .56		. .AD		. . .57		. .AF	
CH 22	. . .5A		. .B5		. . .5B		. . .B7	
CH 23	. . .5E		. . .BD		. . .5F		. .BF	

Each time slot also includes other memories not yet discussed: control memory B, pad memory and some extra memory locations allocated to information memory. The purpose of these memories follows:

Control Memory B

4.3.3 Control memory B accommodates three-way conference calls. This is done as follows:

1. The third party (phone off-hook) has been assigned a different time slot; and, in this arrangement, each time slot contains the information memory address of the other two time slots (one address in control memory A, and the other in control memory B).

2. A threshold detector, based on voice volume, determines which information memory data will be selected for a data sample.

   a. If the data sample in the information memory address associated with control memory B represents a louder volume level than the data sample associated with control memory A, control memory B will be used,

   b. If the volume is lower, then control memory A is normally used.

In summary, the detector decides which control memory (A or B) is used, and the contents of control memory determine which time slot is used. During a simple two-way conversation call, control memories A and B both contain exactly the same information.

Pad Memory

4.3.4 Pad memory contains data bits which are used to select the amount of attenuation (pad) to be applied to the data being transferred through the connection. It reduces the volume of nearby (loud) sources, thereby providing a uniform volume level between different types of connections.

Information Memory

4.3.5 Information memory has extra memory locations (Figure 4.4) which contain data bits that represent various tones (dial tone, ringback tone, busy tone, quiet tone, DTMF tones, etc.) used by the system during call processing. Tones are issued by the system information as follows:

1. When a station goes off-hook, the system connects a dial tone to that subscriber as a signal to begin dialing.

2. Connection is made by writing the address of the location in information memory (time slot number), where the dial tone data is stored, into control memory.

3. When the subscriber dials the first digit, the instrument is connected to quiet termination (another location in information memory).

4. The process continues until the subscriber connects to the information memory address that is associated with the time slot assigned to the party being called.

Line Card Interface

4.4 Transmission of voice/tone signals through the time-switch network is performed using time-division multiplex techniques and by collecting samples of the input signal

1. The incoming signal passes through to the CODEC converter circuits on the line card.

2. A digital sample is taken and put onto the PCM IN bus (an 8-bit serial-line bus).

3. The PCM IN bus passes the data into the PCMI card which converts serial PCM to 8-bit parallel PCM.

4. The data passes to the information memory at the address for the time slot that has been assigned to the hardware (station) generating the input signal.

5. A sample is collected and the data in that information memory address is updated 8,000 times per second.

Digital Line Card Interface

4.4.1 The digital voice/tone signal is retrieved from the above information memory address, using a different method and route.

1. The control memory of the time slot that has been assigned to the station being called (destination) contains the information memory address of the data which is to be retrieved.

2. This data is inserted onto the PCM bus and delivered to the appropriate PCMI card and, eventually, to the appropriate line card.

3. Like the analog process, this occurs 8,000 times each second, which enables the smoothing samples by the filtering circuit to produce an analog signal.

4. This restored analog signal is output from the filter circuit and then passed on through the SLIC hybrid.

Time Slot Group Division

4.4.2 In the system, each group has 24 time slots or channels (Figure 4.6). The channel memory connects to the EPMN (Expanded Pulse Code Modulation Network) card by front tab plug-in cables and uses one-half of the EPCMN card containing information memory. Therefore, 24 time slots per group times 6 groups per system equals 144 time slots. See Section 10.0 for more information on time slots.

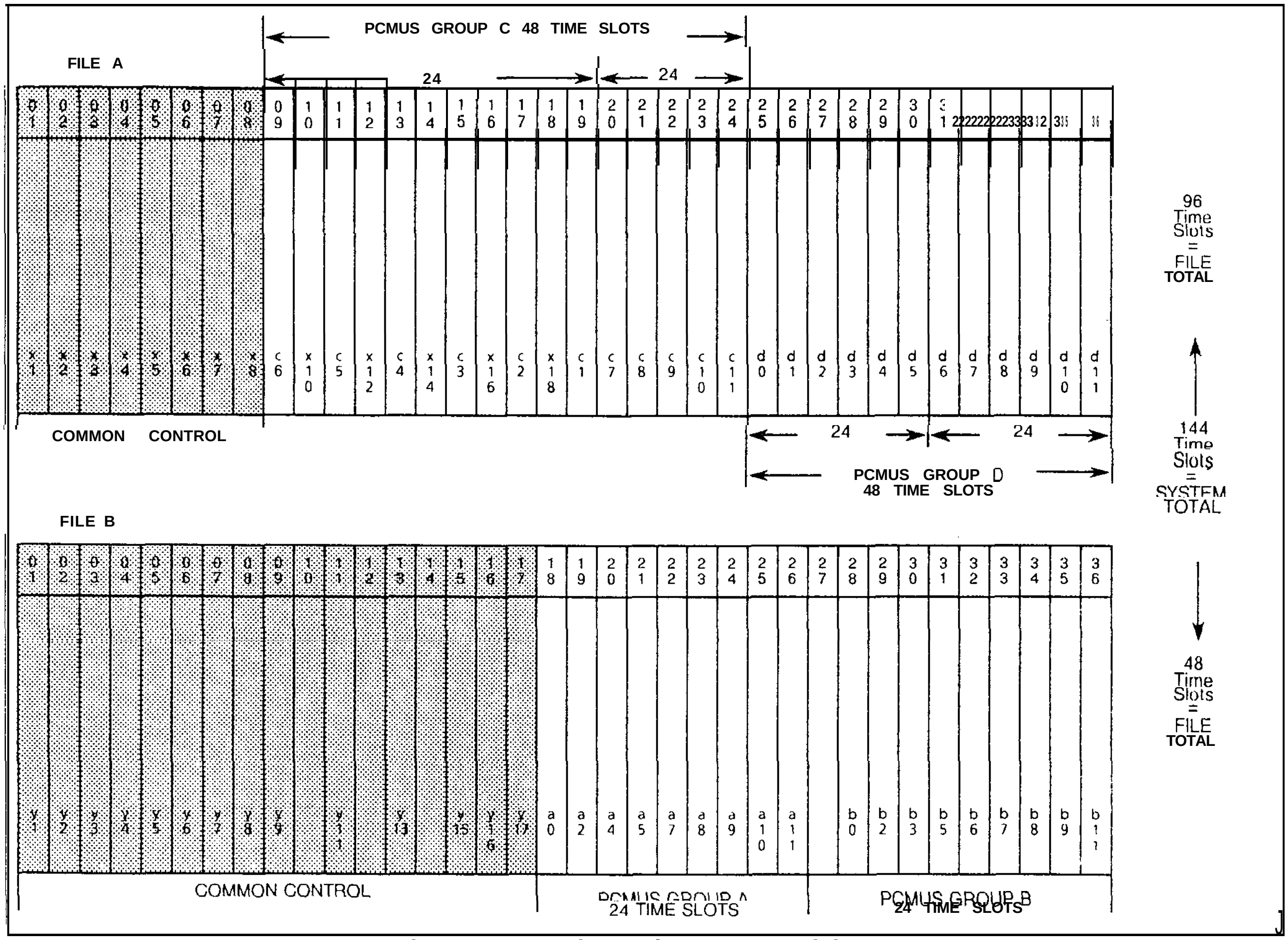


Figure 4.6 Time Slot Group Divison

Network Hardware Operation

4.5 The time-switch network is a self-sustaining operation. It continuously performs memory reading functions and acts on the data obtained without intervention from the system microprocessors. Once the system microprocessor has written instructions into the time-switch network memories for a given phase of any call progression, no further intervention is required. The INCKS (Synchronizable Intermediate Network Clock) card continuously sends out addresses from a time-slot counter to control/synchronize this function. The system samples each active circuit 8,000 times per second which is equivalent to sampling every 125 microseconds. Figure 4.7 shows an example of an analog signal being sampled. The resulting output is shown in Figure 4.8. Figure 4.9 showes the file bus with samples present.

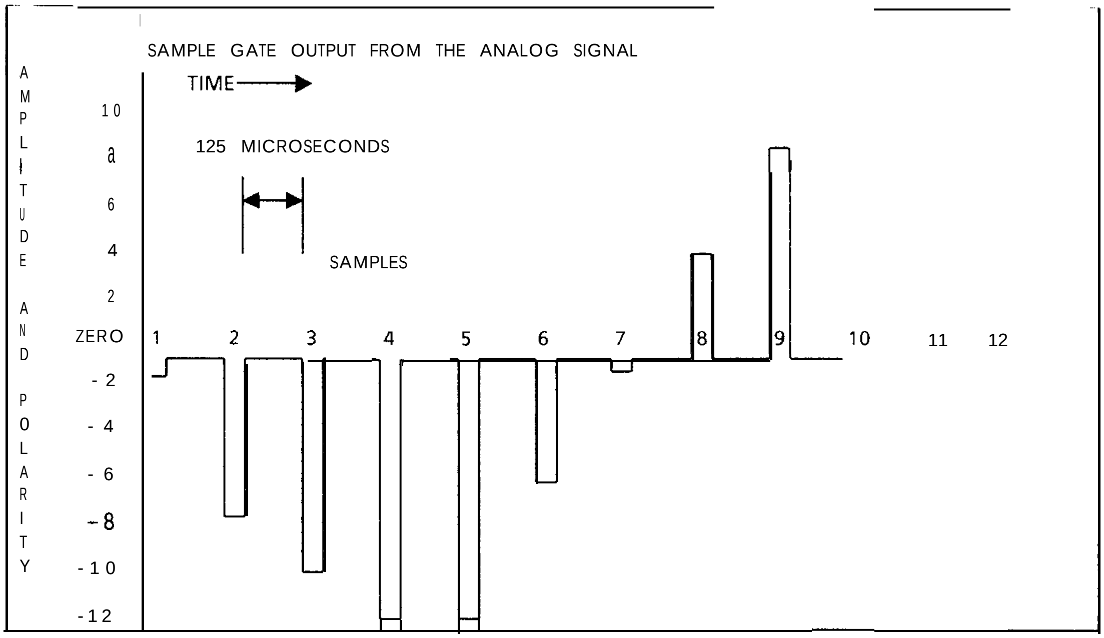


Figure 4.7 Sampling an Analog Signal

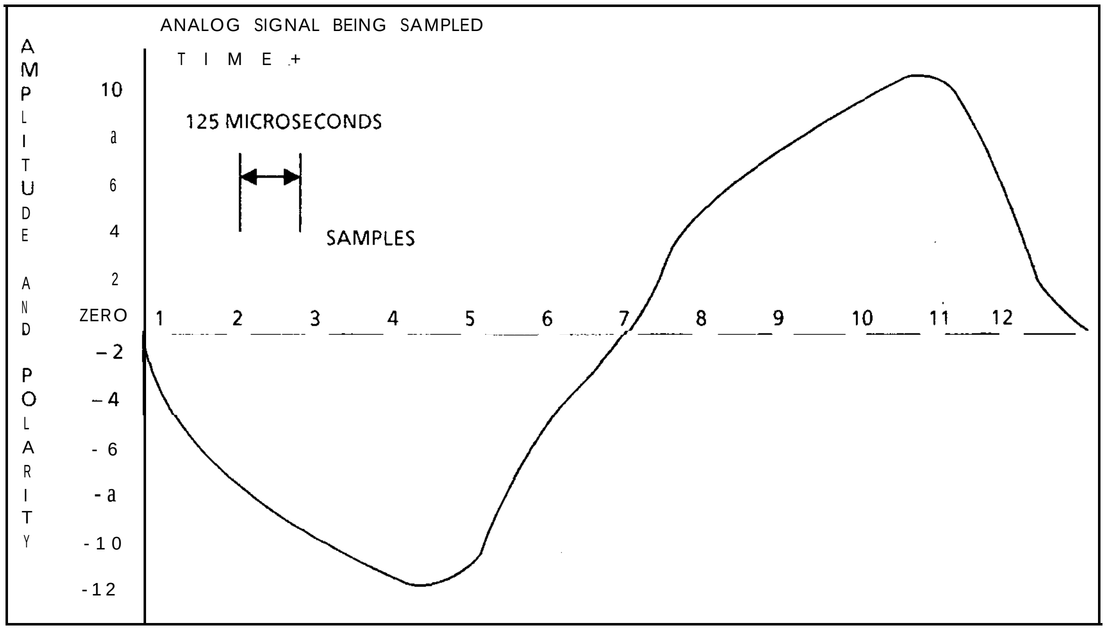


Figure 4.8 Sampling an Analog Signal

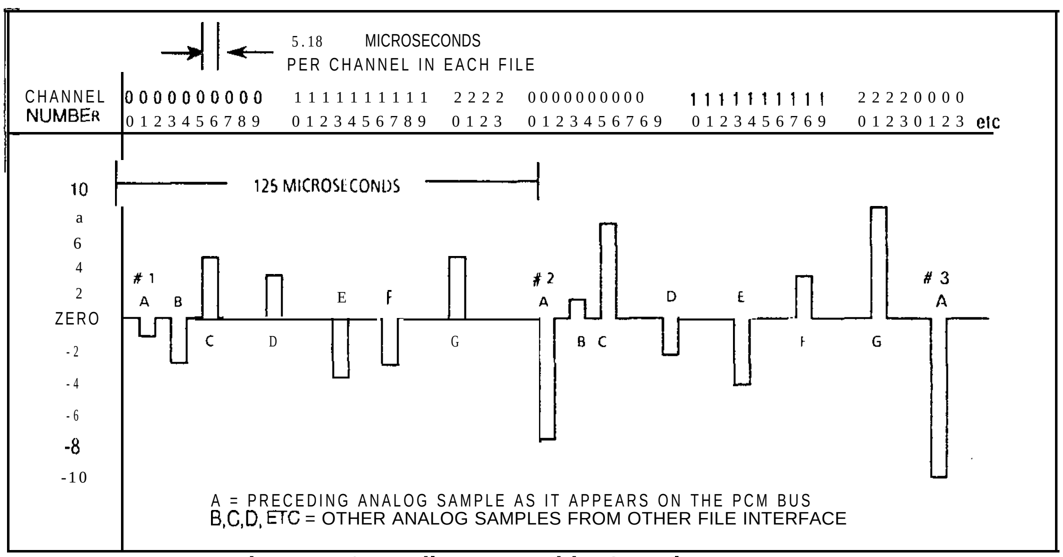


Figure 4.9 File Bus with Samples Present

All time-switch network functions must be performed for each active circuit in only 0.000000648 second. Conversely, the network performs 1,544,000 time slot processes every second. Each time slot process is performed in four different time intervals or periods (called cycles) within this 648 nanosecond period, which means a 162 nanosecond cycle period or a 6.17 MHz rate. Figure 4.10 shows the time-switch network and PCMUS group interleaving.

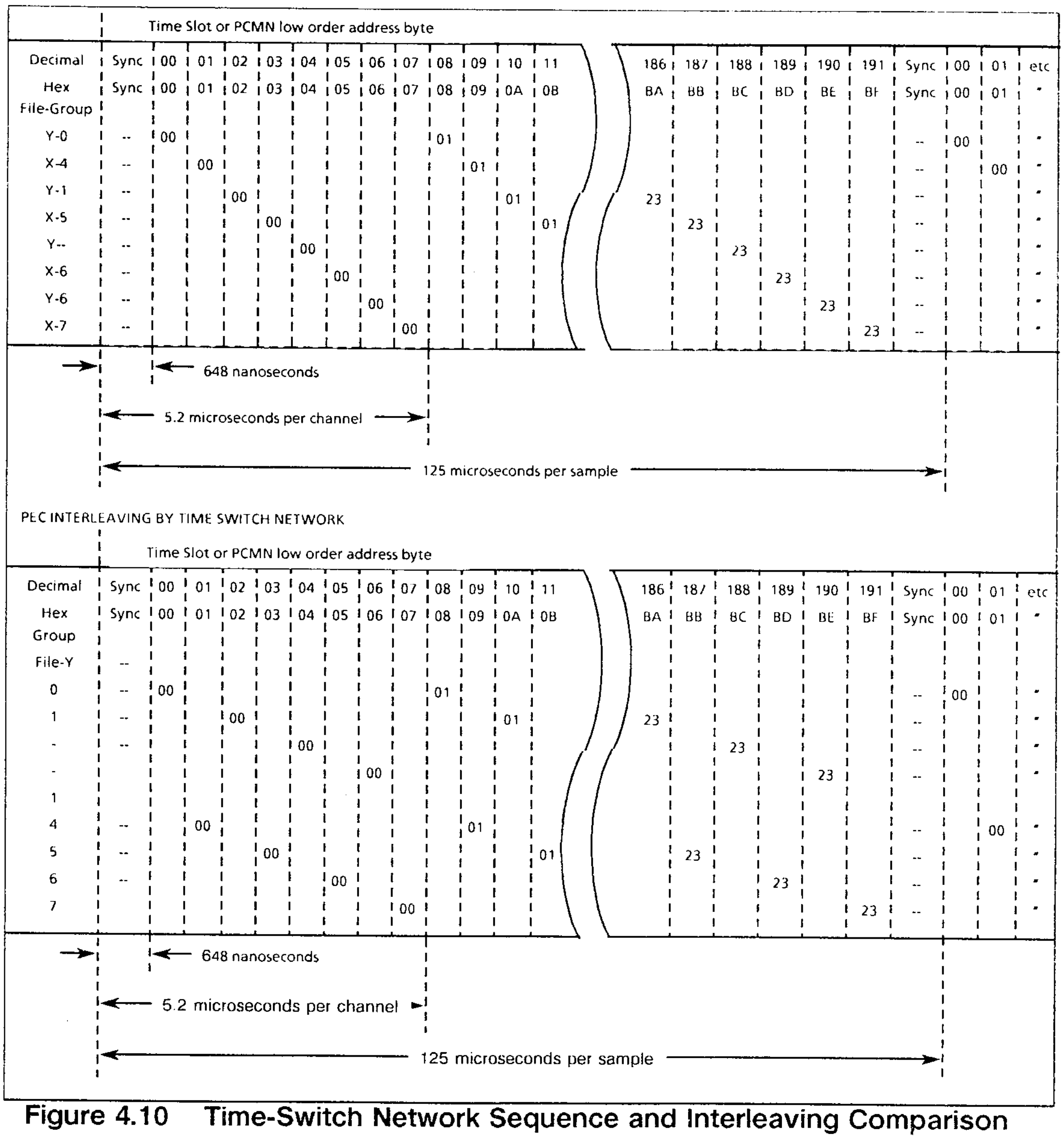


Figure 4.10 Time-Switch Network Sequence and Interleaving Comparison

The INCKS card provides 8-phase sequences or synchronizing intervals for each cycle so that its base rate is 49.408 MHz. The PCMTS (Pulse Code Modulation Tone Source) card and the EPCMN card are time-synchronized by the INCKS card. The instantaneous digital equivalent of the tone samples are loaded into information memory addresses 192 through 255 (hexadecimal CO through FF) during every pass of time slots 64 though 128. Both EPCMNs repeatedly sequence time slots 1 through 192. Thus, during each time slot, new PCM data is written into the information memory (hexadecimal addresses 00 through BF) of the EPCMN card (network). Consequently, the EPCMN stores, accesses, and controls information from 144 differenttime slots and 64 different tones. The EPCMN card also passes on timing signals from the INCK card to each CHM85 card to complete the time synchronization sequence.

Once the CHM85 has accepted the PCM data from the network, the system can complete the remainder of the connection. The CHM85 places this PCM data on the PCM bus and strobes the data into the proper PCMI card for parallel-to-serial conversion. The CHM85 card also directs the PCMFS card to access the correct PCMUS and circuit, and then enables the PCM gate in synchronism with the PCM signal. The circuit, associated with this active time slot then begins the filtering process on the amplitude signal just received. It will be 125 microseconds before the next sample is received by this circuit. During this time, the network will perform the above described actions for the other 191 time slots and returns to perform this time slot's time-switching actions..

Network Clock Card

4.6 To have a slaved-clock, a INCKS (Synchronzable Intermediate Network Clock) card (FB-20922-A) (Figure 4.11) must be placed in the system. The clock synchronization hardware on the INCKS card provides for monitoring four external 1.544-MHz clock signal inputs (SINXO to SINX3) on the card handle. One of the input clock signals is selected by the card and used for frequency synchronization. The clock card will remain locked on an input until it determines that the clock signal has missed a pulse or has been disabled completely. When the SINX input that the clock card is locked onto fails, the card begins scanning the four inputs for a valid clock input. When it finds a good SINX input, the card will frequency-synchronize onto that SINX input.

When the INCKS card is not locked onto an external T1 clock signal, it assumes master configuration and reverts to the system clock. When a valid input is detected, the card locks onto the external clock, cancels its free-running mode, and resynchronizes to the new input. When the INCKS card is in its free-running mode, the system is synchronized to the free-running clock frequency. When the INCKS card locks onto a SINX T1 input, the system is synchronized to the T1 input and functions in the slave mode.

The input to the INCKS card is frequency-synchronized and can be selected manually by the Advance Sync button on the FB-20922-A card (Figure 4.11). Pushing the button selects the next highest SINX input from the one in which the clock card is currently locked. One of the four SINX channels can also be selected by writing onto the data base. A bad input cannot be selected by programming. The card will always automatically release its frequency synchronization mechanism from a faulty SINX input and lock onto the next available good SINX output.

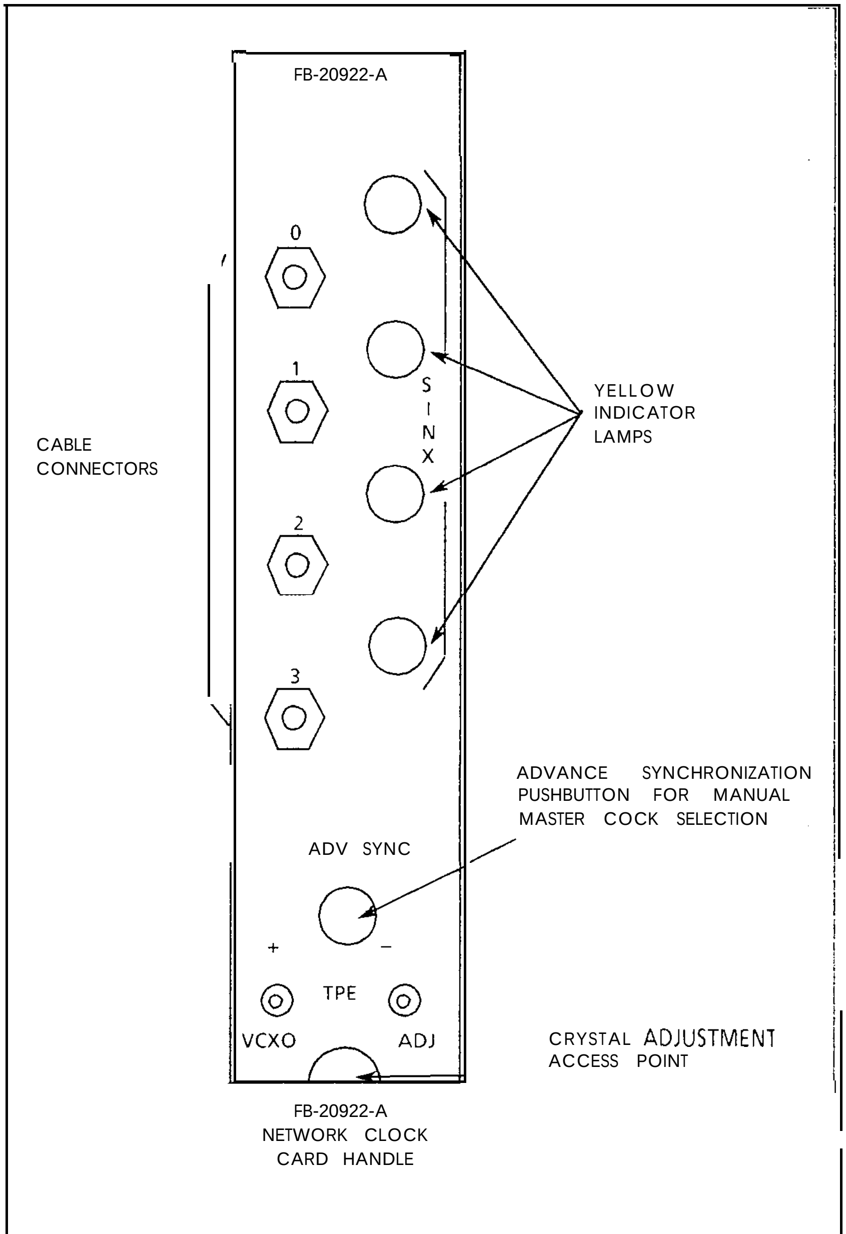


Figure 4.11 **INCKS** Card Handle View

Tones and Tone Generation

4.7 Table 4.3 defines the digitally generated tones supplied by the OMNI SI. These tones are used in the normal operation of various features.

Memory Page Card MPG16

4.8 The selector switch (Figure 4.12) on the Memory Page card (FB-17213-BOA) is for manual selection of the memory data page.

Table 4.3 Tones and Tone Generation

ADDRESS	TONE TYPE	FREQUENCY (Hz)	LEVEL*	DURATION
C0	Dial Tone	350 + 440	-13 dBm	Uninterrupted
C1	DTMF Tone 1	697 + 1,209	-6 dBm	-
C2	MF KP Tone	100 + 1,700	-6 dBm	-
C3	CAS Tone	440	-14 dBm	-
C4	Feature Confirmation Tone	350 + 440	-16 dBm	0.1 sec. on/ 0.1 sec. off
C5	DTMF Tone 2	697 + 1,336	-6 dBm	-
C6	MF Tone 1	700 + 900	-6 dBm	-
C7	CAS Tone	480	-14 dBm	-
C8	Conference Break-in Tone	350 + 440	-13 dBm	1 sec. burst
C9	DTMF Tone 3	697 + 1,477	-6 dBm	-
CA	MF Tone 2	700 + 1,100	-6 dBm	-
CB	CAS Confirmation Tone	440	-14 dBm	-
CC	Interrupted Dial tone	350 + 440	-16 dBm	0.1 sec .on/ 0.1 sec. off -3 times
CD	DTMF Tone 4	770 + 1,209	-6 dBm	-
CE	MF Tone 3	900 + 1,100	-6 dBm	-
CF	CAS Tone	620	-14 dBm	-
D0	Distinctive Dial Tone	440 + 480	-16 dBm	Uninterrupted
D1	DTMF Tone 5	770 + 1,336	-6 dBm	-
D2	MF Tone 4	700 + 1,300	-6 dBm	-
D3	Dial Tone	350 + 440	-19 dBm	Uninterrupted
D5	DTMF Tone 6	770 + 1,477	-6 dBm	-
D6	MF Tone 5	900 + 1,300	-6 dBm	-
D8	Quiet Termination	- -	-	-
D9	DTMF Tone 7	852 + 1,209	-6 dBm	-
DA	MF Tone 6	1,100 + 1,300	-6 dBm	-
DD	DTMF Tone 8	852 + 1,336	-6 dBm	-
DE	MF Tone 7	700 + 1,500	-6 dBm	-
E0	Busy Tone (Feature Confirmation Tone)	480 + 620	-24 dBm	60 IPM
E1	DTMF Tone 9	852 + 1,477	-6 dBm	-
E2	MF Tone 8	900 + 1,500	-6 dBm	-
E5	DTMF Tone 0	941 + 1,336	-6 dBm	-

Table 4.3 Tones and Tone Generation (Continued)

ADDRESS	TONE TYPE	FREQUENCY (Hz)	LEVEL*	DURATION
E6	MF Tone 9	1,100 + 1,500	-6 dBm	-
E8	Reorder Tone, Camp-on & Call-Waiting Tone	480 + 620	-24 dBm	120 IPM 1 sec. burst
E9	DTMF Tone*	941 + 1,209	-6 dBm	-
EA	MF Tone 0	1,300 + 1,500	-6 dBm	-
ED	DTMF Tone	941 + 1,477	-6 dBm	-
EE	MF ST Tone	1,500 + 1,700	-6 dBm	-
F0	Tick Tone	480 + 620	-24 dBm	64 ms on/ 2 sec. off
F1	Milliwatt	1,004	0 dBm	-
F2	MF STP Tone	900 + 1,700	-6 dBm	-
F6	MF ST2P Tone	1,300 + 1,700	-6 dBm	-
F8	Ringback Tone	440 + 480	-19 dBm	1 sec. on/ 3 sec. off
FA	MF ST3P Tone	700 + 1,700	-6 dBm	-
FE	Bad Parity Tone	-	-6.dBm	-

*Level of tones at the point of departure from the system

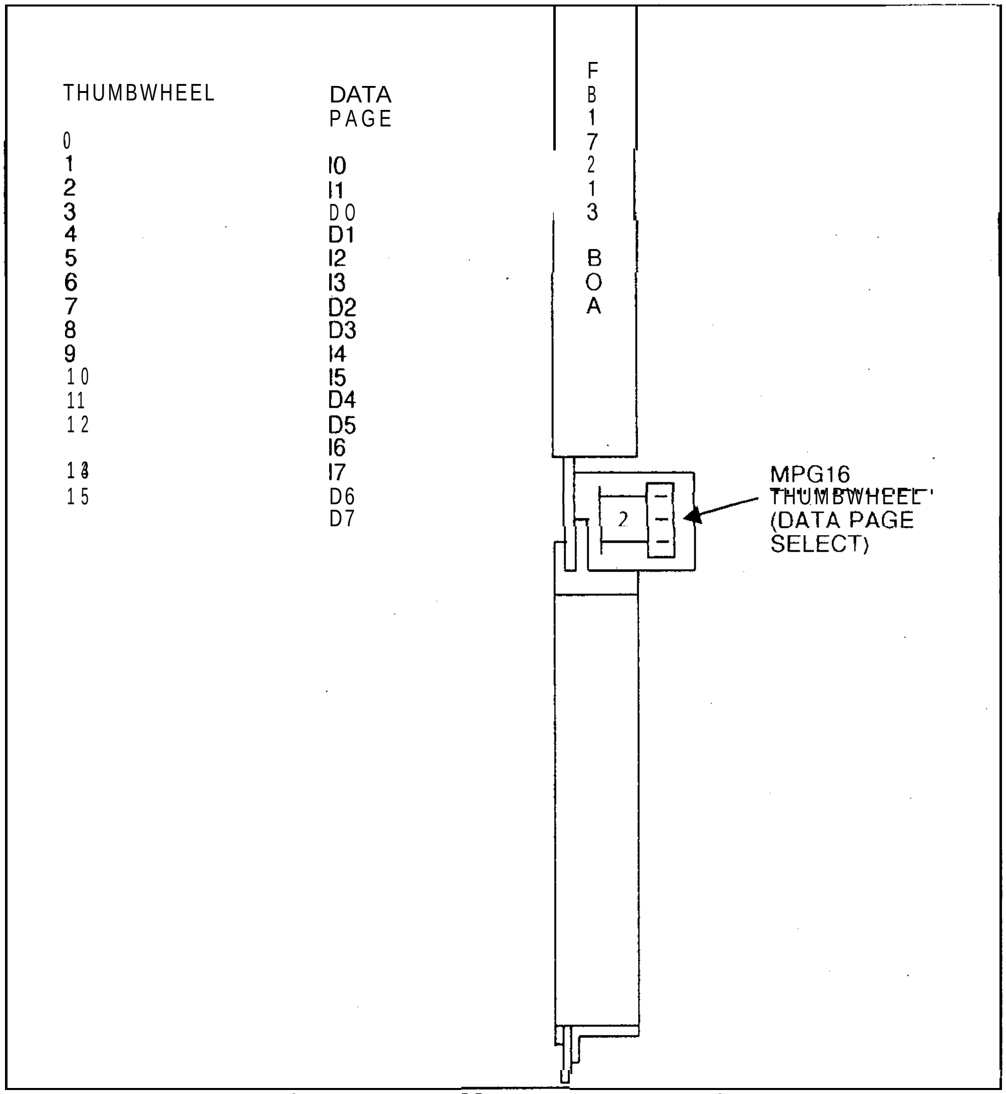


Figure 4.12 **Memory** Page Card

> CAUTION
>
> For normal service operations, thumbwheel must be set at 2.

Mass Storage

4.9 This paragraph describes the mass storage for the system's software.

Disk Subsystem

4.9.1 The OMNI SI uses the disk subsystem for system loading, data base back-up, and loading of recent change overlay programs. The hard disk (Figure 4.13) stores program and data images associated with the voice system processor and program and data base images for the data system.

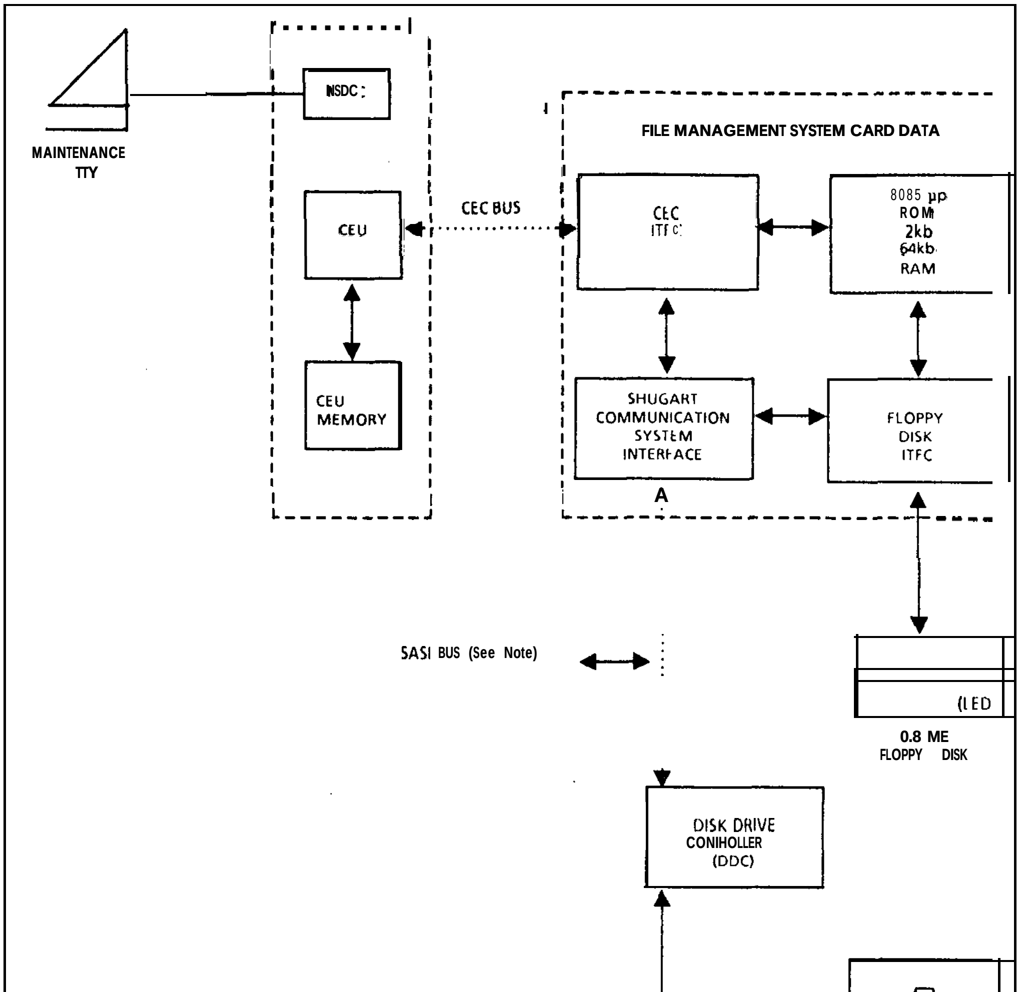
NOTE: See section 6.0 for information on the disk subsystem for a system containing the PD-200 data option. In addition, the hard disk contains the Recent-Change and off-line maintenance programs which are paged into memory for execution.

The system uses a 1 O-megabyte, Winchester-type 5-l /4 inch hard disk drive with or without the data option. The floppy disk (Figure 4.13) is a 5-1/4 inch industry standard. The floppy disk has a capacity of one million bytes of unformatted information and will store over 650,000 bytes of formatted information. The floppy disk provides distribution and back-up media for the system, and loads (initialize) the system during startup.

The FMSD (File Management System Data) card provides the interface between the voice systems microprocesser system and the DDC (Disk Drive Controller) (Figure 4.13).

The FMSD card, located in the Get Started File, slot 7, controls the mass storage device of the system. The FMSD card interfaces the voice switch via the CEC (Common Equipment Complex) processor bus.

NOTE: See paragraph 8.2 for description of the PD-200 Data System access to mass storage.



NOTE: PD-200 interface is shown in Figure 8.1.

Figure 4.13 Simplified Disk System Interface

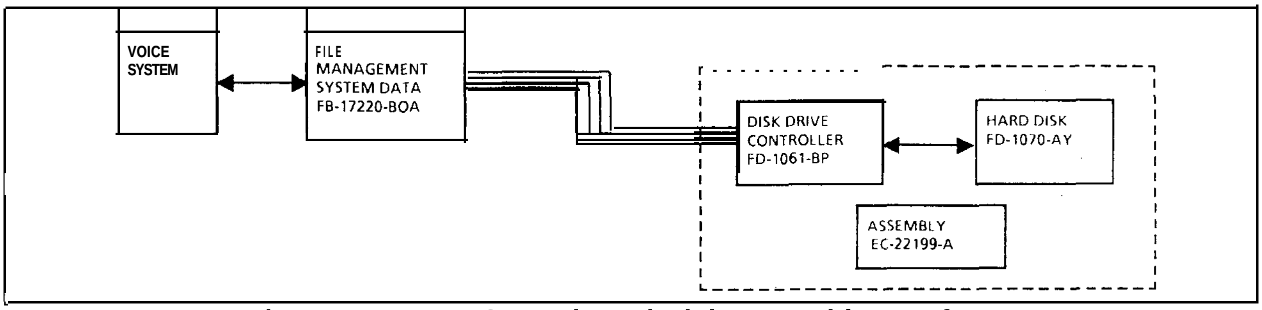


Figure 4.14 FMSD and Hard Disk Assembly Interface

File Management System

4.9.2 Random access to files can be supported with disk storage. The FMS (File Management System) software manages all data storage and transfer on the system disks. Simultaneous access to a single file, along with multiple open files, is possible. Opening a file causes a FID (File Identity) to be associated with that file. FMS calls use the FID to identify the file being accessed.

A high-level command set accesses the mass storage disk system. The command set is configured as an FCB (File Control Block). In executing an FMS command, the system processor constructs an FCB in its memory and informs FMS of the location of the FCB.

FMS commands are divided into the following categories:

- FILE ACCESS
  - Create
  - Open (supports 20 files simultaneously)
  - Close
  - Seize
  - Release
  - Boot
- DATA TRANSFER
  - Read
  - Write
  - Copy
  - Seek
- ADMINISTRATION AND MAINTENANCE
  - Format
  - Diagnostic
  - Configuration
  - Time
  - Mount
  - Dismount

. DIRECTORY MAINTENANCE

- Delete
- Rename
- Security (supports two levels - read and write)

File Access Command Descriptions

This paragraph provides a description of the various system commands.

- Create Command. The Create command can add a new file to the disk with the attributes and name given in the command. With completion of the command, the file opens for the access method specified in the FCB.

- Open Command. The Open command can open a disk file for read, write, or update access. Once opened, the user can access records in the file. The file remains accessible until the Close command is used.

- Close Command. The Close command can terminate access to a disk file. Optionally, the FID associated with the file can be freed for use by other requests. If the user wants guaranteed access to another file after closing this file, the FID should be kept and used in the next open file. If released, the FID is free for another task .

- Seize Command. The Seize command can reserve a specified number of FIDs required for a task. This command helps avoid deadlock by allowing a task to allocate all the FIDs required to complete a task.

- Release Command. The Release command can free up FIDs which were previously seized. This command makes the FIDs available for use in other tasks.

- Boot Command. The Boot command can open the file "FMS$BOOTR" for reading. This command, used by the system ROM program, loads the Smart Loader Program. The Smart Loader Program then loads the system generic program.

Data Transfer Command Descriptions

This paragraph provides a description of the various system data transfer commands.

- Read Command. The Read command requests a specified number of records to be transferred from an open file to the user's buffer area.

- Write Command. The Write command requests transfer of a specified number of data records from the user's buffer area into a file starting at the current record.
- Copy Command. The Copy command allows record transfer between two open files. The records are moved only within FMSD.

- Seek Command. The Seek command allows the CRP (Current Record Pointer) of an open file to be repositioned at the requested record.

Directory Maintenance Command Descriptions

This paragraph provides a description of the various system maintenance commands.

- Delete Command. The Delete command deletes a file from the disk by removing its entry in the Disk File Directory and deallocating the storage used by the file.

- Rename Command. The Rename command can change the name of a disk file.

- Security Command. The Security command can change the read/write security levels of a file.

NOTE: FMS commands that concern the PD-200 Data System are described in paragraph 8.2.3.

Disk Files

4.9.3 Files make up the contents of the disk. Disk files are subdivided into records. Sectors are used to hold the file's data and are called file data blocks. Depending upon the record size, many records may be stored in one sector, or one record may use two sectors.

Sectors on the disk are identified by a DBA (Disk Block Address). DBAs range from 0 to N, where N is the number of sectors on the disk drive-l. The DBA can be calculated by using a formula dependent on the physical characteristics of the drive. Usually, FMSD will only deal with the logical DBA and leave the composition of the DBA to the DDC (Disk Drive Controller). (Refer to Figure 4.15 to calculate the DBA.) The disk file structure shown in Figure 4.16 is typical. The maximum amount of files is 320. (A summary of the disk file organization operating under FMS is shown in Figure 4.17).

DBA = (cyl-addr head-per-cyl head-addr)

* set-per-trk + set-addr

DBA mnemonics =

cyl-addr = cylinder address
head-per-cyl = heads per cylinder
head-addr = head address
set-per-trk = sectors per track
set-addr = sector address

Figure 4.15 DBA Calculation

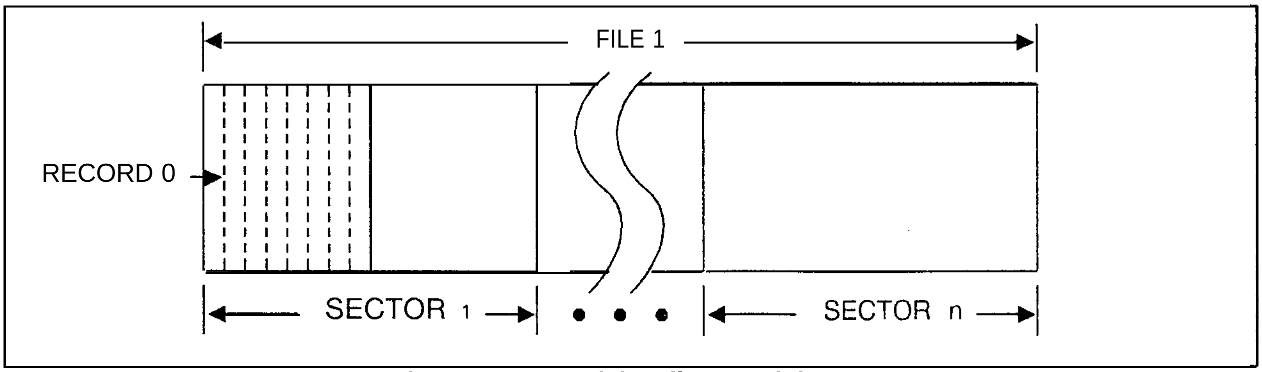


Figure 4.16 Disk File Breakdown

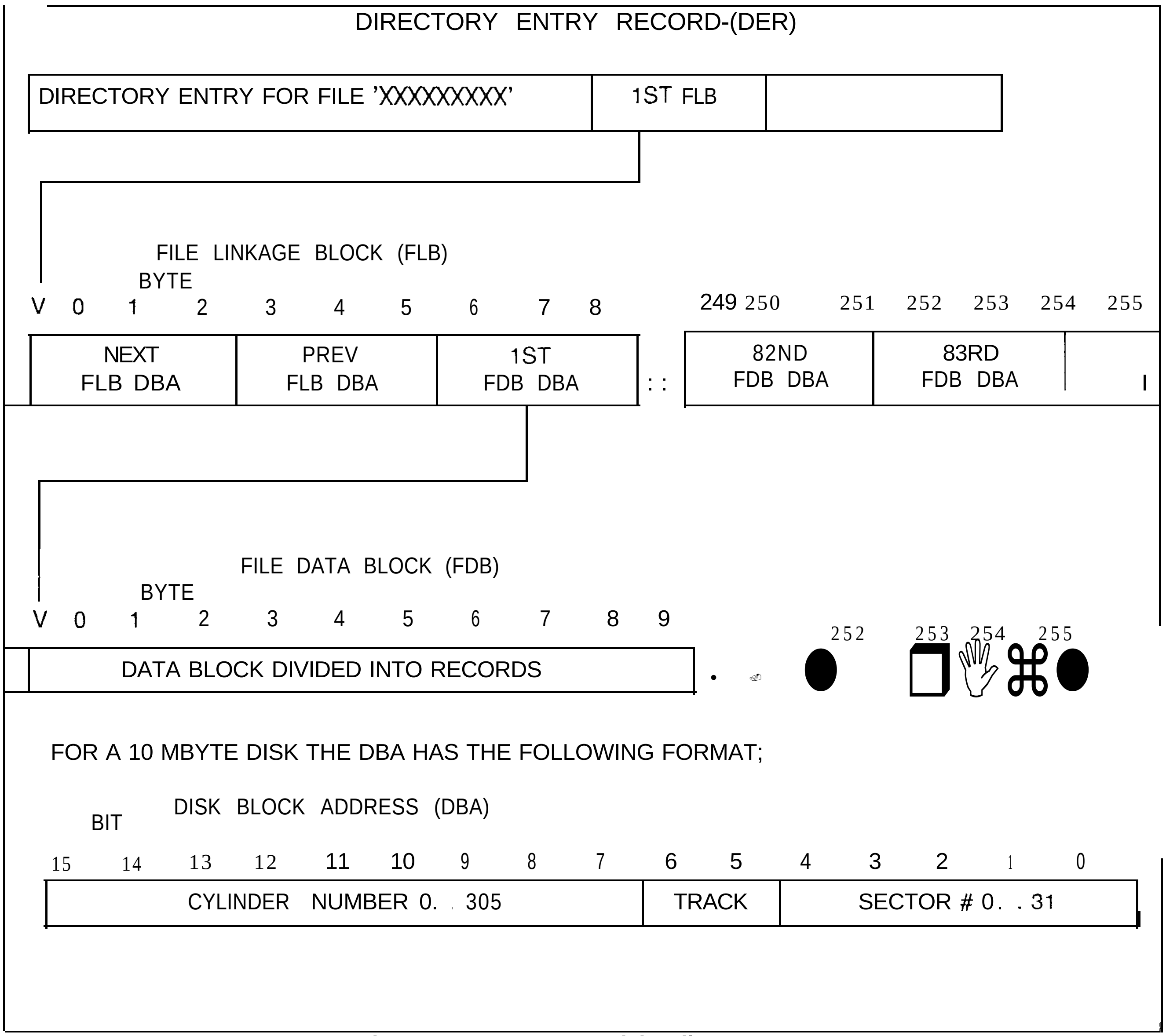


Figure 4.17 FMSD Disk File Layout

Disk File Directory

4.9.4 The Disk File Directory contains the names and attributes of all the files on the disk. The directory is organized as a file of 320 records (floppy disk has only 64 entries) consisting of 32 bytes each. Each record contains one directory entry. The directory is write-protected to prevent accidental overwriting or deletion. The FMS directly accesses the directory which resides on cylinder 0.

Disk File Directory record format (Figure 4.18) and restrictive considerations are defined as follows:

- IU: Directory record is IU (In-Use)
- NU: record has never been used
- SY: FMSD system file
- FC: file was forced closed

The NU (Never Used) bit in byte 0 of a directory record shortens search times for file names. When the first NU bit is found during the search for a file name, the rest of the directory is known to be empty. When you delete a file, the IU (In-Use) bit in byte 0 of the directory record is cleared. The directory record is then free to be used when another file is created. Setting the FC (Forced Closed) bit in byte 0 of the directory record is performed whenever a file open with an access of Write or Update is closed due to a Reset command or FMS restart.

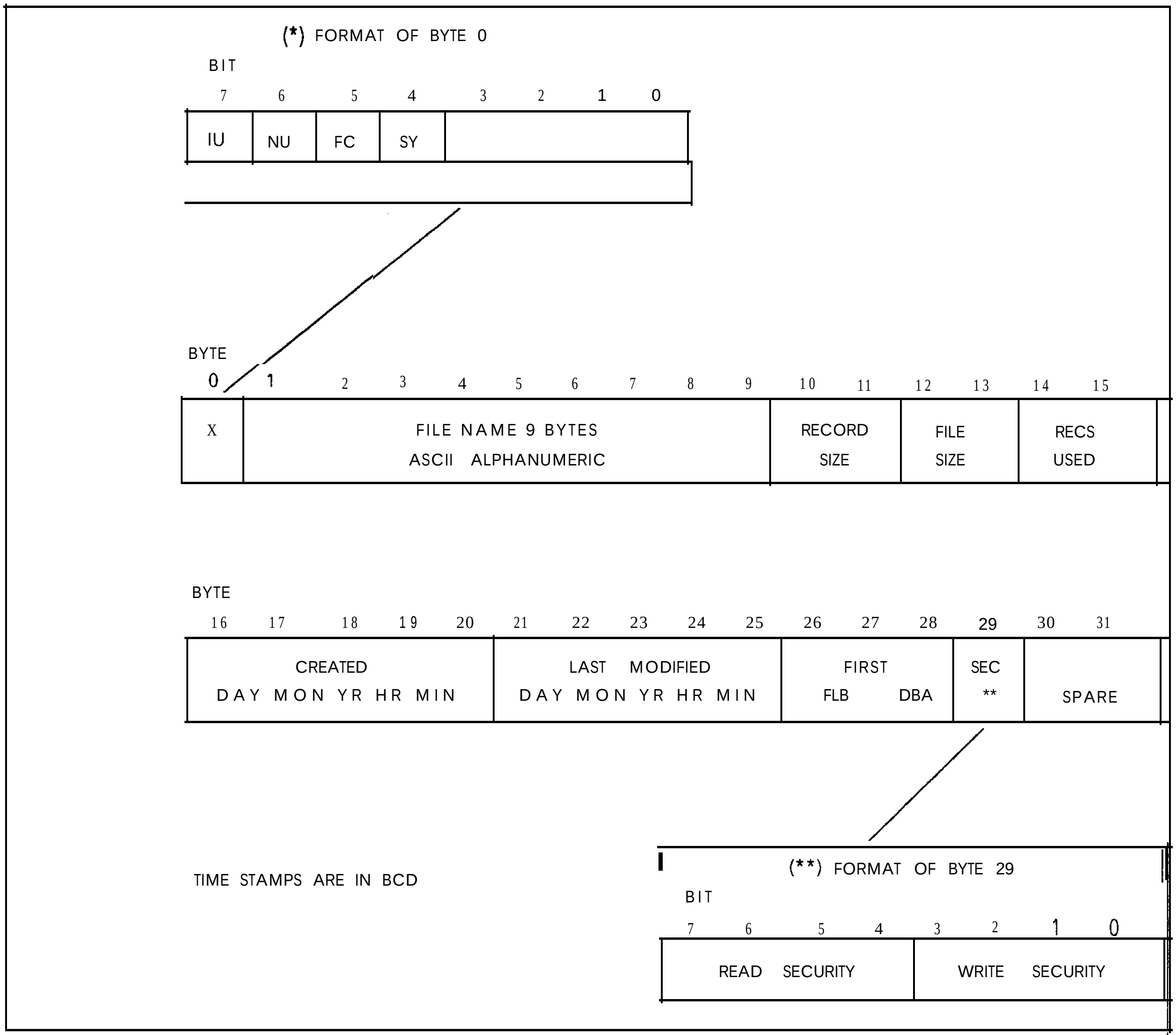


Figure 4.18 Disk File Directory Record Format

THIS PAGE IS INTERTIONALLY LEFT BLANK.

CD-100 Data Option Overview

5.0 This section describes circuit switched data.

Major Hardware Components

5.1 The CD-100 circuit data option uses the existing voice hardware to transfer data. By means of TDM (Time-Division Multiplexing), data is "hopscotched" over voice; separation of data from voice is provided for by use of a selective filter (see Figure 5.1).

The additional hardware components necessary to support this data option are:

- Digital Featurephone with data option
- DVCIP (FB-17236-A Data/Voice Control Interface Processor)
- Single pair of wires providing capability of transmitting data along with voice and control information

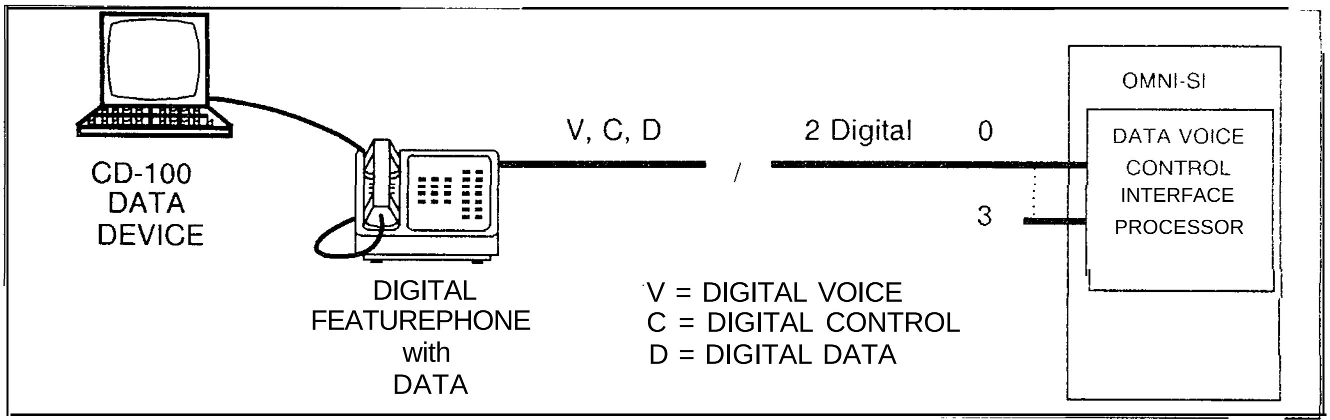


Figure 5.1 Circuit Switched Data

Operational Characteristics

5.2 Operational characteristics of CD-100 are as follows:

- Asynchronous, full-duplex data transmission up to a rate of 19.2 kilobits per second (19.2 kbps).
- Digital Featurephone connects to one ASCII terminal, an auto-answer modem, an auto-dial modem, or a host computer by an RS-232-C interface that is located in the rear of the telephone.
- DCE/DTE configuration is determined by a hardware strapping option that allows the Digital Featurephone to function in either a DCE mode (i.e., connected to a terminal and/or host computer) or a DTE mode (i.e., connected to a modem).

Conceptual Overview

5.3 The voice switch **uses a** PCM (Pulse Code Modulation) bus for voice traffic (Figure 5.2).

- Technical specification for CD-100 are as follows:
  - PCM (Pulse Code Modulation)
  - Bit clock = 1.544 MHz
  - 24 channels/PCM frame
  - 8 bits per channel
  - 1 frame synchronous bit per PCM frame
  - Total bits per frame = 24 x 8 = 192 + frame synchronous bit = 193
  - Time per frame = $\frac{1}{1.544\ \text{MHz}} \times 192 = 125$ usec
- Voice call process (Figure 5.3) for CD-100 is as follows:
  - Analog voice input is sampled every 125 usec.
  - Analog input is converted to 8 bit digital form.
  - 8 bits are placed on a digital link during an assigned channel.
  - Bits are taken from the channel at the receiver side and reconverted to analog voice.

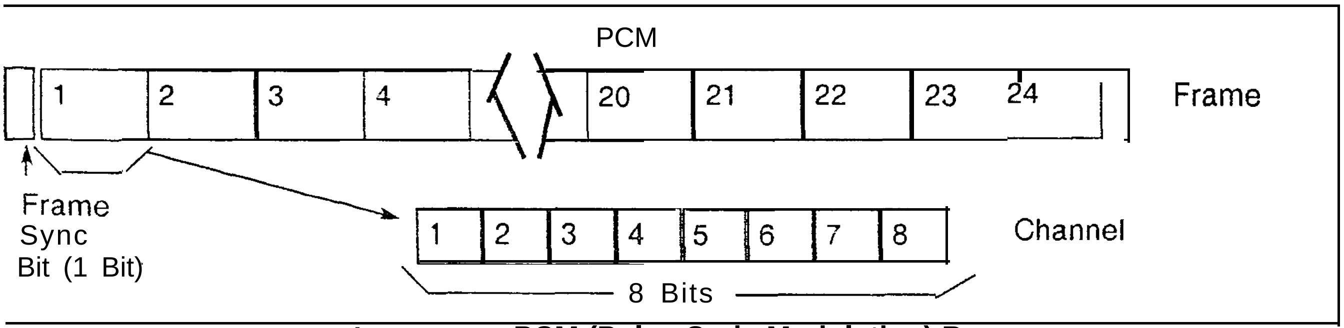


Figure 5.2 PCM (Pulse Code Modulation) Bus

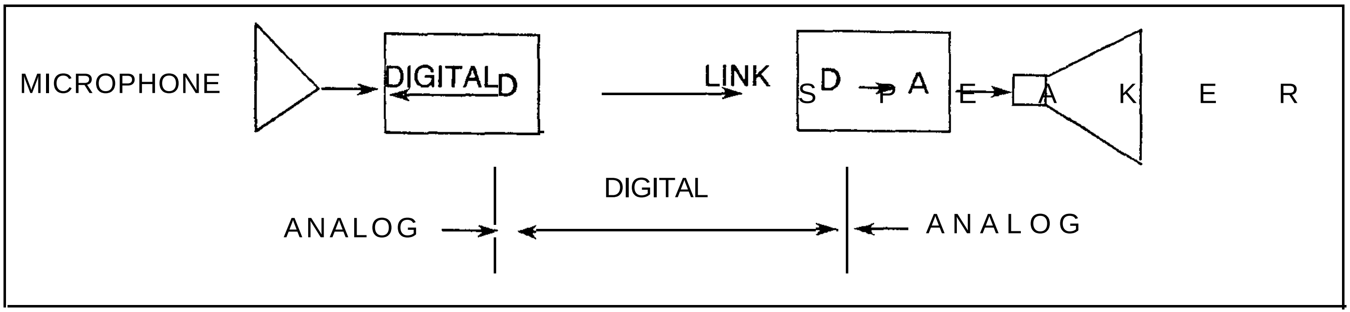


Figure 5.3 Voice Call

Data Call Process

5.4 Data calls (Figure 5.4) are processed by the following:

- Origination -- multiplex data bytes with voice bytes and place on existing digital link (PCM). This concept is possible because digital voice in 8 bytes can be thought of as data bytes. Thus, on a PCM bus, voice and data have identical properties.

- Termination -- demultiplex data bytes from voice bytes and route to data device.

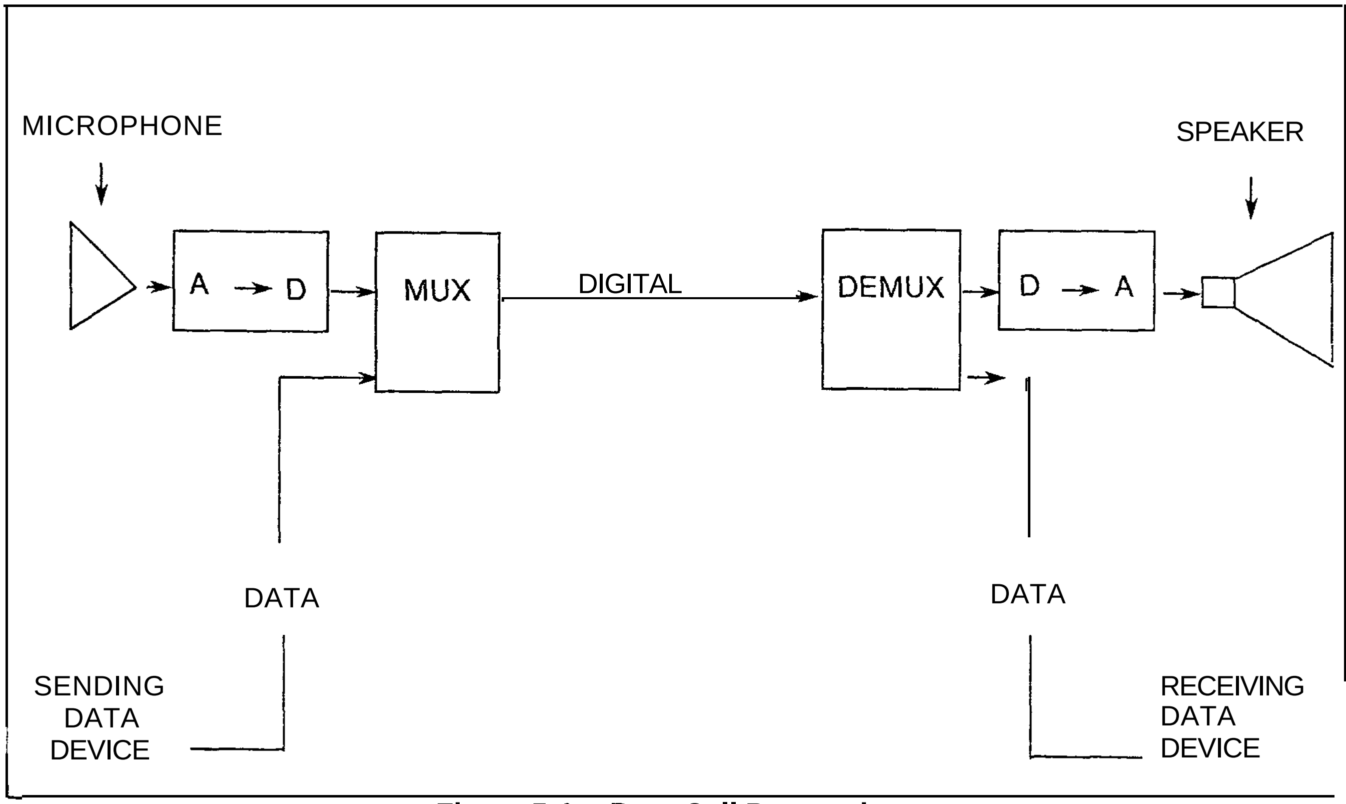


Figure 5.4 Data Call Processing

Data Device Configuration

5.5 Figure 5.5 shows how a data call is placed. Figures 5.6 and 5.7 show incoming and outgoing modem calls to the CD-100.

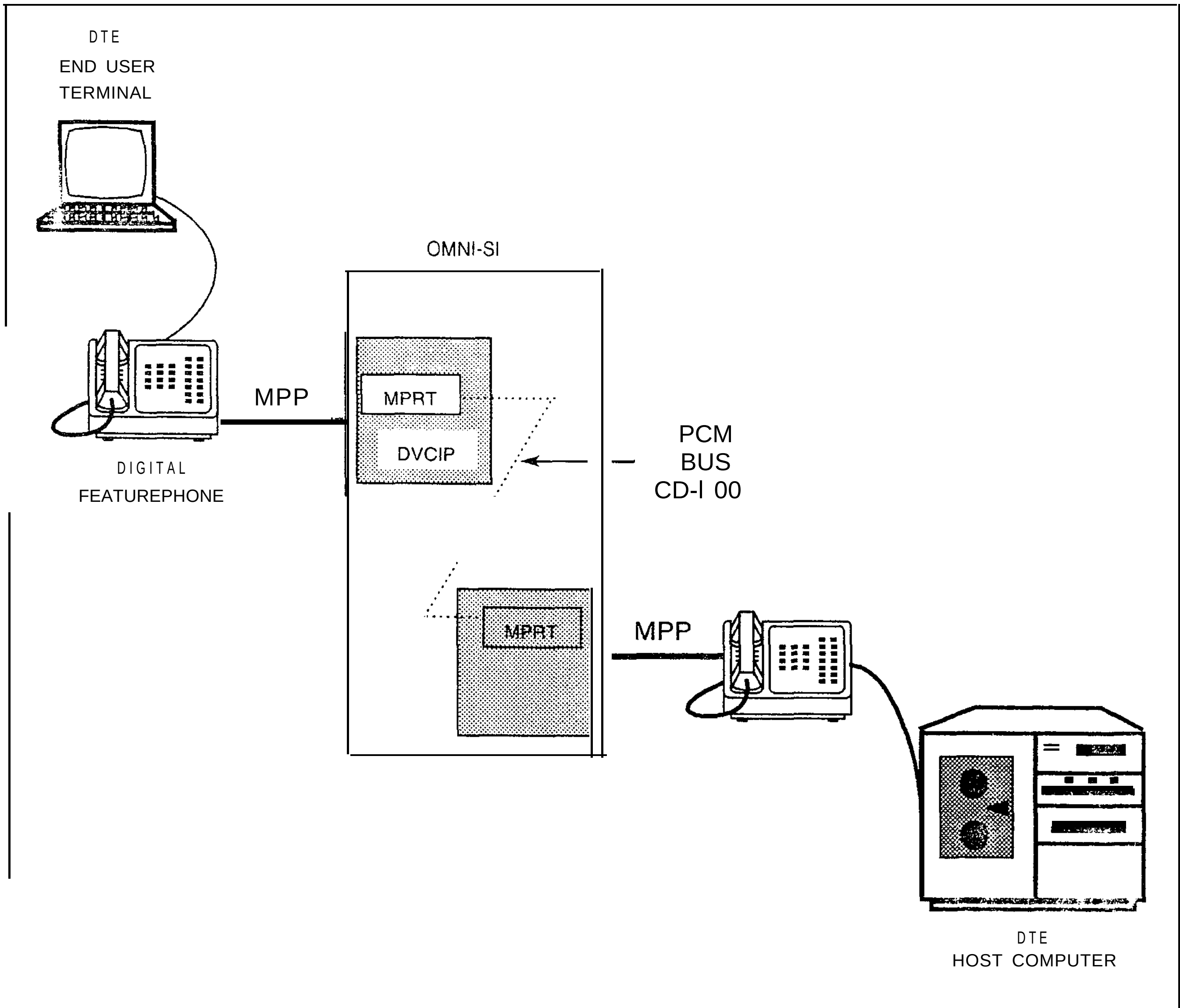


Figure 5.5 End-User Data Terminal and Host Computer Data Call

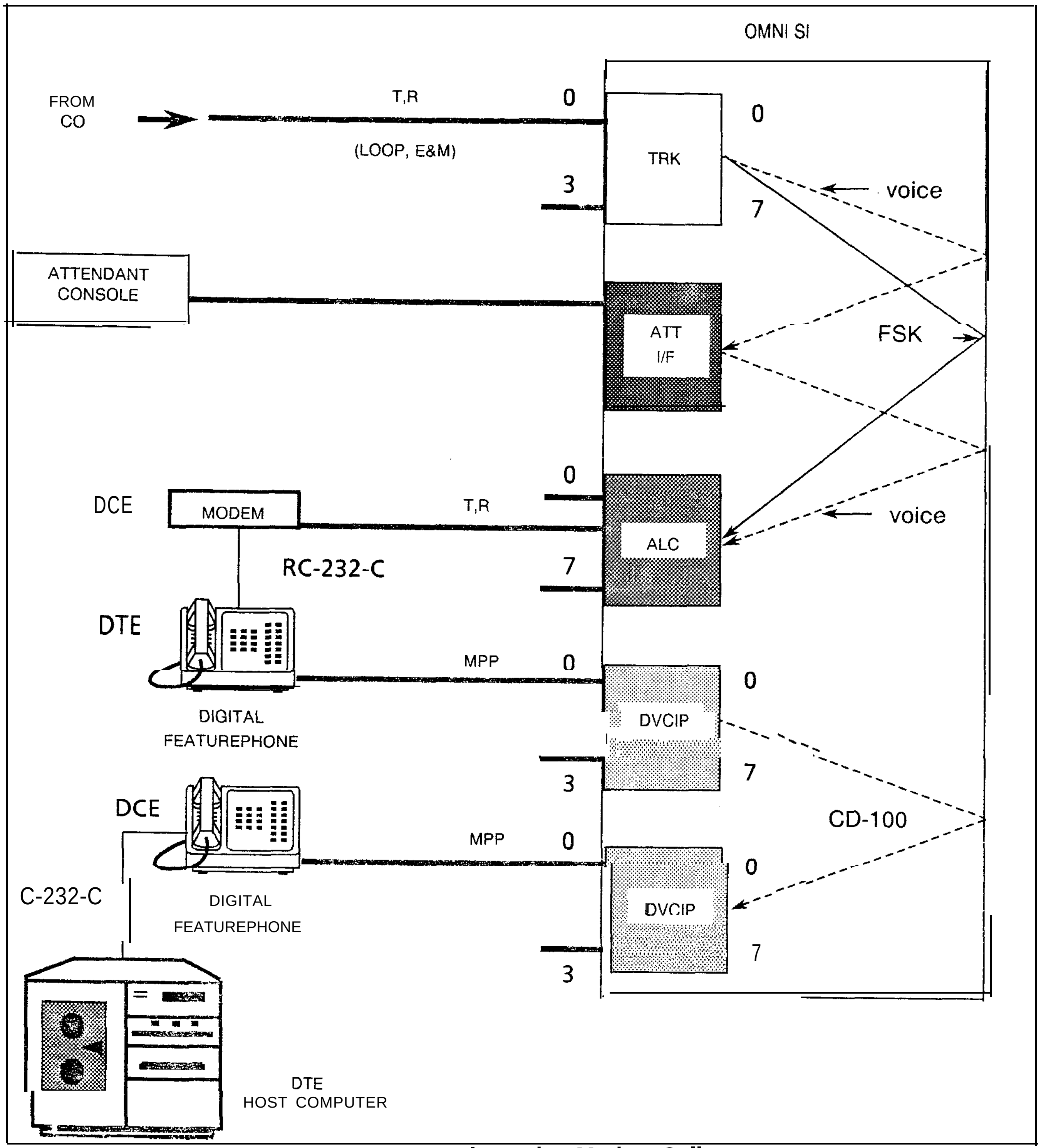


Figure 5.6 Incoming Modem Call

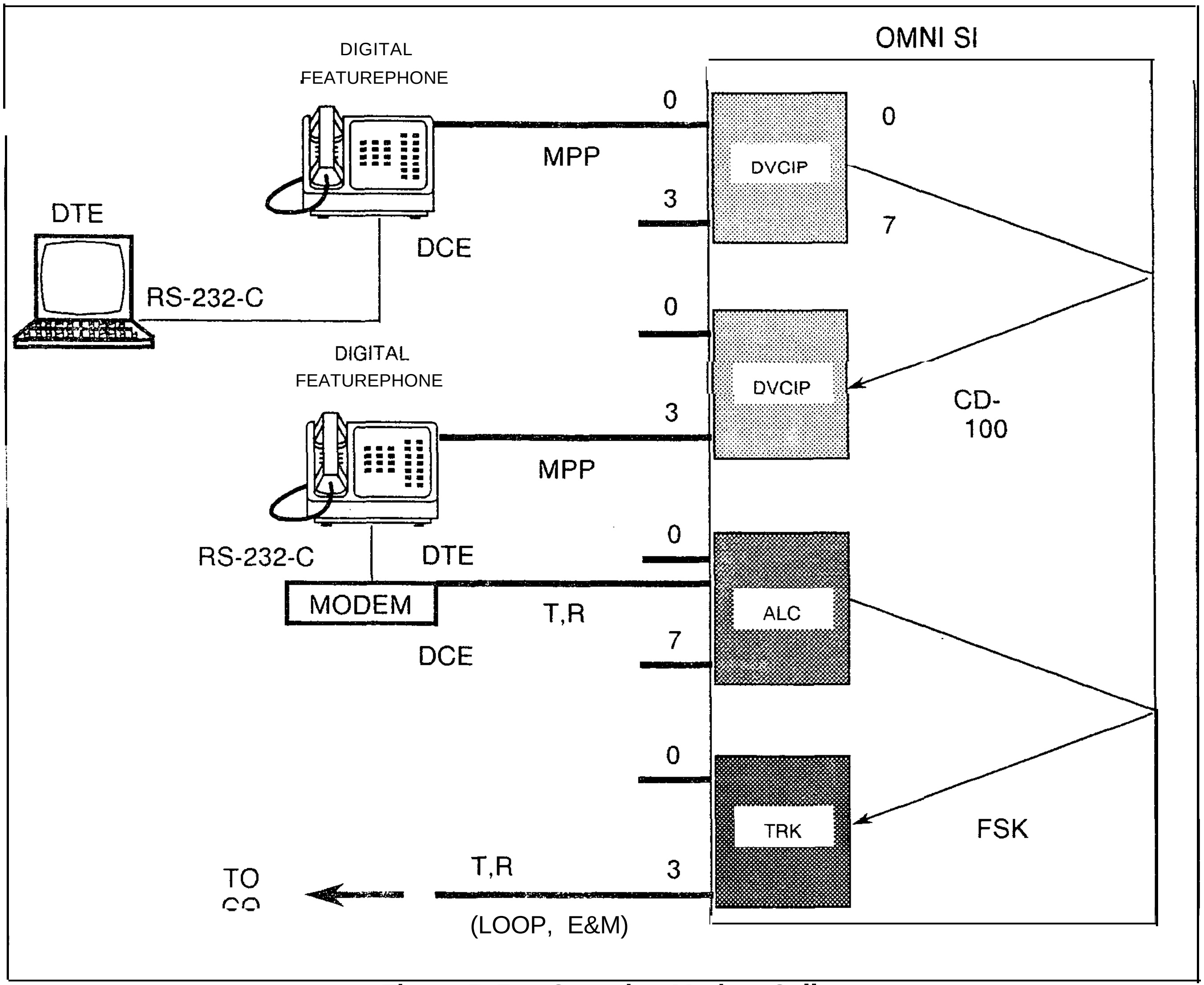


Figure 5. 7 Outgoing Modem Call

Primary Circuitry/ Components

5.6 The primary circuitry and components of CD-100 utilizes the MPRT (Mini-Packet Receiver/Transmitter) to process packet data. MPRT implements the Fujitsu GTE proprietary MPP (Mini-Packet Protocol). The MPRT, equipped in the Digital Featurephone and the DVCIP interface card, allows communication between the two (see Figure 5.8).

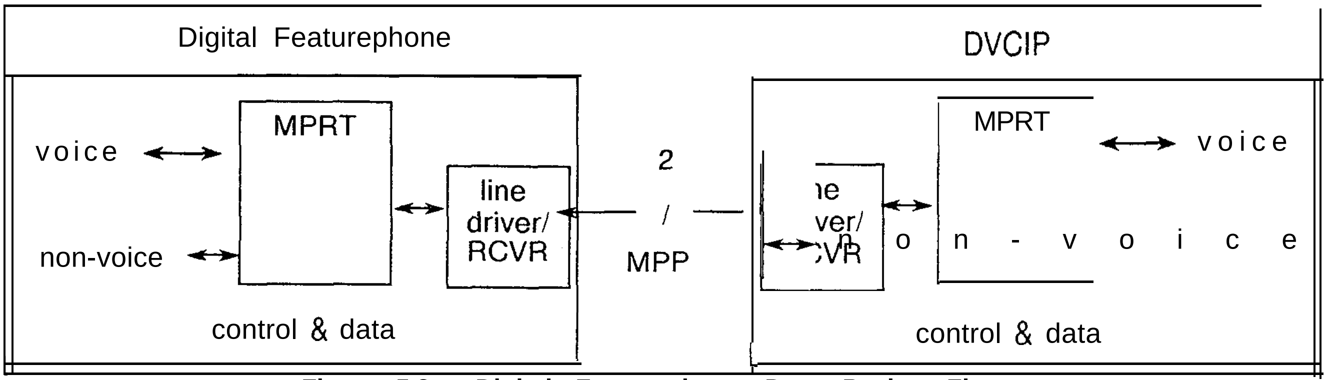


Figure 5.8 Digital Featurephone Data Packet Flow

Mini-Packet Protocol Characteristics

5.7 The following are characteristics of MPP (Mini-Packet Protocol) (see Figure 5.9).

- Link runs at 256 Kbps, ping pong (ping pong = once bus seizure is established, link runs at 128 Kbps in each direction).
- Each mini-packet consists of:
  - Sync Flag (1/2 byte)
  - Header (2 bytes)
  - Data (8 bytes)
  - CRC (1 byte)
- The first bit of first header byte determines whether the current mini-packet is voice or non-voice.
  - 1 = voice
  - 0 = non-voice

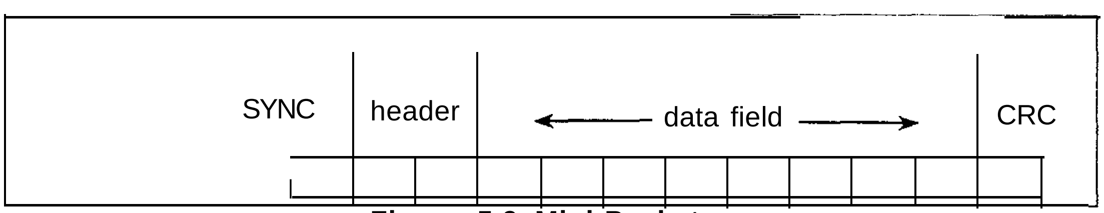


Figure 5.9 Mini-Packet

- Channels available.
  - 64 Kbps Voice
  - 19.2 Kbps Data
  - 4 Kbps Control
  - 40.8 Kbps Overhead and Guardband

THIS PAGE IS INTENTIONALLY LEFT BLANK.

PD-200 DATA OPTION OVERVIEW

6.0 In addition to the voice features available with the OMNI PABX, the PD-200 Data System (Figure 6.1) provides:

- Separate data and voice switching
  - Voice traffic uses circuit switching.
  - Data traffic uses packet switching.
- ASCII (American National Standard Code for Information Interchange) asynchronous data at speeds up to 19.2, Kbps (Kilobytes per second).
- CCITT (Consultative Committee on International Telegraphy and Telephony) X.25 synchronous data at speeds up to 64 kbps.

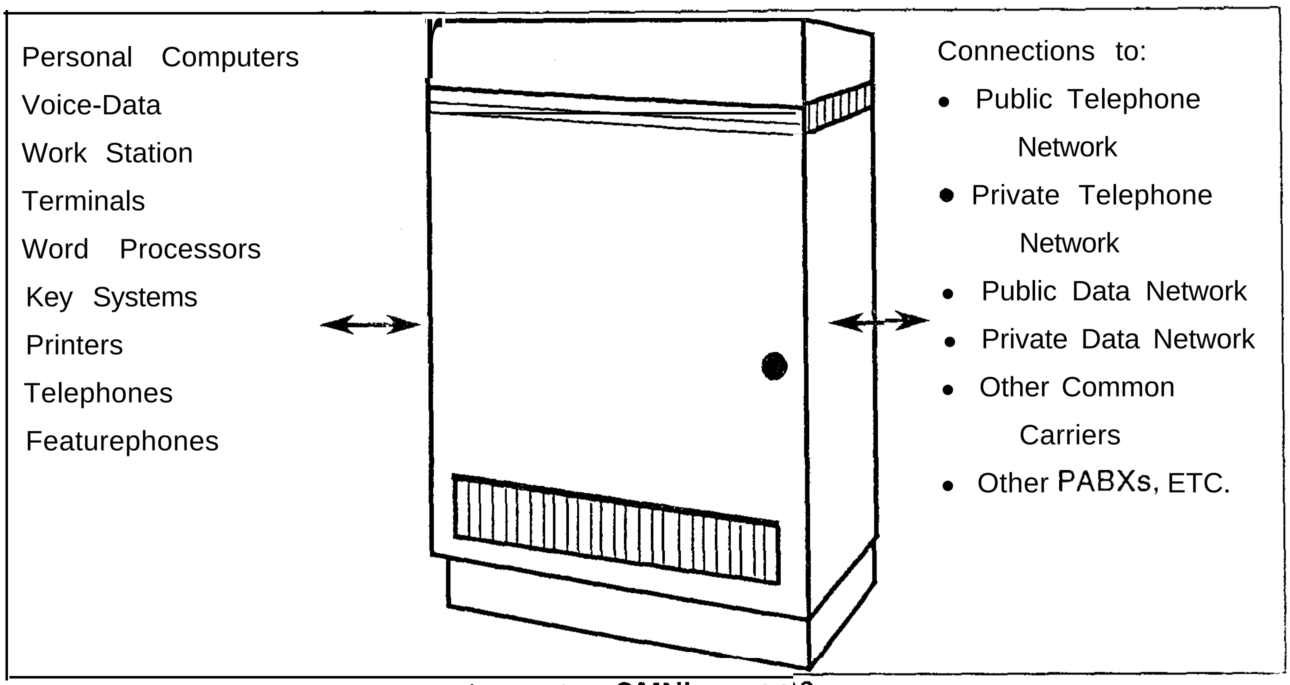


Figure 6.1 OMNI PD-200

Data System Overview

6.1 (Figure 6.1) With the PD-200 Packet Data System option, the voice and data systems are combined within the PABX. The voice and data systems share:

- Cabinet (they are both housed in the same physical location)
- Power source
- Single pair from the DFP/APMs to the FB-17246-A card

- Mass storage system of the hard and floppy disk
  - Access from the data system to the hard disk is via the SASI (Shugart Associates Standard interface) bus (Figure 6.2).
  - The hard disk accesses the generic programs and the data base.

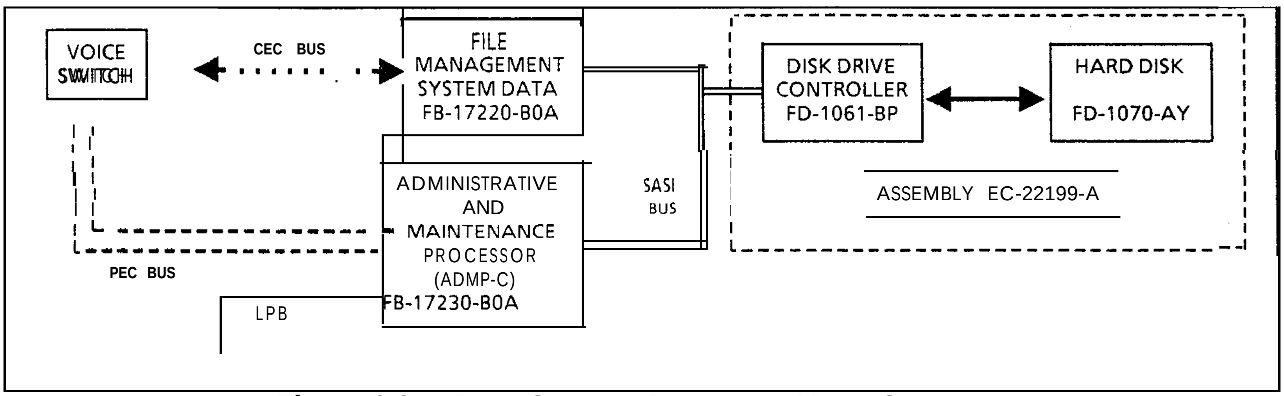


Figure 6.2 Data System Access to Mass Storage

System Buses

6.1.1 The voice and data hardware use separate buses for the switching of signals; this is referred to as dual bus architecture.

Through dual bus architecture, voice and data signals are always separate and have no effect on one another. Buses that are responsible for voice transfer and switch interaction are:

- PCM (Pulse Code Modulation) bus
- PEC (Peripheral Equipment Complex) bus

The bus used by the data system for the transfer of data packets is the LPB (Local Packet Bus).

Data Components

6.1.2 Components of the data system that are separate from the voice switch include:

- Netlink processor for control of data and call setup or takedown
- Administrative processors that maintain and administer the data switch features
- Facilities processor that performs call processing functions and interfaces with the PEC bus.

Advanced data features provided by the data system include:

- Packetization of data and control information for transfer through the system
- Speed and protocol conversion
- CCITT X.25, X.3, X.28, and X.29 signaling based standards

Administration 6.2 Data system administration functions are given as an extension of the existing voice system. The facilities that provide the administration functions are based separately. Access to these functions, which include Recent Change, maintenance, accounting, diagnostics, and event reporting, is through the voice switch maintenance console and a data maintenance console, that interfaces the system via the ADMP (Administrative Maintenance Processor) (Figure 6.3).

NOTE: A separate console for data is required.

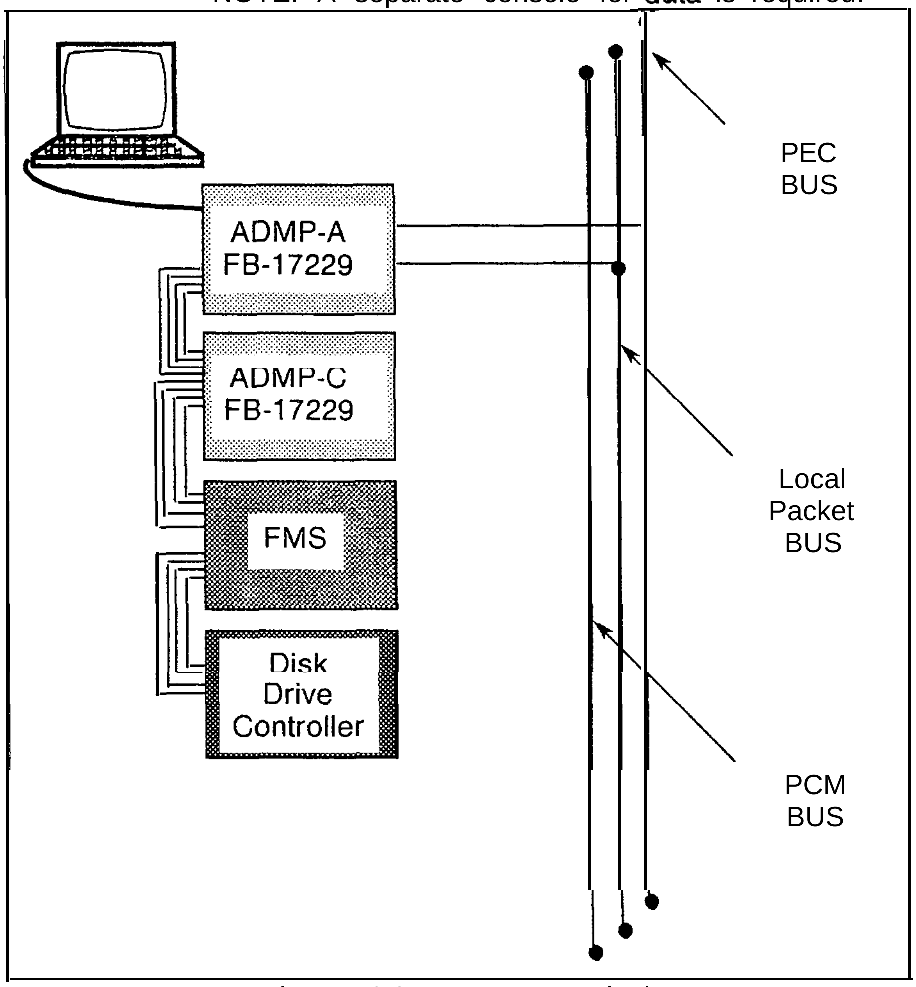


Figure 6.3 ADMP Terminal

NOTE: The ADMP is a two-card set: the maintenance terminal connects to the A card. Any terminal used with the data system must be VT > 101 compatible.

Data Transport

6.3 Data switching that does not affect voice capacity is possible because of the PTS (Packet Transport System). A new bus structure, called the LPB (Local Packet Bus), has been added to the backplane. The LPB is only for data transfer (Figure 6.4).

Self-routing entities, called mini-packets, perform communication within the packet transport system (see paragraph 6.4). Each device communicates with the switch by means of the PTS (Packet Transport System), and is assigned its own PLA (Packet Line Address).

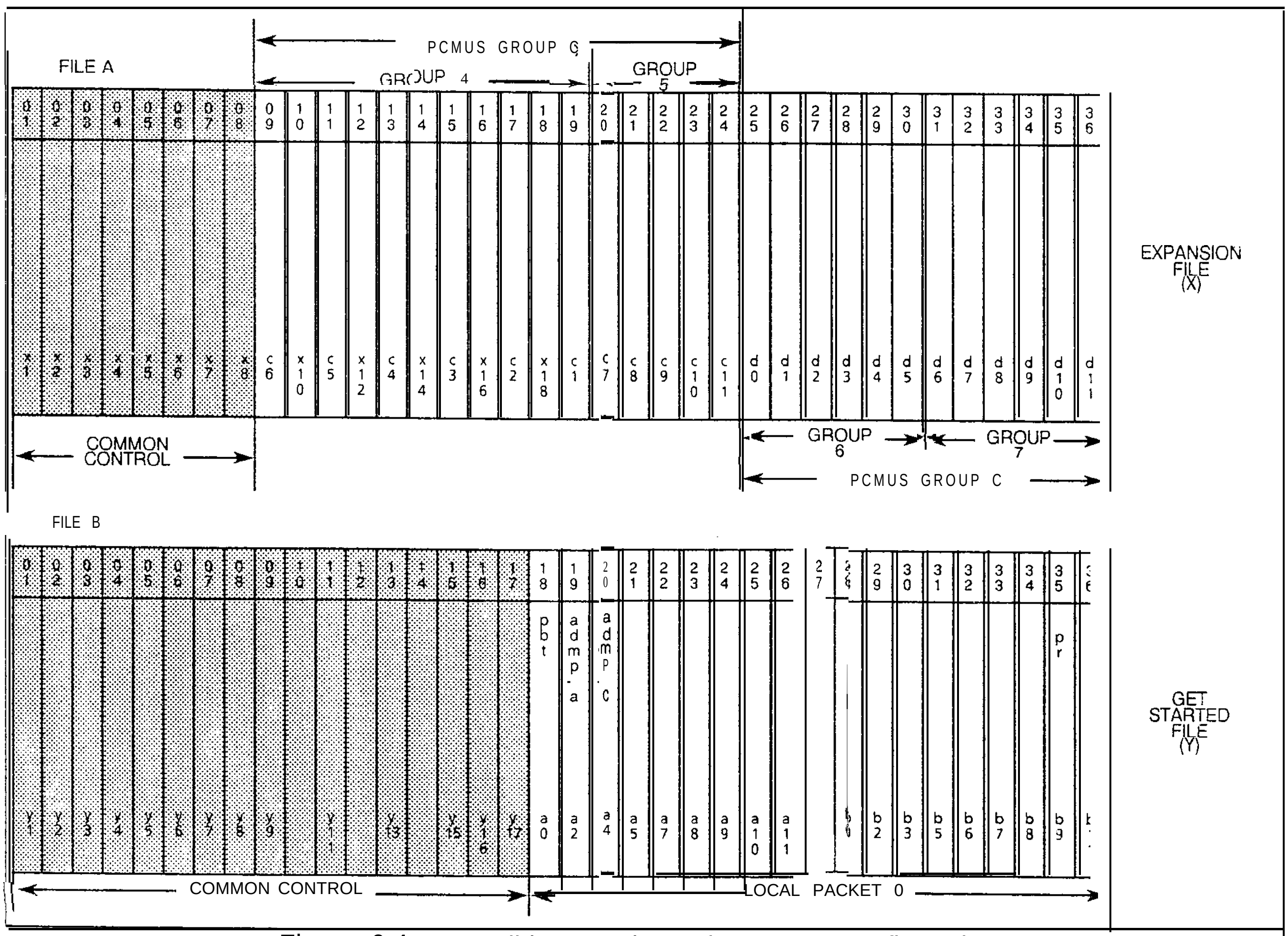


Figure 6.4 Possible Local Packet Bus Configuration

Packet Line Address

6.3.1 This paragraph describes the PLA (Packet Line Address).

- The PLA gives the internal address of the component to which it is assigned:
  - This assignment is performed automatically by the data system.
  - The assignment is based on the hardware configuration of the system.
- The PLA has no relationship to the subscriber line number that the user sees.
- The PLA is made up of 14 bits of address information.

Devices that communicate over the PTS use a FGBS proprietary or internal protocol, known as MPP (Mini-Packet Protocol), to ensure error-free communication. CRC (Cyclical Redundant Checking) performs error detection.

NOTE: A proprietary protocol is one that need not comply to any CCITT recommendations because it is internal to the PABX.

Local Packet Bus interfaces

6.4 The following major components connect to the LPB (Local Packet Bus) and require a PLA (Packet Line Address).

NOTE: Cards that transport data packets are described later in this document.

- UCB/DCP (Universal Controller Board) requires one PLA. This card provides the processing power needed to supply the features of the Digital Featurephone and the data system, such as call setupitakedown.
- ADMP (Administration and Maintenance Processor) requires one PLA. This card provides administrative and loading functions.
- NIC requires one PLA. This card provides a network interface.
- VPLC (Voice Packet Line Card) requires one PLA for each circuit activated (8 maximum).

The VPLC (card type VPLO and VPL1) connects remote processing devices to the PABX. These devices are:

- APM (Asynchronous Packet Manager) which connects to asynchronous devices that operate at lower speeds
- SPM (Synchronous Packet Manager) which connects to synchronous devices that usually operate at a higher speed (9.6-64 Kbps.)
- DFP/APM (Digital Featurephone with APM) which connects to asynchronous devices that usually operate at a higher speed

The VPLC2 (card type VP20) connects remote processing devices to the PABX. These devices are:

- APM which connects to asynchronous devices that operate at lower speeds
- SPM which connects to lower speed (up to 19.2 Kbps.) synchronous devices
- DFP/APM which connects to asynchronous devices that operate at lower speeds

The VPLC2 (card type VP21) connects remote processing devices to the PABX. This device is:

- SPM which connects to higher speed (9.6-64 Kbps) synchronous devices

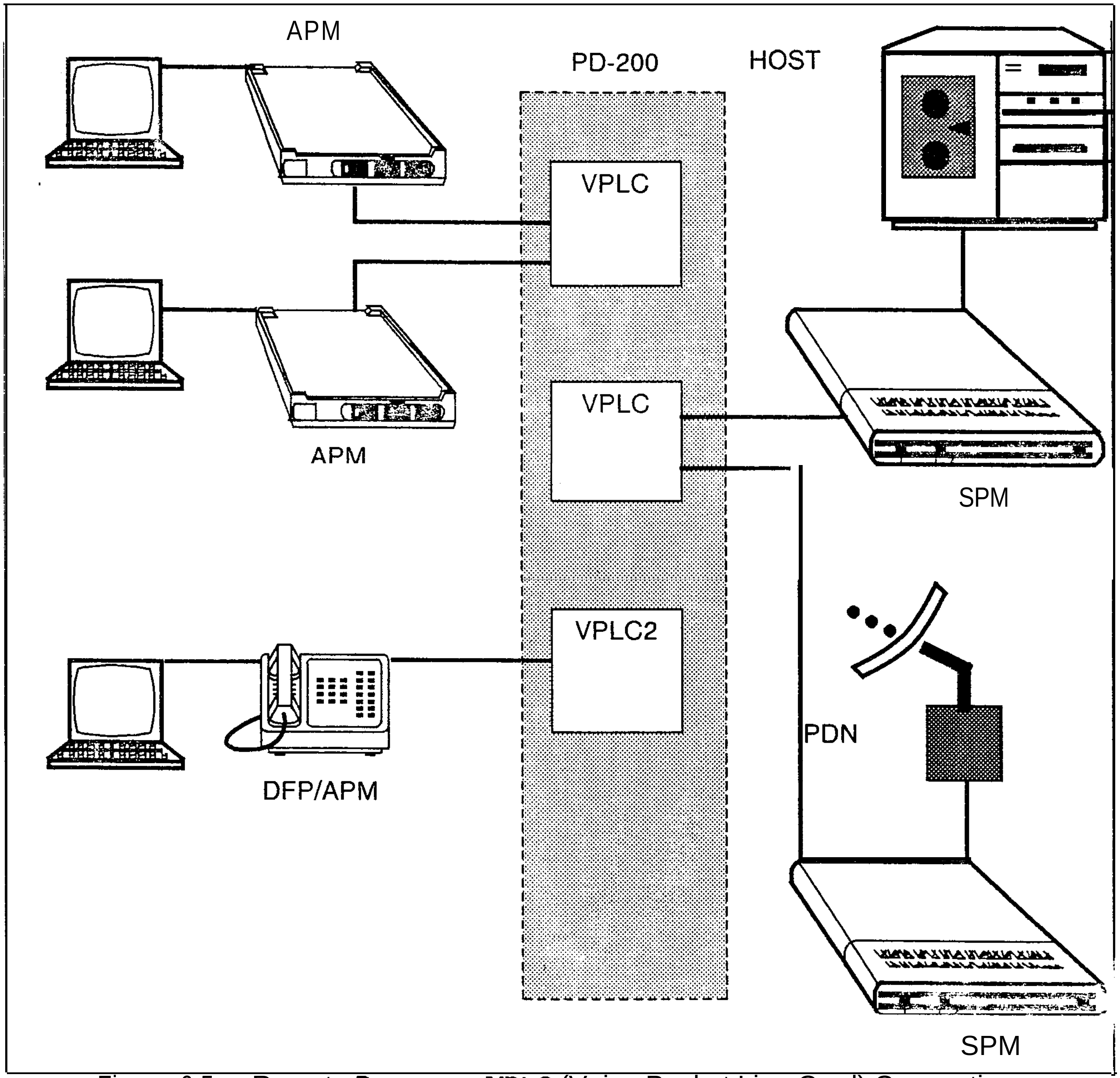


Figure 6.5 Remote Processor/VPLC (Voice Packet Line Card) Connection

Data Architectural Overview

6.5 This paragraph provides an overview of the data. architecture.

Asynchronous Connection

6.51 Asynchronous devices (terminals, hosts, and modems, etc.) connect to the system via APMs (Asynchronous Packet Managers) or DFP/APMs, which are collocated with their devices. Except for call-control-related events (call setup, disconnect, etc.), APMs contain all processing and transmission capabilities necessary to support transmission of asynchronous data. In addition, the APM provides a CCITT X.3 packet assembler/disassembler function. This converts asynchronous data into CCITT X.25 packets and allows signaling information (commands) to be entered via the keyboard on the attached asynchronous device. Local commands are supplied by using a CCITT X.28-type command language (not fully X.28 compatible). Remote commands are supplied by using a CCITT X.29-type command language (Figure 6.6).

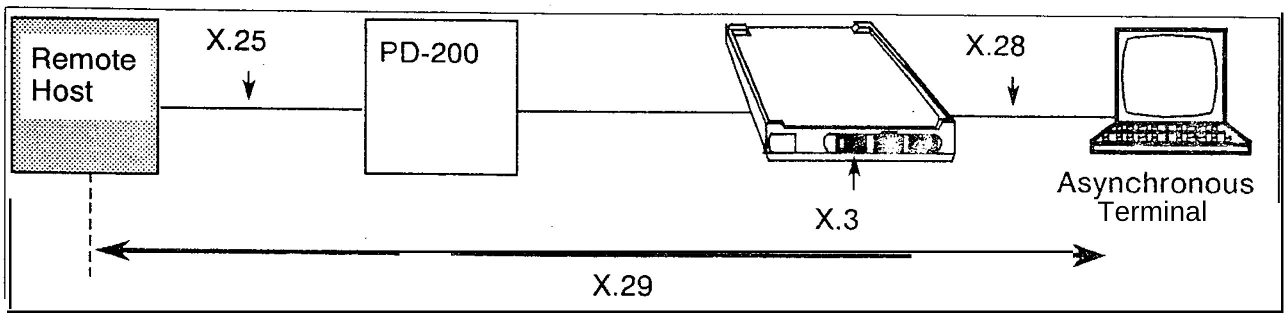


Figure 6.6 X.29 Protocol

The APMs are addressed by using the X.121 numbering scheme, one X.121 number per APM.

Each APM supports one single active data call (known as a virtual circuit). An APM can interface to half or full duplex devices operating at speeds up to 19.2 kbps. The actual data throughput varies according to the PAD (Packet Assembler Disassembler) parameters' settings and data format.

The APM communicates with the system by means of the PTS (Packet Transport System). Each APM connects to its device with a RS-232-C type connector and can be configured as DTE (Data Terminal Equiomoent) or DCE (Data Communications Equipment).

Synchronous Connection

6.5.2 Synchronous CCITT X.25 devices (hosts and public data networks) connect to ‘the system via SPMs, which are col-located with their devices. Connection from the SPM to its device is via an RS-232-C type or V.35 type connector. The W-232-C connectors are used for devices operating at less than or equal to 19.2 kbps, while devices operating at up to 64 kbps use the V.35 connector interface. SPMs can be configured as a DTE or DCE. Each SPM is assigned one or more CCITT X.121 addresses, depending upon the routing desired. CCITT X.25 level 3 (packet level) and level 2 (link level, LAP-B) protocols are supported.

PD-200 PRINTED CIRCUIT BOARDS

7.0 This section provides descriptive information about PCBs installed in a system supporting the packet data option (see Figure 7.1).

All PO-200 cards are installed into PCMUS (Pulse Code Modulation Universal Slot) card slots. All data cards occupy one card slot.

In addition to the hard disk drive, floppy disk drive, and FB-17220-B0A File Management System Data (FMSD) card, which are a part of the basic voice system, the PO-200 Data System can include the data cards listed in Table 7.1. Cards used will vary with the size and peripheral use with the system. Universal card slots house both common control and peripheral interface cards for the PO-200 Data System. Card slots not used by cards supporting the data system can be used by cards supporting voice features. Figure 7.2 provides a block diagram of the card interaction.

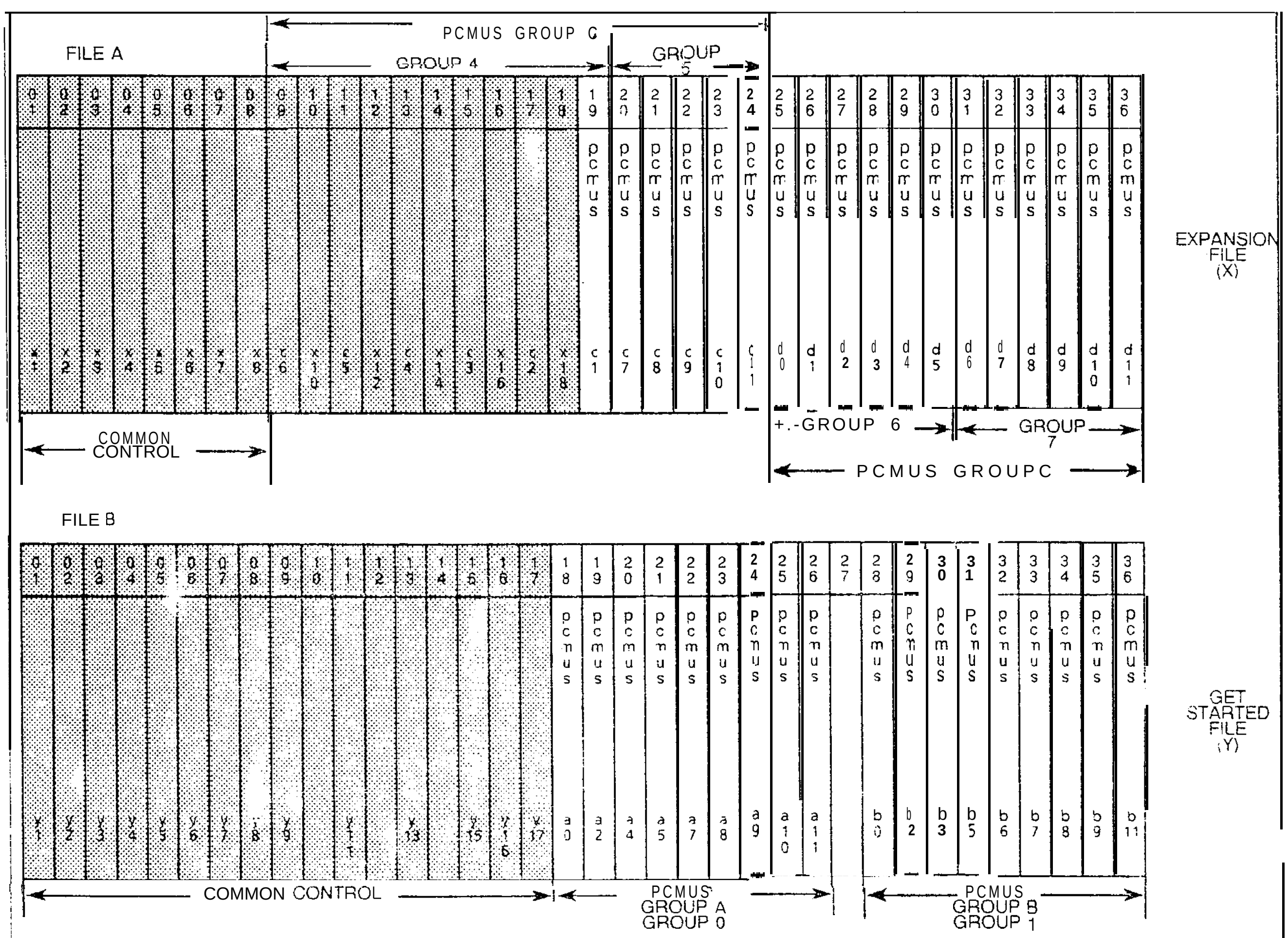


Figure 7.1 Universal PC8 Card Slots Used by PD-200 Cards

NOTE: Card slot 11 cannot be used by data cards.

Table 7.1 **PD-200 Packet Data Cards**

FB#	MNEMONIC	PCMUS CARD FUNCTION
FB-17229-A	ADMP-A *	Administrative and Maintenance Processor
FB-17230-BOA	ADMP-C *	Administrative and Maintenance Processor
FB-17231 -A	UCB/DEP	Universal Control Board
FB-17228-B0A	PR	Packet Router
FB-17227-A	PBE/T	Packet Bus Extender/Terminator
FB-17242-A	NIC	T1 Network Interface
FB-17226-A	VPLC	Voice Packet Line Card (type 0 or 1)
FB-17246-A	VPLC2	Voice Packet Line Card 2 (type 0 or 1)
FB-17242-A	NIC	Network Interface Card
* = two card set		

ADMP (ADMP-A/ FB-17229-A ADMP-C FB-17230-BOA)

7.1 The ADMP (Administrative and Maintenance Processor) is a two-card set made up of the FB-17229-A (ADMP-A) and the FB-17230-BOA (ADMP-C). A ribbon cable, which cannot exceed 6 inches, connects the two cards. The ADMP-A and ADMP-C are installed in two adjacent mounting spaces. Since the ADMP-C card does not require power, it does not require a PCMUS card slot.

NOTE: Installation of either ADMP card into a Tl-span card slot is not recommended.

The cards run the administrative software supporting the data system and consist of two microprocessors connected by an area of common memory.

- The NETLINK microprocessor controls the interface with the LPB (Local Packet Bus) and operates the protocols needed to communicate over it.
- The second microprocessor performs the administrative processing and controls the file management system interface.

Card FB-17229-A (ADMP-A) contains all of a NETLINK microprocessor circuitry, a portion of a second microprocessor circuitry, and shared memory and the ADMP terminal interface connection. Card FB-17230-BOA (ADMP-C) contains the majority of the second microprocessor's circuitry.

The ADMP-A card contains the FGBS 65SC102 microcircuit, while the ADMP-C card contains the Intel 8088 microprocessor.

UCB/DCP
FB-17231 -A

7.2 The UCB (Universal Controller Board) houses the DCP (Data Call Processing) software. This software runs the data switching needed by the DFP/APM (Digital Featurephone with Asynchronous Packet Manager), APM, SPM, and the system.

The UCB/DCP is divided into two major eight-bit microprocessor circuits.

- The NETLINK microprocessor is used for communicating with the LPB (Local Packet Bus). It also runs the necessary protocols for communications over the LPB.
- The facilities microprocessor performs the data call-processing functions for the PD-200 Data System. The facilities microprocessor is a slave to the NETLINK microprocessor and controls the low-level bus interface and feature processing for the DFP/APM, APM, SPM, and NIC.

Both circuits of the card communicate through a block of RAM (Random Access Memory) accessible by both microprocessors on a time-shared basis. The two microprocessors communicate with each other via common memory; the netlink processor functions as the master.

All software on the UCB communicates with other cards within the PD-200 Data System via the transfer of messages across the LPB; other communication is over the PEC low-level backplane bus.

NOTE: The UCB can also be used to terminate an LPB, which is normally done by the PBE/T (Packet Bus Extender/Terminator) card. For the UCB to perform its normal functions as well as those of the PBE/T, it must be located in a slot defined for PBE/T and be so defined in the data base. If the UCB is not configured as a bus terminator, it can be placed onto only PCMUS used by data cards.

When depressed, the front panel RESET SWITCH causes a UCB card reset. If self-test or error recovery software determines that the UCB card is not operating properly, the front panel LED flashes at a high rate (60 IPM). During software loading, the LED flashes at the rate of 30 IPM.

PR
FB-17228-BOA

7.3 The PR (Packet Router):

- Electrically enables LPB 0

  - The PR directly accesses LPB 0 off the backplane.
  - Since the PR activates LPB 0, the card must be at one end of that bus. It is recommended to place the PR in card slot CO8 or D10.

  NOTE: LPB 0 connects to the PR directly off the backplane. This bus can be located in either file.

- Receives and transmits mini-packets on LPB 0 and LPB 1

  - Each PR can support 2 LPBs (LPB 0 and LPB 1).
  - LPB 1 is accessed by the PR via the PBE. The connection from PR to PBE is called the bus extender cable.

  NOTE: The PR has four front edge tab connectors. When used in an OMNI SI, only one of these is used, that being the connection for the bus extender cable.

PBE/T
FB-17227-A

7.4 The Packet Bus Extender/Terminator card electrically enables or terminates an active LPB (local packet bus).

The card can be configured as either a PBE (packet bus extender) or a PBT (packet bus terminator).

- When configured as a PBE:

  - the card enables LPB1; this connection is directly off the backplane of the system

  - the PBE is only used when access to the second LPB (LPB1) is needed

  - by enabling LPB1, the PBE allows a PRE (packet router enhanced) to gain access to the second LPB (LPB1)

  - the access for the PRE to the PBE is via a front edge cable connection call a bus extender cable

- When configured as a PBT:

  - the card terminates a bus
  - a PBT is required for both LPBs (LPBO and LPB1)
  - the PBT must only be put into certain PCMUSs
  - a UCB card can function as a PBT as well as performing its UCB functions

NIC
FB-17242-A

7.5 The NIC (Network Interface Card) in coordination with a T1 trunk card set, allows PD-200 packet data to be sent over digital common carrier lines to a remote PD-200 system. The card contains two microprocessors. One microprocessor communicates with the LPB., and the other with the PCM bus.

Mini-packets sent to the NIC card from the various peripherals are converted to true X.25, HDLC-type (High-Level Data Link Control) packets. These packets are in turn “chopped” into seven or eight bit “chunks” and inserted into the backplane PCM stream, much as a CODEC (Coder/Decoder) would. Then the data is circuit switched to the T1 trunk card set for transmission to a remote switch. This allows transcontinental networking of PD-200 systems.

Conversely the NIC can receive “bursted” X.25 packets from the PCM network. It can put these “bursted” pieces together and then convert them into mini-packets. The mini-packets can then be sent down the LPB to the packet router for distribution to the various peripherals.

The NIC card requires a nailed connection.

VPLC
FB-17226-A

7.6 The VPLC (Voice Packet Line Card) used for data transfer only provides the interface between the system LPB, PEC bus, and external synchronous/asynchronous packet managers.

The card has eight circuits and is software programmable as either a type 0 or type 1 card.

- When programmed as type 0 (card type VPLO):
  - the card can use all of its eight circuits
  - the card can transport asynchronous data at a rate of 19.2 Kbps (kilobits per second) and/or
  - the card can transport synchronous data at a rate of 9.6 Kbps
- When programmed as type 1 (card type VPL1):
  - the card can only use two of its eight circuits (due to power restraints)
  - the card can transport synchronous data at a rate of 64 Kbps and/or
  - the card can transport asynchronous data at a rate of 19.2 Kbps

NOTE: Normally an unused circuit on a type 1 card would not be used to support asynchronous data or low-speed synchronous data, but can be used if required.

VPLC2
FB-17246-A

7.7 The Voice Packet Line Card 2 is primarily used for voice and data transfer interfacing the LPB, PEC bus, and DFP/APM (Digital Featurephone with Asynchronous Packet Manager).

The card has eight circuits and can be software programmed as either a type 0 or type 1 card:

- When programmed as type 0 (card type VP20):
  - the card can use all of its eight circuits
  - the card can transport asynchronous data (and voice) at a rate of 19.2 Kbps, and/or
  - the card can transport synchronous data (and voice) at a rate of 19.2 Kbps
- When programmed as type 1 (card type VP21):
  - the card can only use two of its eight circuits (due to power restraints)
  - the card can transport synchronous data (and voice) at a rate of 9.6-64 Kbps, and/or
  - the card can transport asynchronous data (and voice) at a rate of 19.2 Kbps

Table 7.2 provides a quick-reference chart of data card functions and characteristics.

Table 7.2 PD-200 Data Card Functions

Card		Function
FB-17229-A	(ADMP-A)	• Controls the interface to the local packet bus and operates protocols needed to communicate over it.
FB-I 7230-B0A (Two-card set)	(ADMP-C)	• Controls administrative processing and file management system interface. • ADMP-C is the only data card that does not electrically connect to the LPB.
F-B-I 7231 -A	(UCB/DCP)	• Data call processing software is resident on this board and communicates with other PD-200 cards via the LPB. • Dual-function board which operates PD-200 data switching features. Can also be used as PBT, to terminate an LPB.
FB-17228-B0A	(PR)	• Receives/transmits mini-packets on the two LPBs while maintaining a continuous flow of data to/from data switch users.
F-B-I 7227-A	(PBE/T)	• Allows a packet router to gain access to the second LPB which it is to control • Also used to electrically terminate a LPB.
FB-1 7242-A	(NIC)	• Provides a link between the LPB and the system's PCM (voice switching) bus. • Allows PD-200 packet data to be sent over digital common carrier lines to a remote PD-200 system. • Provides channel T1 carrier for data. • Provides X.25 protocol. • Acts as a trunk and connects T1 from one OMNI SI to T1 in another.
FB-17226-1A	(VPLC)	• Provides interface between APM/SPM, LPB, and PEC bus. The PEC connection provides control of the card.
FB-17246	(VPLC2)	• Provides interface from APM/SPM or DFP/APM to the LPB, PCM, and PEC buses

- System Generic Disk
- System Data Base Disk
- Options Disk I, Overlay Programs/MDR
- Options Disk III, Diagnostic
- Options Disk IV, Data Switch Generic
- Options Disk V, Data Switch Generic

NOTE: File contents of disks are subject to change due to new SVRs and/or customer alterations.

ADMP-C accesses the system hard disk via the SASI (Shugart Associates Standard Interface) bus and the FMS (File Management System) card.

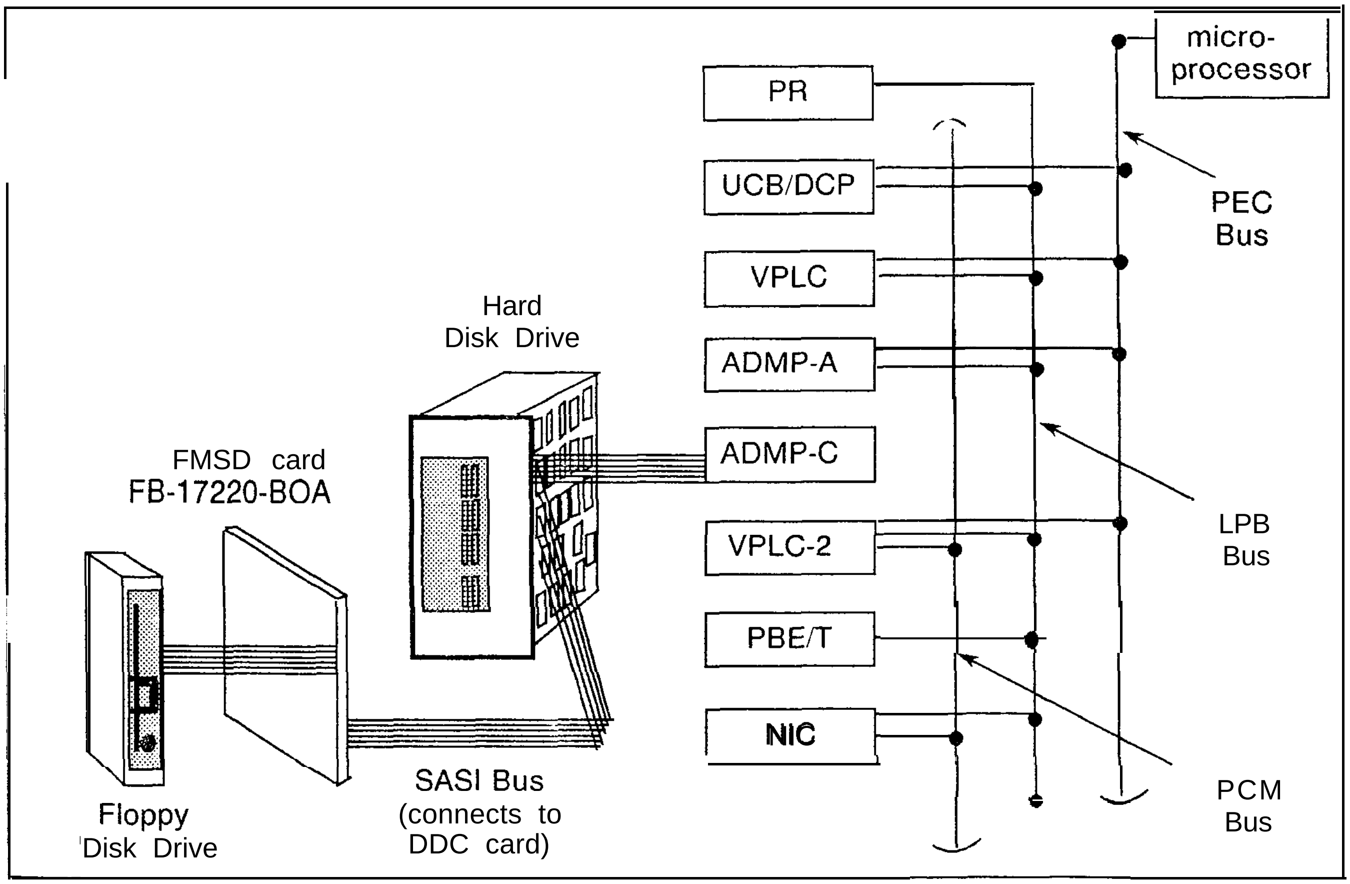


Figure 8.1 FMSD Interface

NOTE: The FMSD card is interfaced to the PD-200 Data System ADMP-C card via the DDC (Disk Drive Controler) that is attached to the hard disk.

File Management System Data

8.2.2 With the data option, the FMSD provides an interface for the voice system, microprocessor, and disk drive controller as well as the data system (see Figure 8.2, 8.3). This interface is through the ADMP via the SASI (Shugart Associates Standard Interface) bus. The SASI bus is a ribbon cable.

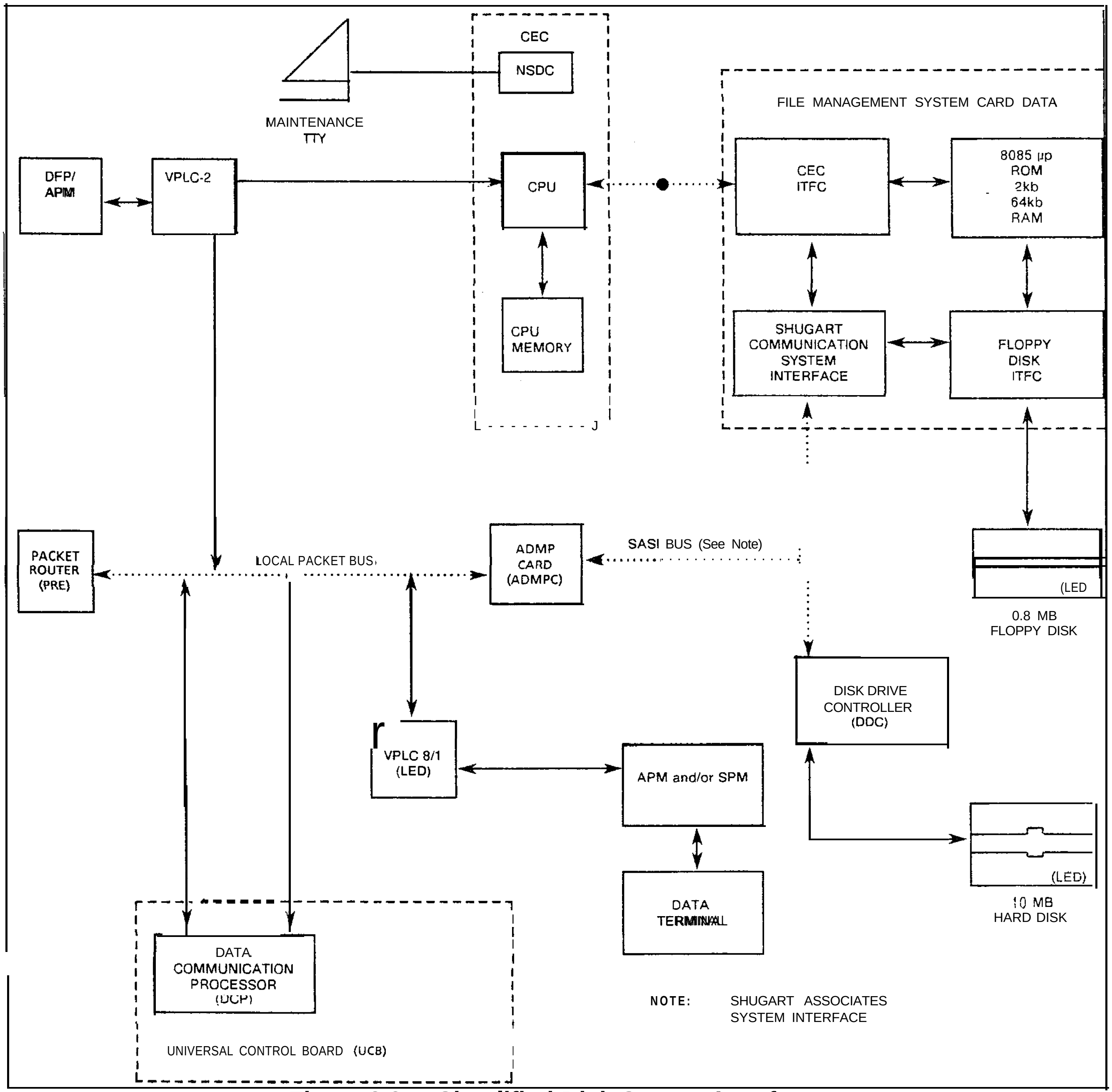


Figure 8.2 Simplified Disk System Interface

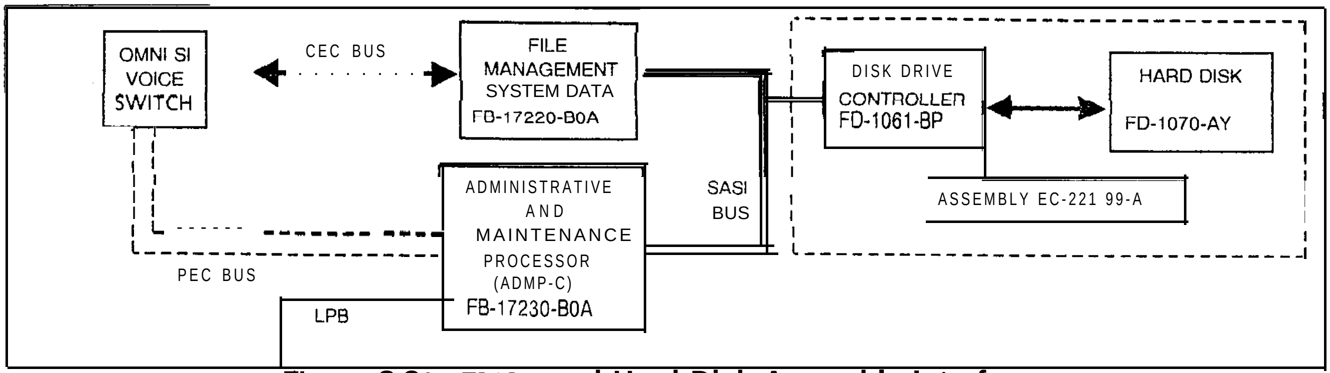


Figure 8.3 FMSD and Hard Disk Assembly Interface

File Management System Data Commands

8.2.3 The file management system's files are described in full in paragraph 4.9.3. FMS commands that are used only for the data option are as follows:

- Administration and Maintenance Command Descriptions:
  - Format Command. The Format command can initialize a blank disk for use by the FMS.
  - Reset Command. The Reset command requests FMS to close all open files and release all FIDs which this port (system or ADMP) has opened and seized. Devices which the port has dismounted are mounted, and devices marked private are marked public.
  - Diagnostic Command. The Diagnostic command can check the integrity of the FMS and associated hardware.
  - Configuration Command. The Configuration command allows the user to determine what devices are attached to FMS.
  - Time Command. The Time command allows the system or the ADMP to set the FMS's internal clock, which is used to time stamp file accesses.
  - Mount Command. The Mount command is used to allow access to a device. As a subcommand, Mount marks a device as public or private. Devices marked as private will only allow access from the processor which mounted the disk. After the Mount command is issued for a device, FMS will allow files to be opened on that device.

- Dismount Command. The Dismount command is used to prevent access to a device. After the Dismount command is issued for a device, FMS will not allow files to be opened on that device. As a subcommand, the device can be marked or retained as a private device, preventing the other processor from mounting the device. Before removing a floppy disk, this command should be used to lock out unwanted access to the disk. Removing a disk on which files are open can result in a loss of data.

SASI Bus

8.2.4 The SASI bus passes blocks of data between the ADMP and the FMSD. The SASI bus connects the FMSD to the DDC. Refer to Figure 8.4. The ADMP connects to the middle of the SASI bus cable between the FMSD and the DDC (via a mass-terminated 50-pin connector). The SASI bus allows for the following:

- Processor-to-processor data transfers
- Arbitration for SASI bus control
- Host-to-controller communications
- ADMP emulates host for disk control (Future)
- ADMP passes data to/from FMSD.
- ADMP requests disk data transfers via processor-processor functions.

The SASI bus contains the following signals:

- DO-D7 = bidirectional data bus
  - transfer of eight-bit data between master and slave
  - addressing slave during selection
- RESET = reset slaves, not used for ADMP
- SEL = active indicates address of slave on data bus
- BUSY = acknowledgment of select
- REQ = slave ready to transfer or receive data
- ACK = handshake for byte transfers
- (C/D) = Command/Data signal from slave as to type of transfer (not used for FMSD/ADMP interface)
- MSG = message, indicates type of bus transfer (not used for FMSD/ADMP interface)

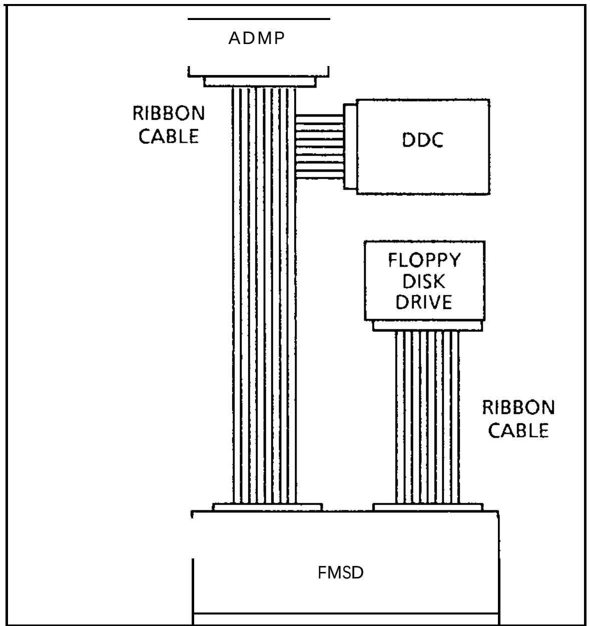


Figure 8.4 Disk System Cabling

Voice and Data Transfer

8.3 This paragraph describes the transfer of voice and data.

Complex Low -Level Bus

8.3.1 The VPLC, VPLC2, UCB, PR and ADMP-A PCBs communicate with the voice system processor over the PEC low-level bus. The low-level bus is separate from the PCM bus. The low-level bus communicates with and controls the cards that are installed in the PCMUSs (PCM Universal Slots). The processor reads from, and writes to, the cards on the bus by accessing a section of the address space that is mapped to the ports of the cards. The bus control logic translates the access into a card select signal, a port address, and a read/write signal on the bus. It then transmits a byte of data to, or receives a byte of data from, the designated card bus port. Cards such as the PR (Packet Router) and the VPLC (Voice Packet Line Card) are supplied with a register, and cards such as the UCB (Universal Controller Board) card and the ADMP (Administrative and Maintenance Processor) card are supplied with a first-in/first-out interface to the PEC low-level bus. This interface allows them to exchange information with the system.

NOTE: Only the VPLC2 and NIC data cards electrically connect to the PCM bus.

Voice Switch Pulse Code Modulation Network

8.3.2 The voice switch network provides the ability to connect any two (or more) voice ports (lines or trunks). Communication through the network is based upon time-division multiplexing of PCM-encoded samples. The voice signals convert from analog to PCM at the line/trunk card. However, in the Digital Featurephone, the voice signal is converted to PCM in the telephone and passed in the form of mini-packets to the VPLC (Voice Packet Line Card,) where it is de-packetized. A card select and port select signal is sent from the network control logic to the VPLC to gate a PCM sample onto the 24-channel file PCM bus. The file buses combine into a 96-channel bus that is fed into the 192-connection time-slot interchange switch. The time switch transfers the sample to the appropriate outgoing time slot, and the sample follows the reverse path to the terminating port.

Packet Transport System

8.4 The packet transport system hardware (Figure 8.5) switches mini-packets between the various devices connected to it. All control and user information exchanged within the data system uses the packet transport system. Mini-packets are self-routing, fixed-length entities, which contain addressing information (packet line addresses), user data, and error-detection information. Two versions of mini-packets are used. Internal mini-packets are those that are transmitted within the central switch itself. External mini-packets are those that are transmitted across a local access medium (such as a pair of wires) to/from a remote processor. The conversion between internal/external formats is performed in the voice packet line cards which provide access to the local access medium.

Mini-packets are assigned access to the switching and transmission hardware upon demand and are transported over the LPB.

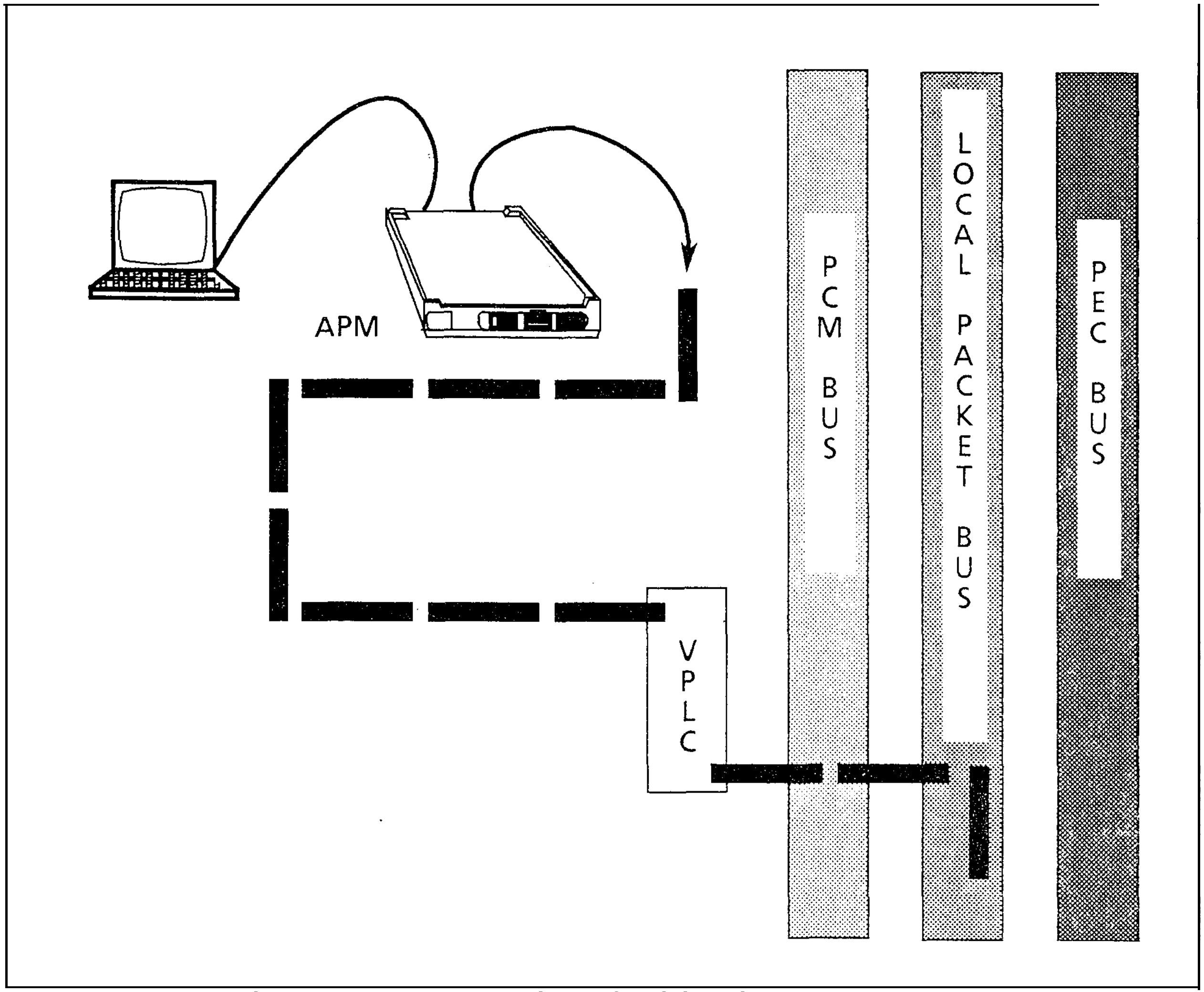


Figure 8.5 Transportation of Mini-Packets over LPB

Local Packet Bus

8.5 The LPB (Local Packet Bus) consists of eight leads and is essentially two separate buses. The first is a two-bit-wide data bus. The second, a two-bit-wide time-division-multiplexed bus, transfers packets and encodes busy status indications. Both buses operate off the same 1.544-MHz clock, and both use the same frame synchronization line to maintain the link.

Any hardware which connects to the LPB for communication is referred to as a packet line interface. Each packet line interface uses 1 of the 504 packet line addresses on the local packet bus. In addition to providing a full-duplex data transmission facility capable of receiving 12,000 incoming (toward the packet router enhancer) and sending 12,000 outgoing (from the packet router enhancer) mini-packets per second, the local packet bus provides for the real-time exchange of the status information of each packet line interface.

THIS PAGE IS INTENTIONALLY LEFT BLANK.

PD-200 BASIC OPERATING PRINCIPLES

9.0 Described in this paragraph are the basic operating principles of the packet transport system used to support the data system and the Digital Featurephone.

Call Path Concepts

9.1 Simultaneous independent paths (logical links) can be established between various devices within the data system. This lets a single card simultaneously communicate with several other cards in the system by multiplexing transmission across the packet transport system. For a description of logical links usage, see Table 9.1.

Table 9.1 Logical Links

END POINTS			LOGICAL LINK USAGE
Remote Processor	<-->	Remote Processor	Exchange of subscriber data
Remote Processor	<-->	Universal Control Board	Exchange of call control information
Remote Processor	<-->	Administrative and Maintenance Processor	Exchange of system control information
Universal Control Board	<-->	Administrative and Maintenance Processor	Exchange of system control information and call accounting information

For exchange of subscriber data, the UCB- and ADMP-resident software controls allocation of logical links. Since each remote processor has a unique packet line address, call routing for data connections can be defined as the relationship of an X.121 address to a packet line address. Two such packet line addresses (source and destination) are required to set up a call between remote processors. The data connection logical link between two remote processors is the path that user data mini-packets will use. Each data connection logical link supports one or more virtual circuits, or logical connections.

The data connection logical links carry the virtual circuits. These are distinct from the call control logical links between the UCB card and a remote processor. This call control logical link is the only one over which call setups, clears, and other controlling commands go. Data call control is based on a two-path concept. All calls are internally known (after setup) by their call identification. This call identification is dynamically mapped to a call control block on the controlling UCB card. A call control block contains all necessary source and destination mapping information about a specific data call.

PD-200 Data System

9.2 The PD-200 Data System functions as an independent data switch within the system. It provides interconnections among asynchronous DTE (Data Terminal Equipment), DCE (Data Communication Equipment), and packet mode X.25 DTEs. The data system also supports interfaces to X.25 PDNs (Public Data Networks). All data devices not collocated with the switch connect directly to interface circuitry that allows the device to transmit data over a single pair of wires. Asynchronous DTEs or DCEs interface APMs (Asynchronous Packet Managers) or DFP/APMs. The X.25 DTEs and X.25 PDNs interface SPMs (Synchronous Packet Managers).

Each APM and SPM that interfaces with the packet transport system communicates over a single-pair wire using mini-packet protocol. The wire pair terminates on a VPLC (Voice Packet Line Card) or VPLC2 which connects to an LPB (Local Packet Bus) within the backplane of the switch. High-speed hardware components within the packet transport system accept mini-packets from their source VPLC cards and switch them to their destination without additional processing. The end points run mini-packet protocol, not the switching hardware within the packet transport system.

Data call-processing software runs on the UCB card configured with data call-processing software. This card is responsible for all of the non-real-time critical functions, such as command interpretation, call setup, and call takedown. Except for command information, user data does not pass through the UCB card. It goes directly between the end points (APM-to-APM, APM-to-SPM, or SPM-to-SPM).

Administrative software resides on the ADMP cards. The ADMP has high bandwidth access to the system disk where event and data accounting information are kept. The ADMP contains software which provides a simulated X.25 end point. This allows access to the data administrative functions, such as Recent Change and maintenance. via virtual circuits. Any device connected to the PD-200 Data System can be used as a data maintenance console as long as enough security information is given.

Software oaded devices that have a LED (Light Emitting Diode) (UCB cards, ADMP cards, APM, and SPM) use the LED to signal hardware malfunction. A card that passes ROM self-tests flashes this LED at approximately 60 IPM. A card that fails self-tests flashes this LED at approximately 120 IPM. When these devices are loaded with the appropriate software, the LED is lit steadily.

The VPLC and VPLC2, which are not software-loaded cards, also have an LED. This LED is off until the card is initialized. Then the LED lights steadily as long as the card is operational.

Data Interfaces

9.3 The PD-200 Data System supports the following data devices:

- ASCII asynchronous data terminals at speeds up to 19.2 Kbps
- Host computers emulating asynchronous data terminals at speeds up to 19.2 Kbps
- A limited set of full-duplex modems (private line, auto-answer, and auto-dialer) providing an asynchronous, serial, ASCII, single-speed interface
- X.25 DTEs at speeds up to 64 Kbps with HDLC framing
- X.25 PDNs at speeds up to 64 Kbps with HDLC framing

Signaling

9.4 For asynchronous devices, user commands (call setup, call disconnect, etc.) are entered via the device keyboard by using a command language based on and similar to CCITT Recommendation X.28. The X.25 devices use X.25 control packets to provide signaling.

Asynchronous Call Origination

9.5 This paragraph presents a summary of a call setup as seen at an asynchronous terminal. It details the connection being established between a terminal and an X.25 host, both directly connected to the PABX:

- DTR (Data Terminal Ready) is raised by the terminal connected to the originating APM.
- A <carriage return ><period ><carriage return > is entered on the terminal keyboard. This informs the PD-200 Data System of the correct operating speed and parity for the terminal and that the terminal is to run full duplex. For a half-duplex application, <carriage return > < hyphen > < carriage return> should be used (this only applies to ports marked autobaud in the port table).
- The herald is received, and on autobaud devices the user is prompted for the “Terminal Type“.
- After receiving the command prompt, the user enters the connect command on the keyboard, followed by the X.121 address to which he wants to connect. Mnemonic addressing can also be used.
- An X.25 incoming call packet is sent to the X.25 host or asynchronous port, informing it of the incoming call. The host can accept or reject the call.
- If the connection cannot be established, or if the X.25 end point rejects the call, a message indicating the reason (error, busy, etc.) is displayed on the originating terminal. (These messages are suppressed if X.3 parameter 6 is set to a zero.)

- A connected message is displayed on the originating terminal when the connection has been made. (This message is suppressed if X.3 parameter 6 is set to a zero.)

- The call indicator on the originating APM will be lit.

Call Disconnect

9.6 Either party can disconnect a data call. Asynchronous devices disconnect a data call by dropping DTR/DSR, or by escaping to the network command mode and entering the disconnect command on the keyboard. (This option is not allowed if X.3 parameter 1 is set to zero.) The X.25 devices disconnect a data call either by issuing a call clear packet, or by dropping data set signals or restarting the X.25 line. This will cause all calls on this line to be disconnected and is considered an abnormal condition. Either way, the call indicator on an APM will be extinguished when a connection is terminated.

Protocol Conversion

9.7 The data system provides an asynchronous to X.25 protocol conversion for each asynchronous port. Recommendation X.25 allows a data device to transmit data as a series of messages or packets, and to simultaneously support multiple virtual circuits to a number of different devices. The following description assumes that an X.25 host connected to an SPM is communicating with a terminal connected to an APM.

When the terminal user enters data from the terminal, data is transmitted to an APM asynchronously, byte by byte, where it is converted into X.25 data packets. This is done by collecting the asynchronous characters to form X.25 data packets to send to the VPLC. Special software in the APM, referred to as PAD (Packet Assembler/Disassembler) software, converts the asynchronous data to X.25.

Once the X.25 packet (X.25 level 3) is created at the APM, it is broken down by the APM into a series of mini-packet protocol packets. These are transmitted, via the wire pair, to the VPLC. The VPLC places the mini-packets onto the LPB; then they are switched by the packet transport system hardware to their destination. Mini-packet protocol is used as the link level (level 2) protocol to ensure that the packet is delivered without errors or duplication to its destination. At the SPM, the mini-packet protocol packets are collected to reconstruct the original X.25 data packet that was formed at the APM. Next the SPM envelops the X.25 level 3 packet information in level 2 frames, using HDLC framing. Since the X.25 host computer directly accepts X.25 data packets, the PAD function is not required here.

Asynchronous interfaces are supported for speeds up to 19.2 Kbps with a standard EIA RS-232-C connector, whereas synchronous interfaces are supported at speeds up to 64 kbps operating with X.25 protocol and connected via EIA RS-232-C (for up to 19.2 Kbps links) or V.35 (for higher-speed links).

Data Device Addressing

9.8 This paragraph describes addressing for data devices.

Numbering Plan

9.8.1 To plan a data numbering scheme, the system users must first be grouped in the following categories.

- A rotary hunt group (89 users maximum) - asynchronous
- 89 individual users (maximum) of the non-rotary hunt group - asynchronous
- 1 pair of primary/secondary X.25 lines - synchronous
- 1 pair of load-sharing X.25 lines - synchronous
- 1 single X.25 line - synchronous
- 1 ADMP

Each group is identified by a unique address (server number). The addresses are used to complete the data base sheets for Record Code RT.

The addressing used by the data system is the X.121 addressing scheme.

CCITT X.1 21 Addressing

9.8.2 The CCITT has defined an international numbering plan for public data networks known as Recommendation X.121 (Figure 9.1). Under this plan, all network addresses are composed of a 14-digit number. The first four-digit block is called a DNIC (Data Network Identification Code). One or more DNIC codes are assigned to each country by the CCITT. The remaining 10 digits are called a Network Terminal Number; this is subdivided into an 8-digit server number and a 2-digit sub-port number.

The synchronous (X.25 DTEs) device is normally assigned a 12-digit X.1 21 address, while an asynchronous device is assigned a 14-digit X.1 21 address. The format of the complete X.121 address is DNIC SSSSSSSS PP (Figure 9.1).

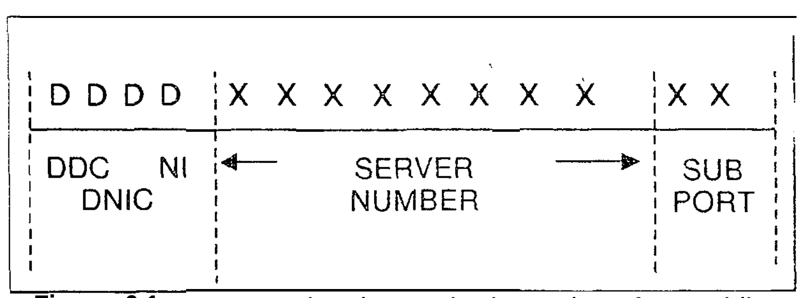


Figure 9.1 International Numbering Plan for Public **Data** Network Addressing

Data Network Identification Codes

9.8.3 In the United States, the DNICs are assigned to a network NI = Network Identifier digit by the Federal Communications Commission. The GTE-Telenet Public Network has been assigned DNIC 3110 (see Figure 9.2). Some currently assigned DNIC codes are as follows:

DNIC	NETWORK	COUNTRY
3110	Telenet	USA
3106	Tymnet	USA
2341	IPSS	England
2080	NTI	France
2081	Transpac	France

In the PD-200 Data System, it is suggested that DNIC 3110 be used in the X.121 address. For ports which connect to other public data networks, the user must obtain the appropriate X.121 address from the public data network vendor.

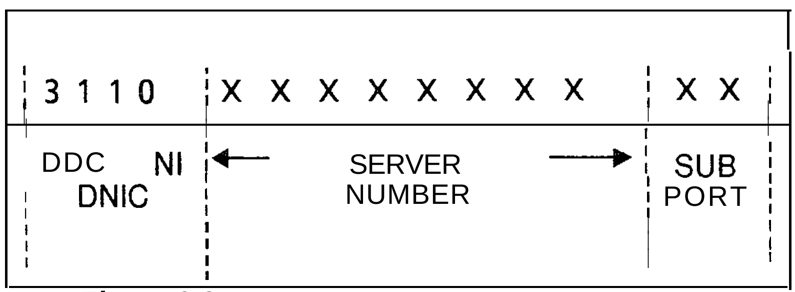


Figure 9.2 DNIC for Telenet

NOTE: The data system cannot be configured without an X.121 address. This is supplied by a PDN (Public Data Network). Telenet is recommended for usage by the PD-200 Data System.

Server Number

9.8.4 The server number (Figure 9.3) is an eight-digit number which identifies an X.25 host, terminal, or network. A server number may also identify a collection of asynchronous lines/devices that are grouped into a set. The lines/devices in this set can either be accessed individually or as a group. A main pilot number provides access to a rotary hunt group. One set may contain a maximum of 89 terminals in the PD-200 Data System.

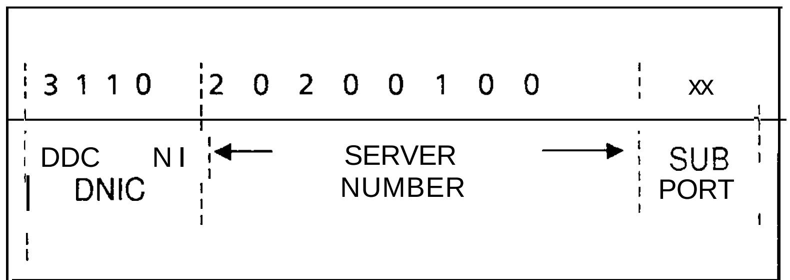


Figure 9.3 Example of DNIC Server Number

NOTE: This example represents a user served by Telenet in Washington, D.C., which has only one area code (202). Telenet uses the area code as the first three digits of the server number

Sub-port Number

9.8.5 This is a two-digit number that is used to identify a specific asynchronous terminal with the set (Figure 9.4). The sub-port is not used with X.25 synchronous lines/devices. The sub-port number may range in value from 01 to 89 for each server number.

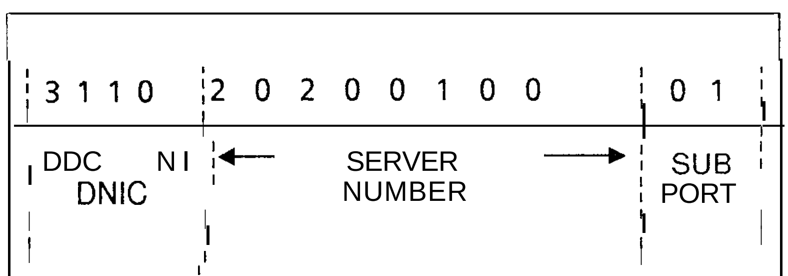


Figure 9.4 Example of DNIC Server Number and Sub-Port

Users of asynchronous terminals have the option of specifying complete or partial X.1 21 addresses when setting up a virtual circuit. If the address specified begins with a zero, then a complete address will be assumed. The address used will be the same as that which the user entered, with the leading zero removed. If the address does not begin with a zero, a partial address will be assumed. In this case, the called X.121 address is calculated by placing the user-entered number into the server field right justified and padded to the left with zeros. The DNIC field will be the same as the DNIC associated with the originating terminal. If a port number has been specified, then it will be used; otherwise, the port number will be set to zero. Sub-addresses 90-99 are reserved for internal administrative functions. The ADMP uses an address which must have a unique server number and its support number must be 00. No other restrictions are imposed on the number of digits or the values of the digits comprising the address.

Routing

9.9 All routing of X.25 data packets is based upon the virtual circuit concept. When a data call request packet is received (or created) by the data switching software, a path will be established (if possible) between the data port from which the call request was received and another port whose identity is determined from a routing table. All subsequent data packets associated with the virtual circuit will follow the same path between the two ports. This path will be maintained by the data switching software until the virtual call is cleared.

Routing decisions are based upon a table of partial X.121 addresses. This table holds up to 127 entries. Each entry contains a partial X.121 address made up of actual binary-coded digits, wild-card digits, and an indication of how many significant digits of that address are to be used in routing decisions. Thus, many destination addresses may be represented by a single routing table entry. This entry can contain one or more wild cards, or a single specific address may be represented in the table. A single address needs all 14 digits specified where a wild card does not. The routing tables also contain a primary port identification, primary port controlling DCP, an optional secondary port identification, a secondary port controlling DCP, and an indicator of how the secondary port is to be used in conjunction with the primary port (load shared based on number of virtual circuits, or when the primary port is busy or out of service).

NOTE: The ADMP must have its own routing table entry.

Rotary Hunt Group

9.9.1 A group of lines/devices that share the same server number are called a rotary hunt group (Figure 9.5). The following describes the operation of a rotary hunt group: When a call is placed to the pilot number of a rotary hunt group, the system will automatically scan through the defined set of ports in the group looking for the first available non-busy line. Upon finding one, the system will make the connection to that specific port and inform the remote user of its choice.

Since only 1 server number is needed for each group of 89 users, this can be used when interfacing multiple lines into an asynchronous host, group of printers, or group of modems. The PD-200 Data System automatically assigns ports upon demand. Also, even though the ports have been defined as members of a rotary group, a specific port can still be called by using the port number. If the specific port is busy, the system informs the user. The system does not select an alternate address (Figure 9.6).

DDC NI DNIC	SERVER NUMBER	SUB PORT
3 1 1 0	2 0 2 0 0 1 0 0	0 1
3 1 1 0	2 0 2 0 0 1 0 0	0 2
3 1 1 0	2 0 2 0 0 1 0 0	0 3

Figure 9.5 Rotary Hunt Group Example

NOTE: Example of a rotary hunt group with 3 members (89 maximum).

DDC NI DNIC	SERVER NUMBER	SUB PORT
3 1 1 0	2 0 2 0 0 1 0 0	0 1

Figure 9.6 Example of Rotary Hunt Group Numbering Plan

NOTE: This example represents a rotary hunt group access to a group of printers. Any DFP/APM can access the group by dialing the X.121 address (DNIC and server number, 311020200100) however, if a certain printer is needed, the sub-port would have to be dialed as well (example 31102020010001).

Load Sharing

9.9.2 Load sharing on the data system when used in conjunction with the SPMs and the NIC PCBs provides equal distribution of calls over X.25 paths to a set of destinations (Figure 9.7 and 9.8).

DDC NI DNIC	SERVER NUMBER	SUB PORT
3 1 1 0	2 0 2 0 0 1 0 0	

Figure 9.7 Example of Load Sharing Numbering Plan

NOTE: In a load sharing application, this X.121 address would be shared by two synchronous users.

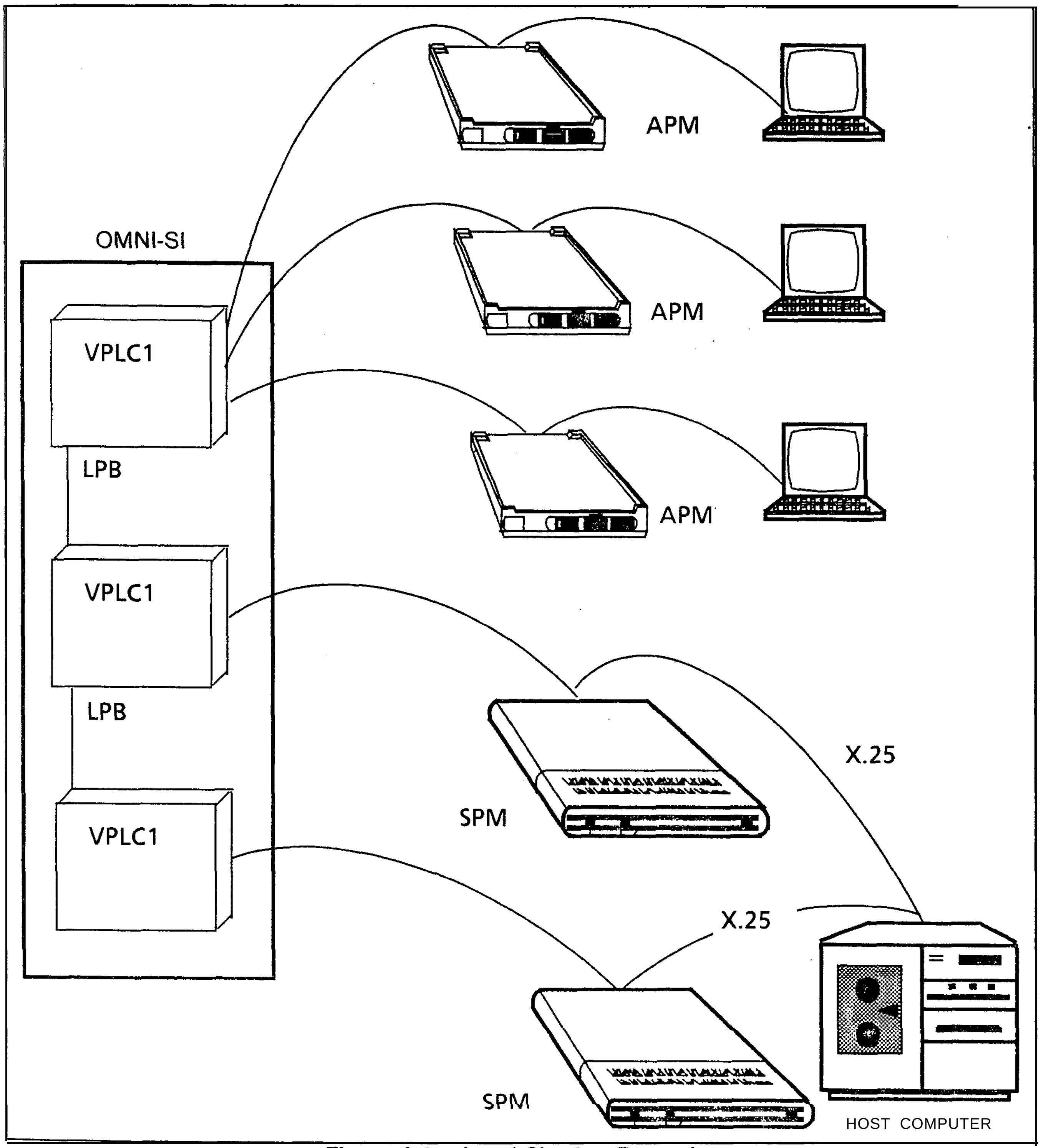


Figure 9.8 Load Sharing Example

NOTE: The APMs are able to access the host computer via the SPMs. This application provides an automatic back-up connection should one of the links fail. It is recommended to put load sharing SPMs on different VPLCs.

List of Cards

9.10 Tables 9.2 and 9.3 provide an alphabetical and a numerical list of the cards used in the OMNI SI PABX for both voice and data.

NOTE: The FB mnemonic, which is placed in front of each card number, stands for functional board.

The following list, provided for your reference, gives of all the cards in a OMNI SI PABX. Figure 9.8 shows the system data connections.

Table 9.2 Alphabetical List of Printed Circuit Boards

MNEMONIC	CARD NUMBER	DESCRIPTION
ADMP-A	FB-17229-A	Administrative and Control Processor (Note 3)
ADMP-C	FB-17230-B0A	Administrative and Control Processor (Note 3)
ATT12	FB-17208-A	Attendant Interface Number 2 (Note 2)
BCSR	FB-17204-A	Battery Charger 5 Volt Regulator (Note 1)
CHM85	FB-17218-A	Channel Memory 8085 (Notes 1 and 3)
CIP	FB-17225-A	Control Interface to Periphery (Note 2)
CP85E	FB-17288-A	Central Processor Unit (Enhanced) (8085) (Note 1)
DVCIP	FB-17236-A	Data Voice Control Interface Processor (Notes 1 & 2)
FDC	FB-15278-A	Frame Detector Circuit for T1
FMSD	FB-17220-B0A	File Management System Data Card (Notes 1, 2, & 3)
INCKS	FB-20922-A	Synchronizable Intermediate Network Clock
INCK	FB-20771-1A	Intermediate Network Clock
LCM	FB-15280-A	Line Compensator for T1
MPB85	FB-17215-A	Multi-Processor Buffer 8085 (Notes 1 and 3)
MPG16	FB-17213-B0A	Memory Paging 16 Page (Note 1)
M1MB	FB-17314-1A	Memory 1 Megabyte (Note 1)
NIC	FB-17242-A	Network Interface Card
NSDC	FB-20992-A	Narrow Serial Device Controller (Notes 1 and 3)
OAIOD	FB-17276-A	Omni Automatic Identification of Outward Dialing
OCA	FB-17265-A	Outrigger Cable Adapter
PADIC	FB-17210-A	Public Address and Dictation
PBE/T	FB-17227-A	Packet Bus Extender/Terminator (Note 3)
PCMI	FB-17187-A	PCM Interface
PCMFS	FB-17189-A or B0A	PCM Frame Synchronization
PCMTS	FB-20974-A	PCM Tone Source Card (Note 1)
PCONF	FB-51279-A	PCM Conference Card
PCOT	FB-17202-A or BOA	PCM Central Office Trunk (Note 2)
PDTMF	FB-17203-A	PCM Dual Tone Multi-frequency
PEMT	FB-17201 -A	PCM Two-Wire E&M Trunk (Note 2)
PFT	FB-51051 -A	Power Fail Transfer Card

Table 9.2 Alphabetical List of Printed Circuit Boards (Continued)

MNEMONIC	CARD NUMBER	DESCRIPTION
PFWTA	FB-51267-A	PCM Four-Wire E&M Trunk (Note 2)
PLCC	FB-17254-A or 1A	PCM Line Circuit Card (Note 2)
PMI	TR-100119-1	Property Management System Interface Card (Note 1)
POPS	FB-17250-A	PCM Off-Premises Station Line Card (Note 2)
PRE	FB-17228-B0A	Packet Router/Extender (Note 3)
PRLT	FB-17251-A	PCM Release Link Trunk (Note 2)
PSUPY	FB-17197-A	Power Supervisory (Note 1)
RABR	FB-20996-A	Recorder Announcer Buildout Resistor
RPTR	FB-17312-A	Repeater Card (Note 3)
SIDML	FB-17209-A	SI Dual Modem and Current Loop (Note 2)
SIL	FB-17277-A	Span Interface and Output Format for T1
SIL	FB-15277-1A	Span Interface and Output Format for T1
EPCMN	FB-17217-A	Network (Notes 1 and 3)
TC	FB-17280-A	(Tone Control) PCM Progress Tone Recognizer
TPI2	FB-17188-A	Test Panel Interface Version 2 (Notes 1 and 3)
T1B2	FB-17192-A	T1 Buffer Circuit 2
T1S	FB-20718-I A	Supervisory Alarm Circuit for T1
UCB/DCP	FB-17231 -A	Universal Controller Board
VCIP	FB-17235-A	Voice Control Interface Processor
PILT	FB-51280-A or BOA	PCM Incoming Loop Trunk DID (Note 2)
VPLC	FB-17226-A	Voice Packet Line Card
VPLCD	FB-17226-I A	Voice Packet Line Card Derived
VPLC2	FB-17246-A	Voice Packet Line Card 2

NOTES:

1. These cards cannot be removed or replaced without removing power from the cabinet in which the card is located. (Refer to power-down procedures.)
2. These cards cannot be removed without first placing the card position in the maintenance busy state. (Refer to power-down procedures.)
3. Disconnect the cables from the front of these cards before removing or replacing them.

Table 9.3 Numerical List of Printed Circuit Boards

CARD NUMBER	MNEMONIC	DESCRIPTION
FB-15277-I A	SIL	Span Interface and Output Format for T1
FB-15278-A	FDC	Frame Detector Circuit for T1
FB-15280-A	LCM	Line Compensator for T1
FB-17187-A	PCMI	PCM Interface
FB-17188-A	TPI2	Test Panel Interface Version 2 (Notes 1 and 3)
FB-17189-A or BOA	PCMFS	PCM Frame Synchronization
FB-17192-A	T1B2	T1 Buffer Circuit 2
FB-17201 -A	PEMT	PCM Two-Wire E&M Trunk (Note 2)
FB-17202-A or BOA	PCOT	PCM Central Office Trunk (Note 2)
FB-17203-A	PDTMF	PCM Dual Tone Multi-frequency
FB-17208-A	ATTI2	Attendant Interface Number 2 (Note 2)
FB-17209-A	SIDML	SI Dual Modem and Current Loop (Note 2)
FB-1721 O-A	PADIC	Public Address and Dictation
FB-17213-B0A	MPG16	Memory Paging 16 Page (Note 1
FB-17215-A	MPB85	Multi-Processor Buffer 8085 (Notes 1 and 3)
FB-17217-A	EPCMN	Network (Notes 1 and 3)
FB-17218-A	CHM85	Channel Memory 8085 (Notes 1 and 3)
FB-17220-BOA	FMSD	File Management System Data Card (Notes 1, 2, & 3)
FB-17225-A	CIP	Control Interface to Periphery (Note 2)
FB-17226-A	VPLC	Voice Packet Line Card
FB-17226-1A	VPLCD	Voice Packet Line Card Derived
FB-17227-A	PBE/T	Packet Bus Extender/Terminator (Note 3)
FB-17228-B0A	PR	Packet Router (Note 3)
FB-17229-A	ADM P-A	Administrative and Maintenance Processor (Note 3)
FB-17230-B0A	ADM P-C	Administrative and Maintenance Processor (Note 3)
FB-17231 -A	UCB	Universal Controller Board
FB-17235-A	VCIP	Voice Control Interface Processor (Notes 1 and 2)
FB-17236-A	DVCIP	Data Voice Control Interface Processor (Notes 1 & 2)
FB-17242-A	NIC	Network Interface Card
FB-17246-A	VPLC2	Voice Packet Line Card 2

Table 9.3 Numerical List of Printed Circuit Boards (Continued)

CARD NUMBER	MNEMONIC	DESCRIPTION
FB-17250-A	POPS	PCM Off-Premises Station Line Card (Note 2)
FB-17251-A	PRLT	PCM Release Link Trunk (Note 2)
FB-17254-A or 1A	PLCC	PCM Line Circuit Card (Note 2)
FB-17265-A	OCA	Outrigger Cable Adapter
FB-17276-A	AIOD	Automatic Identification of Outward Dialing
FB-17277-A	SIL	Span Interface and Output Format for T1
FB-17280-A	TC	(Tone Control) PCM Progress Tone Recognizer
FB-17288-A	CP85E	Central Processor Unit (Enhanced) (8085) (Note 1)
FB-17314-A	M1MB	Memory 1 Megabyte (Note 1)
FB-17314-1A	M1MB	Memory 1 Megabyte (Note 1)
FB-20718-1A	T1S	Supervisory Alarm Circuit for T1
FB-20974-A	PCMTS	PCM Tone Source Card (Note 1)
FB-20992-A	NSDC	Narrow Serial Device Controller (Notes 1 and 3)
FB-20996-A	RABR	Recorder Announcer Buildout Resistor
FB-51051-A	PFT	Power Fail Transfer Card
FB-51267-A	PFWTA	PCM Four-Wire E&M Trunk (Note 2)
FB-51279-A	PCONF	PCM Conference Card
FB-51280-A or B0A	PILT	PCM Incoming Loop Trunk DID (Note 2)
TR-100119-1	PMI	Property Management System Interface Card (Note 1)

NOTES:

1. These cards cannot be removed or replaced without removing power from the cabinet in which the card is located. (Refer to power-down procedures.)
2. These cards cannot be removed without first placing the card position in the maintenance busy state. (Refer to power-down procedures.)
3. Disconnect the cables from the front of these cards before removing or replacing them.

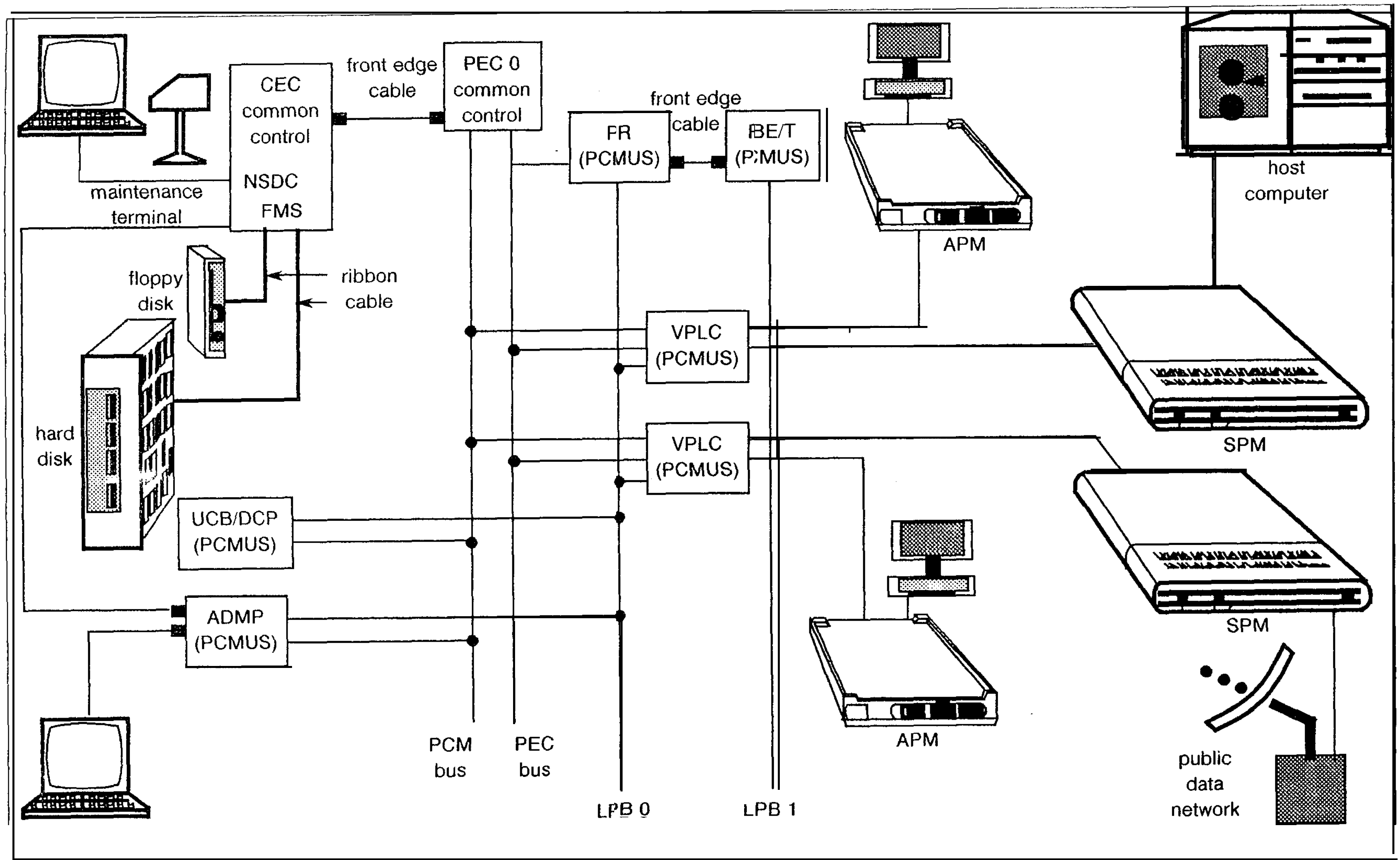


Figure 9.9 Data Connections

VOICE EQUIPMENT/TRAFFIC ENGINEERING

10.0 The system supports a maximum of 256 lines or 64 trunks. If optional features are used, this number is reduced. This section discusses:

- The impact of line traffic on the system
- The impact of trunk traffic on the system
- The impact of features on the system

Voice Transportation Inside the PABX

10.1 Time slots transport analog (voice and tone) signals inside the digital environment of the switch (Figure 10.1).

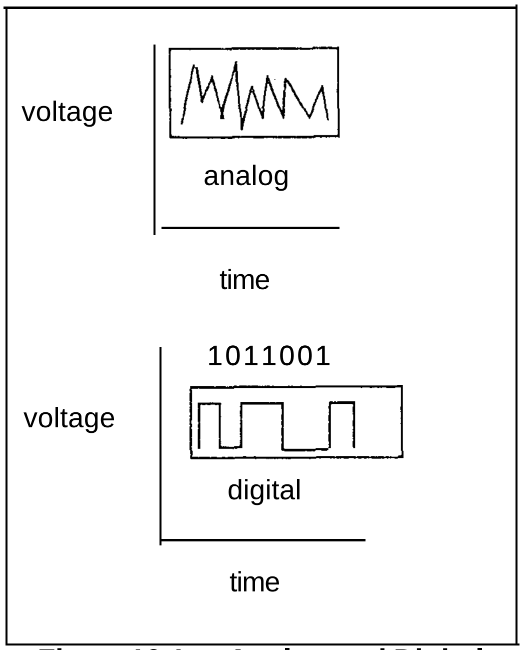


Figure 10.1 Analog and Digital Signals

Each call requires a time slot. Lines, trunks, and certain features share time slots.

NOTE: Cards that transport voice or tone use time slots, as well as NIC and DVCIP cards that use time slots for data transport. Data link cards such as the ATTI2, CIP, and SIDML do not use time slots.

Time slots are assigned on demand by use of time-division multiplexing techniques. These techniques let different users access the same physical channel for short periods of time, thus accommodating all of the allowed users.

The PCM (Pulse Code Modulation) low-level bus allocates use of the time slots (Figure 10.2).

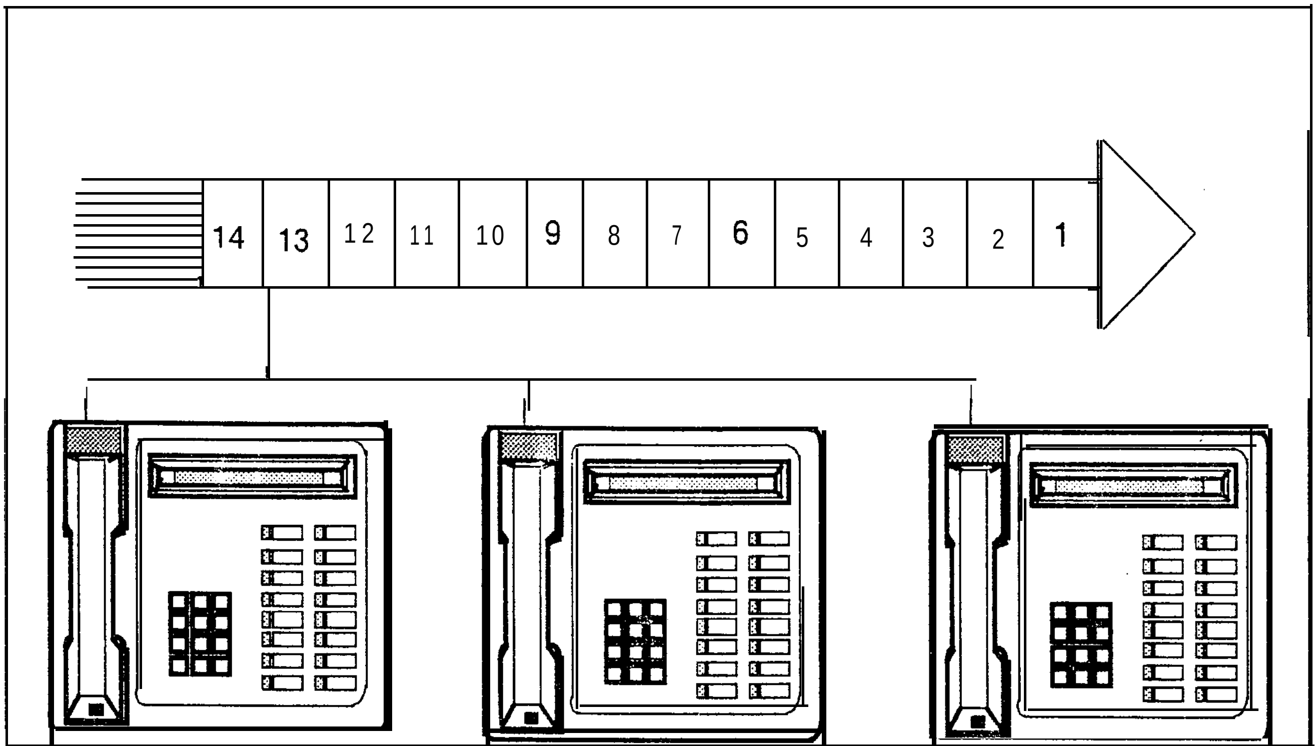


Figure 10.2 PCM (Pulse Code Modulation) Low-Level Bus

The use of time slots permits the system to coordinate the placement of a phone call:

- When a call originates, the system assigns a series of memory locations to the originator within 10 milliseconds after the phone is taken off-hook. This is called assigning a time slot.

- The time slot permits access to the signaling tones and timing sequences needed to complete a connection. The access allowed depends upon the signaling mode used when dialing, and on the feature used.

- If a connection is completed, the system assigns an additional set of memory locations (time slots) to carry return information to the originator. (Two time slots are needed for a two-way conversation.)

Time Switching Groups

10.2 Each group within the system has access to certain time slots (Figure 10.3).

- System hardware is divided into six time switching groups:
  - Each group has twenty-four time slots.
  - The Get Started File has two time switching groups that correspond with groups A and S.
  - The Expansion File has four time switching groups - two are located in group C and two are located in group D.
- The Get Started File has a total of 48 time slots.
- The Expansion File has a total of 96 time slots.
- The system has a total of 444 time slots. Each party in a conversation needs a time slot. The system can handle 72 simultaneous conversations if no features requiring dedicated time slots are used.

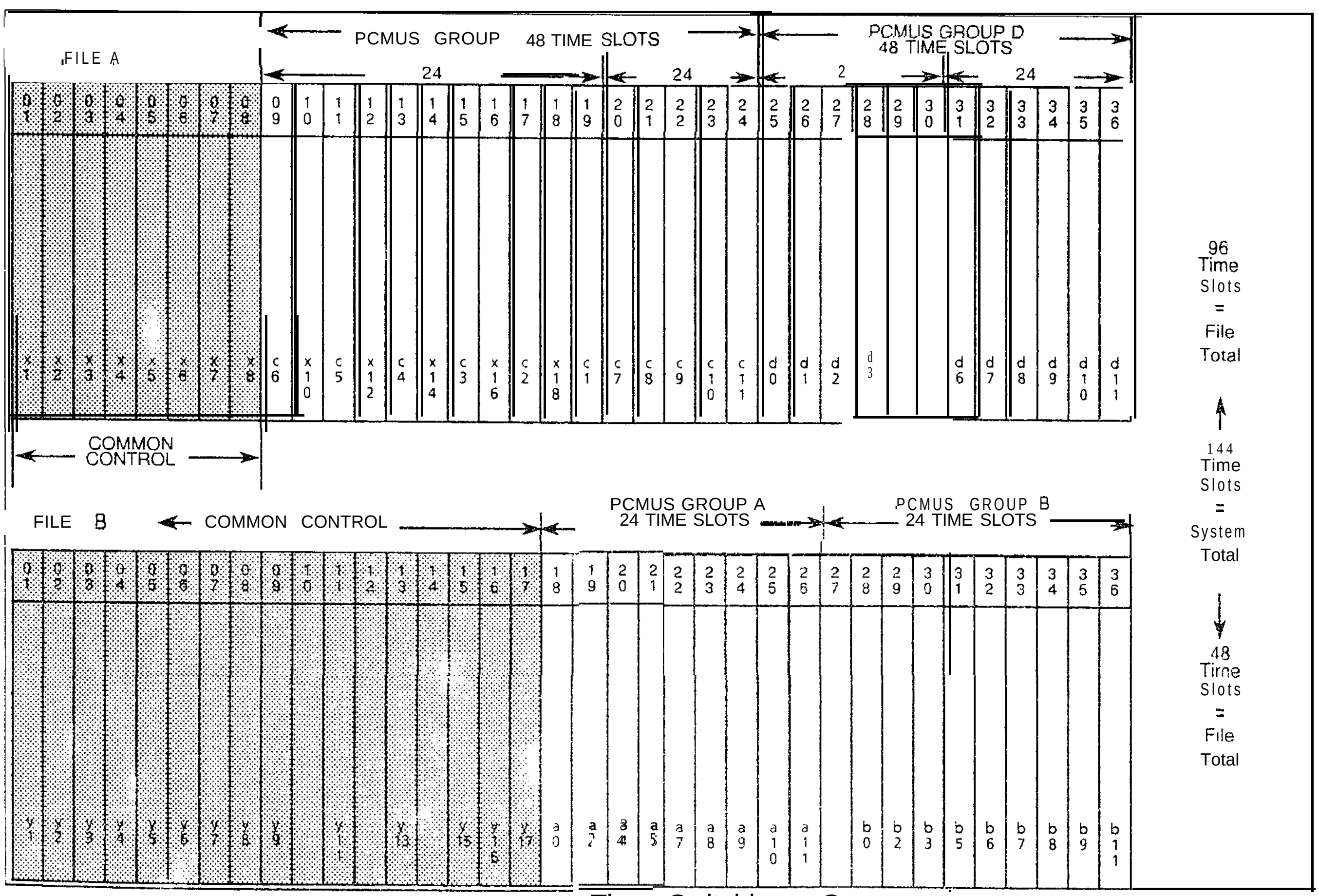


Figure 10.3 Time Switching Groups

Group Usage of Time Switching

10.2.1 The OMNI SI software makes extensive use of the time switching group division. One such use permits the system to coordinate the placement of a phone call:

- When a call originates, the system assigns a series of memory locations to the originator within 10 milliseconds after the phone is taken off-hook. This is referred to as assigning a time slot.

- Depending upon the signaling mode used when dialing, and on the feature invoked by the phone user, the time slot permits access of the signaling tones and timing sequences needed to complete a connection.

- If a connection is completed, the system assigns an additional set of memory locations (time slots) to carry return information to the originator.

Blockage of **Time Switching**

10.3 The line and trunk cards physically located in a time switching group have access to the time slots associated with that group only.

- If all the time slots in a group are busy, and a line or trunk circuit attempts to access a time slot, the attempt will be blocked.

- If a phone user attempts a call when all the time slots associated with the group are busy, the dial tone will be delayed until a time slot becomes available.

Types of Blockage

10.3.1 An originating call can encounter two types of traffic blocking congestion:

- Networking Blocking

  - Due to internal blocking of the network, the system is unable to send dial tone to the originating party at the time slot group level of the originator

  - ue to blocking of the network. the system is unable to onnect ring or voice to the called party at the time slot group level of the called party.

  This type of blocking happens when the number of concurrent originations and/or term/nations is greater than the number of channels (time slots) available in the PCMUS group.

SVR 5210

Blockage Occurrence

10.3.2 Blockage can only occur when the number of voice traffic line and trunk circuits in a PCMUS group exceeds the number of available time slots. Since it is unlikely that all the time slots in a PCMUS group will be busy simultaneously for any length of time, blockages rarely happen. To further reduce the possibility of blockage, a simple traffic study (traffic engineering) should be done. This will prevent putting too much hardware into a group. Traffic engineering is explained in the next paragraph.

Traffic Engineering

10.3.3 Traffic engineering compares the amount of hardware (line and trunk) used against the amount of time slots available to determine if blocking will occur. Traffic engineering should be done before installation and if any additions are made to the system.

Determining Blockage Probability

10.3.4 If actual traffic data is not available to calculate the probability of blockage for a PCMUS group (the P level). it will be necessary to make some assumptions about the line and trunk usage for the group. A catalog department would create a heavy usage for the associated lines and trunks, while a group of mid-level executives are likely to be a light load in comparison. These estimates permit the installer to make a preliminary card layout. Later line and trunk usage can be empirically measured to fine tune the card layout.

The standard measure of traffic in the United States is the CCS (Hundred Call Seconds). One call which lasts for 100 seconds constitutes one CCS, 60 seconds x 60 minutes = 3600 seconds in one hour = 36 hundred call seconds in one hour. Example - 18 CCS = 30 minutes; 9 CCS = 15 minutes; 6 CCS = 10 minutes. Based on experience, a number of CCS can be assumed for a line or trunk according to its expected usage. Table 10.1 summarizes these levels.

Table 10.1 Estimated Traffic Equivalents

USAGE LEVEL	LINE CCS	TRUNK ccs
Very High	9 & up	22 & up
High	6 to 9	20 to 22
Medium	3 to 6	18 to 20
Low	Less than 3	Less than 19

Determining CCS Usage of a System

10.3.5 (Figure 10.4) To calculate the estimated usage of a PCMUS group:

1. Multiply the number of lines by the letermined CCS value.

2. Multiply the number of trunks by the determined CCS value.

NOTE: Because DID and DOD trunks are one way, they can be considered low traffic.

3. Add these totals together.

(L X CCS/L) + (T X CCS/T) = CCS/group

L = total lines in the group

T = total trunks in the group

NOTE: Calculate all circuits on **a** line or trunk card, whether in service or not. In case of expansion, this prevents having to reconfigure the system.

Because the offered CCS for each time slot group is 529, the total CCS output for the hardware located in each group cannot go over 529 CCS.

(L X CCS/L) + (T X CCS/T) = or < 529

Example: Determine traffic volume for group A.

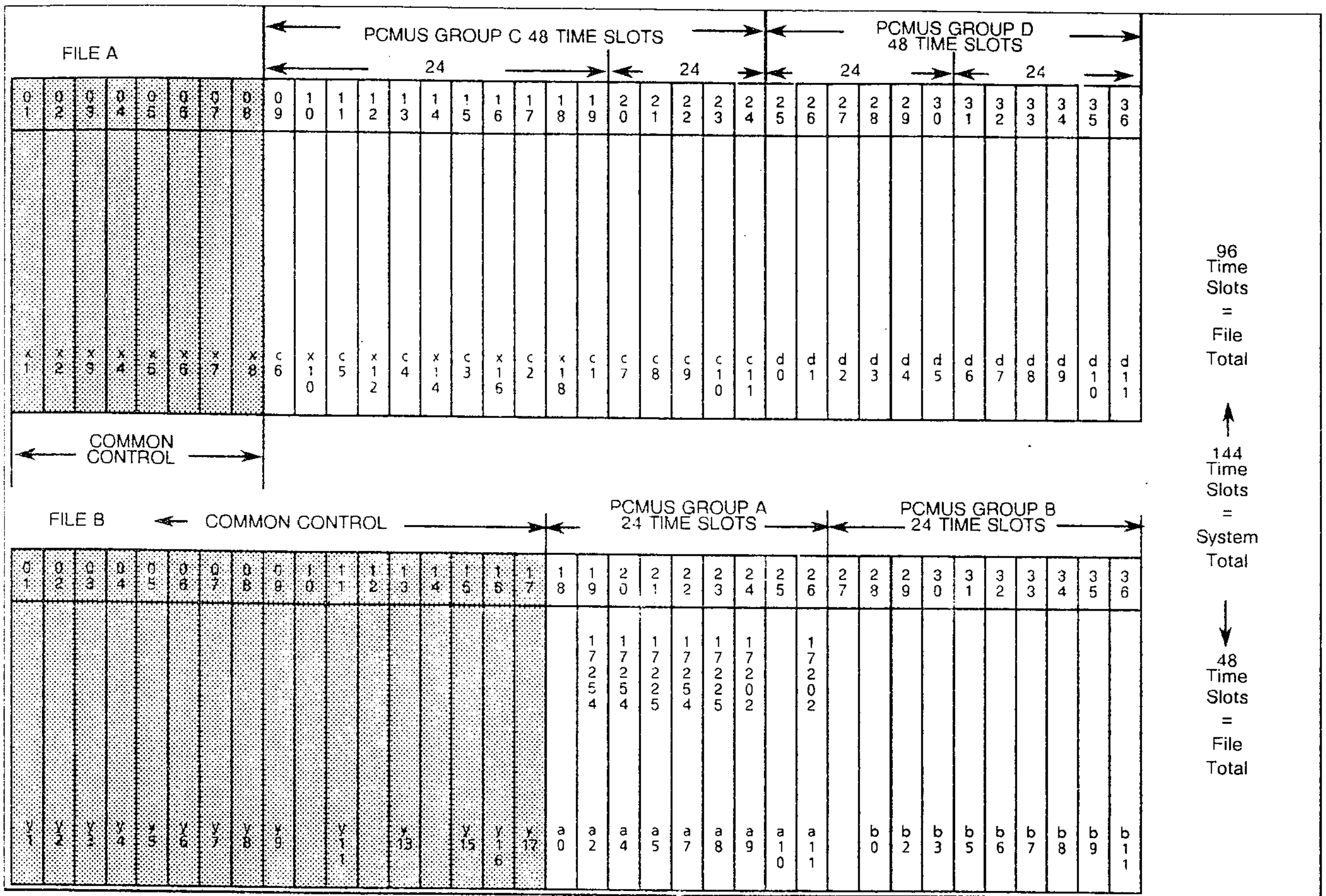


Figure 10.4 Traffic Study 1

1. Determine if the traffic of this system is high, medium, or low.
2. See Table 10.1 and use the CCS count that corresponds to the traffic volume.

NOTE: This example uses 6 CCS per line and 18 CCS per trunk.

3. There are 3 line cards (17254) x 8 circuits per card = 24 lines in group A.
4. There are 2 trunk cards (17202) x 4 circuits per card = 8 trunks in group A.

NOTE: The CIP (17225) data link cards do not use time slots.

5. (24 x 6) + (8 x 18) = 288/CCS for group A.
6. 288 is less than the 529 offered CCS, so group A has less than a P.01 grade of service.

NOTE: The CCS numbers (529) offered to a group are reduced when dedicated time slots are required by certain features.

Dedicated Time Slots

10.4 Certain equipment must have a time slot reserved for its use at all times (this is called a dedicated time slot). Each dedicated time slot is effectively removed from the pool of time slots associated with a PCMUS group.

The Attendant Console requires a dedicated time slot because the attendant must be accessible at all times. Dedicated time slots affect the traffic handling capacity of a PCMUS group. Table 10.2 lists the equipment that uses dedicated time slots.

Each Attendant Console requires a PLCC line circuit for voice connection. The line circuit uses a dedicated time slot. The music-on-hold feature also requires a PLCC line circuit which uses a dedicated time slot.

Off-hook queuing takes a time slot. Do not take this into account when doing a traffic study; however, it should be considered when setting data base parameters - as too many calls in a queue can affect the traffic of the system.

Table 10.2 Equipment Utilizing Dedicated Time Slots in a PCMUS Group

IN-SERVICE EQUIPMENT	NUMBER OF DEDICATED TIME SLOTS PER CIRCUIT
DMTF Receiver Circuits	1
Attendant Line Circuits	1
Music-on-Hold Line Circuits	1
Release Line Trunk Circuits	1
Recorder Announcer Trunk Circuits	1
Guaranteed Access Trunk Circuits	1
Nailed Trunk Circuits (both)	2
Progress Tone Circuits	1
TOTAL DEDICATED TIME SLOTS	______
NOTE: A NIC card is one end of a nailed trunk connection	

NOTE: Table 10.2 can be used to add the total number of dedicated time slots in a PCMUS group.

Progress Tone Recognizer

10.4.1 Access to a PPTR (FB-17280-A) card circuit is required for stations allowed to access an SCC through MERS (Most Economical Route Selection).

The system supports a maximum of two cards with four circuits per card. Each circuit of the PPTR requires a dedicated time slot. The number of stations using the feature determines the number of circuits used (Table 10.3).

Table 10.3 Required PCM Progress Tone Recognizer PCBs

Number of PCM Progress Tone Recognizer Circuits Required	Number of FB-4 7280-A PCBs Required	Maximum Number of Stations Allowed MERS SCC Access
2	1	17
3	1	56
4	1	111
5	2	176
6	2	250

NOTES:

- This table is based on a nominal holding time of 8 seconds. It is recommended that all PPTR PCBs be spread evenly throughout the system.
- The previous table is for reference only; for specific applications, a traffic study should be performed.
- Each PPTR circuit enabled utilizes a dedicated time slot.

DTMF (Dual Tone Multi-frequency)

10.4.2 The following require access to a DTMF circuit:

- All telephones except the Integrated Featurephone (analog, digital, and rotary)
  - The DTMF card converts analog signals into digital signals.
  - Analog to digital conversion for an Analog Featurephone is done on the PLCC. For the Digital Featurephone, this is done at the phone.
- Attendant Console
  - The Attendant Consoles in the system share a dedicated circuit on one of the PDTMF receivers; the rest of the system must share the remaining circuits.

- CAS Branch system with DTMF stations
  - DTMF receivers should be provided for each E&M Tie trunk (FB-17201 -A or FB-51267-A)
  - Each Tie line (E&M circuit) will be counted as two more DTMF stations
  - Each Tie line (E&M trunk circuit) that sends PDTMF signals to the PABX will be counted as two more PDTMF stations

To determine additional PDTMFs required for release link trunks in the CAS Branch option, see Table 10.4.

Table 10.4 Required PRLT Circuits

PRLT Circuits	PRLT PCBs	PDTMF Receivers	Required PCBS
1	1	1	1
2	1	2	1
3	1	3	1
4	1	4	1
5	2	5	2
6	2	5	2
7	2	6	2
8	2	6	2

NOTE: The system has a limit of 8 DTMF receiver circuits.

The system maximum for DTMF cards is two with four circuits on each card. The number of circuits required to support the system can be determined in one of two ways:

- By the number of stations requiring access to DTMF circuits
- By the traffic in CCS for lines and trunks accessing the DTMF receivers.

Determining DTMF Circuits by Number of Stations

10.4.3 Table 10.5 provides the number of DTMF cards and circuits per card required to support a given number of telephones. Example - 150 telephones would require five circuits on two cards.

Table 10.5 Required PDTMF Receivers

Number of PDTMFs (TCR Circuits)	FB-17203-A PDTMF PCBs Required	Maximum Number of PDTMF Stations
2	1	17
3	1	56
4	1	111
5	2	176
6	2	250

NOTES:

- This table is based on 3.75 originating calls per PDTMF station, and eight seconds holding time.
- This table is used only to determine the number of circuits used by telephones.

The total quantity of PDTMF receivers is established by adding all PDTMFs required for stations, CAS Tie trunks, PRLTS (PCM release link trunks), Attendant Consoles, and Agent Instruments.

DTMF Receiver Engineering

10.4.4 When a DTMF receiver is not available to a user, blockage occurs. The CAS Agent Instruments and Tie lines which are offered dial access into the system are heavy DTMF receiver users. This is due to their concentration of calls into the system. To compensate for this, two DTMF receivers are reserved for CAS operation (if implemented). The remaining users must be considered by the system planner.

The DTMF signaling lines, incoming tie, DID, and CCSA (Common Control Switching Arrangement) trunks generate traffic to DTMF receivers. The formula for determining DTMF receiver traffic generated by DTMF lines is as follows:

(Number of originating DTMF lines x originating calls per DTMF line) x (DTMF receiver holding time per call)/100 = DTMF receiver traffic in CCS.

The formula for determining DTMF receiver traffic generated by incoming trunks presenting DTMF signals to the system is as follows:

(Number of incoming DTMF trunks) x (Incoming calls per DTMF trunk) x (DTMF receiver holding time per call)/100 = DTMF receiver traffic in CCS

One additional receiver must be provided with each of the first two RLTs if the CAS Branch option is used. These DTMF receivers are dedicated to the RLTs. The remaining RLTs share the pooled DTMF traffic with the lines and trunks, and the RLTs must be considered in sizing the receiver pool. Any receiver can be used for line and trunk DTMF receiver traffic use, but by reserving two receivers (for RLTs), lines and trunks cannot exhaust the available supply of receivers.

Determine the DTMF receiver traffic load for the various sources. The total of these values determines the number of DTMF receivers required. The typical busy-hour parameters for a line are 3.75 originating calls per line per hour with an &second DTMF receiver holding time, each equal to a 0.3 CCS traffic load per line. The typical busy-hour data for a one-way incoming Tie trunk is 12 originating calls per trunk with a 3-second DTMF receiver holding time, which is equal to a 0.36 CCS traffic load per trunk. Thus, if 100 lines were installed, a load of 30 CCS would be presented to the DTMF receivers and 10 DID trunks would bring the total to 33.6 CCS. Table 10.6 defines the DTMF receivers required to support various amounts of traffic.

Determining DTMF Circuits by CCS Count

10.4.5 A typical line, during its peak traffic period, accesses a PDTMF circuit 3.75 times per hour. The average holding time for each access is 8 seconds. This is equivalent to 0.3 CCS. Similarly, a typical trunk arranged to receive DTMF will create a 0.36 CCS load at its peak traffic load. Use the following formula to determine the peak load presented to the PDTMF circuit.

(L x 0.3 CCS) + (T x 0.36 CCS) = PDTMF Load

Table 10.6 provides a guide to help determine how many PDTMF receivers are needed to avoid potential congestion.

Table 10.6 DTMF Receivers Required to Provide a P = 0.01 Grade of Service

Traffic In CCS	DTMF Receivers Required
5.3	2
17.0	3
33.4	4
53.0	5
75.0	6
99.0	7
124.0	8

NOTES:

- The actual probability of a delay greater than 3 seconds = 0.005.
- The CCS/line and CCS/trunk values used in this example are assumed; actual value will vary from system to system.
- The Eight-Party Conference card uses up to eight time slots when fully loaded, but typically requires five time slots.
- The typical busy-hour parameters for a one-way incoming trunk (DID) are 12 originating calls per trunk with a 3-second DTMF receiver holding time, which equals to a 0.36-CCS traffic load per trunk.

If 100 lines are installed, a 30 CCS load is presented to the DTMF receivers and 10 DID trunks brings the total to a 33.6 CCS load.

Recommendations for placing the DTMF cards are as follows:

- The first DTMF card normally is placed into card slot AO.
- When a second DTMF card is required, it is put into the Expansion File, normally into group C2 (C7 • Cll). Group C2, which only has five slots, offers the best ratio of time slots because it has the smallest number of physical slots.
- If a circuit on a DTMF card is not currently in service, but customer growth may cause it to be put in service, it should be counted in the traffic study to prevent having to reconfigure the system at a later time.

Traffic Engineering

10.5 Table 10.7 provides the number of available CCS for both a PO.01 and PO.02 grade of service. When the line and trunk CCS usage has been calculated and the number of dedicated time slots determined (if any), compare the CCS used (under P = 0.01) against the CCS offered. With each dedicated time slot usage, the amount of offered CCS decreases approximately 30 CCS units.

The calculated CCS/group should not exceed the values as listed in the P = 0.01 column of the table below. Table 10.5 shows the various maximum traffic levels per PCMUS group that will result in a probability of blockage (P level) of 0.01 or 0.02.

Table 10.7 Traffic Capacity of a PCMUS Group Summary

Number of Dedicated Time Slots	Number of Remaining Channels	Traffic Capacity (CCS)	
		P = 0.01	P = 0.02
0	24	529	570
1	23	500	540
2	22	472	510
3	21	443	480
4	20	415	451
5	19	388	422
6	18	360	393
7	17	331	364
8	16	302	335

NOTES:

- PO.01 and PO.02 represent the blocking factors: A PO.01 rating indicates the probability that one call may be blocked when 100 calls are attempted in a busy hour.
- The number of remaining channels is the number of time slots remaining after dedicated time slots have been subtracted from the 24 available time slots.
- The traffic capacity for a fully configured system is 3174 CCS at P = 0.01 with no dedicated time slots.
- Tie lines are normally configured at a lower grade of service (PO.04 or P0.05) due to their high cost.

- The various groups (A, B, Cl, C2;,D1, D2) should be at approximately the same CCS level to compare a group containing no dedicated time slots with one that does. Subtract the offered CCS from the used CCS. An example containing dedicated time slots is shown in Figure 10.5.

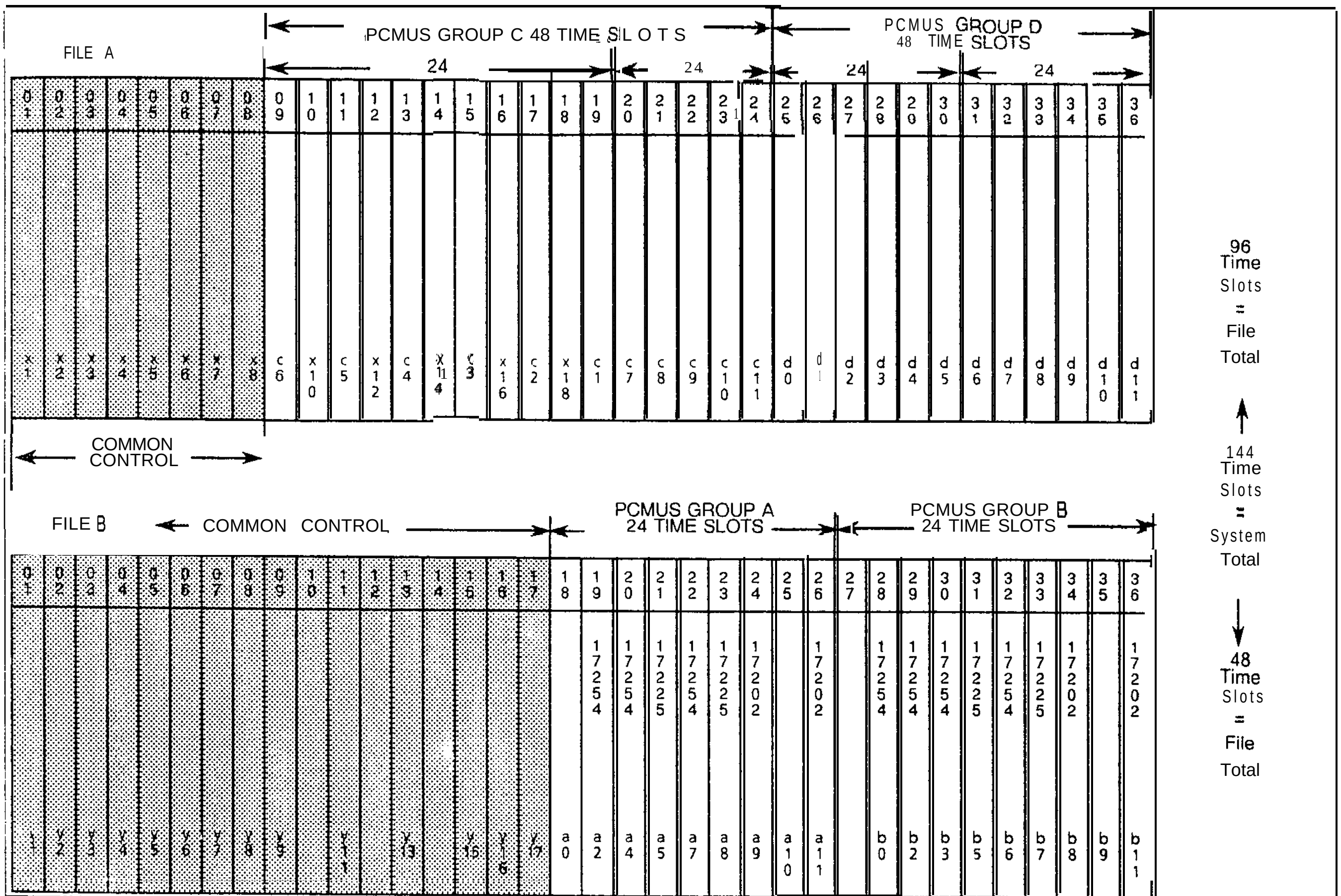


Figure 10.5 Traffic Study 2

Example: Comparison of group traffic for a group containing dedicated time slots with a group with no dedicated time slots (Figure 10.5).

- Group A (24 x 6) + (8 x 18) = 288iCCS
(Line Circuits x CCS) (Trunk Circuits x CCS) = Total CCS

- For this example, the PLCC cards in slots A2, BO, and B2 are used to support non-IFP phone and therefore require the access to a DTMF card circuit. (The other PLCC cards in conjunction with their collocated CIP cards support IFP and do not require access to a DTMF card circuit.)

- 3 PLCC are supporting 24 phones.
  (3 cards x 8 circuits) = 24

- Table 10.5 indicates that 24 phones will require one DTMF card and use three circuits on that card.

- Comparing the number of CCS used in group A to the number of dedicated time sots required in the 3 DTMF circuits in group A (Table 10.5) with 3 dedicated time slots, the offered CCS is 443. The 443 offered CCS is greater than the 288 used CCS so group A fits into the PO.01 grade of service.

● Group B

- 4 line cards (17254) x 8 circuits per card = 32

- 2 trunk cards (17202) x 4 circuits per card = 8 trunks

NOTE: The CIP (17225) data link cards do not use time slots.

- (32 x 6 CCS) + (8 x 18 CCS) = 336

- Table 10.5 gives the offered CCS with no dedicated time slots as 529 so group B with 336 CCS fits into the PO.01 grade of service.

● Comparing the used CCS of group A (288) to group B (316) does not give an accurate comparison because dedicated time slots are needed by group A.

- To get a more accurate comparison, subtract the used CCS of each group from the offered CCS and compare to two to determine how well balanced the system traffic is.

- The difference in CCS available per group is 38 more available CCS for group B.

Examples of Proper Trafficing

10.6 The following examples of typical line and trunk layouts further illustrate how to consider traffic capacities. An asterisk indicates traffic estimates that exceed a 0.01 P level.

S 1 CCS Calculations

- The following examples are based on 9 CSS per line and 20 CSS per trunk.

Example 1

Group	Line	Trunk	DTMF	ATTI	ccs
0	39	4	4	1	431
1	48	8			592
	87	12	4	1	1023

NOTE: To reduce the possibility of blockage:

- Remove a line card from PCMUS group 0.
- Remove both a line and a trunk card from PCMUS group 1.

Example 2

Group	Line	Trunk	DTMF	ATTI	CCS
0	48	4	4		512
1	64	4			656
4	3	16		1	347
5	40				360
6	40		4		360
7	40	4			440
	235	28	8	1	2675

NOTE: To reduce the possibility of blockage:

1. Remove two line cards from PCMUS group 0.
2. Remove two line cards from PCMUS group 1.
3. Remove two trunk cards from PCMUS group 4.
4. Enable the five spare line circuits on the existing line card in PCMUS group 4.
5. Add a line card and DTMF card to PCMUS group 4.
6. Remove a line card from PCMUS group 5.
7. Add a trunk card to PCMUS group 5
8. Remove the DTMF card from PCMUS group 6.
9. Assign one of the line circuits in PCMUS group 6 as the Attendant's voice line with guaranteed access.
10. Remove a line card from PCMUS group 7.

- The following examples are based on 6 CCS per line and 18 CCS per trunk.

Example 3

Group	Line	Trunk	DTMF	ATTI	CCS
0	39	4	4	1	306
1	48	8			432
	87	12	4	1	738

Example 4

Group	Line	Trunk	DTMF	ATTI	CCS
0	48	4	4		360
1	64	4			456
4	3	16		1	306
5	40				240
6	40		4		240
7	40	4			312
	235	28	8	1	1914

- **The following examples are based on 3 CCS per line and 16 CCS per trunk.**

Example 5

Group	Line	Trunk	DTMF	ATTI	CCS
0	39	4	4	1	181
1	48	8			272
	87	12	4	1	453

Example 6

Group	Line	Trunk	DTMF	ATTI	CCS
0	48	4	4		208
1	64	4			256
4	3	16		1	137
5	40				120
6	40		4		120
7	40	4			184
	235	28	8	1	1025

CAS Main/ ACD System Engineering

10.7 Table 10.8 describes engineering for the CAS Main and ACD operations.

Table 10.8 CAS Main/ACD Related System Capacities

ITEM	SYSTEM CAPACITY	REFERENCE NOTE
CAS Main/ACD Groups	8	
CAS Main/ACD Agent PACET	16	1 (see 10.7)
CAS Branch Locations	32	
CAS/ACD Source Groups	32	
ACD Single Line Agents	192	
Supervisor Positions	8	
Supervisor I/O Ports	1	
Recorder Announcer Trunks	5	2 (see 10.7)
Release Link Trunk (RLT) Interface Circuit (Branch Location)	16	3 (see 10.7)

NOTES:

1. CAS/ACDAagent instruments per system is 16. Depending upon a per-job data base option, CAS/ACD Agent Instruments require one or two station line circuits and one data link port. For every two data link ports used, the number of lines is reduced by eight or the number of trunks is reduced by four.

2. CASACD associated recorder announcer can access access five trunk circuits (first delay, first alternate delay, second delay, second alternate delay, and night answer).

3. Incoming RLT interface trunks (loop or E&M) 15. All RLTs directed to a specific CAS group from several CAS Branch systems are considered a single-trunk group (i.e., as one of the system's 64 trunk groups). Therefore, a maximum of 4 of the system's 64 trunk groups is required for CAS Main operation. Each RLT card reduces the maximum mix of lines and trunks by eight lines or four trunks.

System Capacities

10.8 Table 10.9 identifies the system capacities for lines, trunks, and peripheral devices. Table 10.10 identifies the capacities for site-dependent universal cards and also identifies the number of circuits per card. Table 10.11 identifies the voice data circuit ratios without a T1 span. Table 10.12 identifies the voice data circuit ratios with a T1 span.

Table 10.9 Line, Trunk, and Peripheral System Capacities

ITEM	GET STARTED FILE	EXPANSION FILE	SYSTEM TOTALS	REFERENCE NOTE (see following page)
Lines	136	184	256	1
Analog Featurephone			127	2
Digital Featurephone			127	10
Trunks	36	54	64	1
Trunk Groups			64	
Attendant Consoles			2	3
Busy Lamp Display Unit			4	4
Key Entry Display Unit			4	
Hotel/Health Care Printers			4	
DTMF Receiver Circuits			8	
DTMF Line Capacity	111	139	250	5
T1 Digital Interface with 24 Trunks	-	1	1	
Recorder Announcer Accesses	-		64	6
8-Party Conference Circuits			2	7
Dictation Access Circuits			No Limit	8
Display Telephone			Equal to total number of stations in the system	9

NOTES:

1. The line and trunk maximums cannot occur simultaneously. At 256 lines, the maximum trunk quantity is 28. Adding additional trunks or adding feature cards decreases the maximum by as many as 16 lines, depending on the required mounting space.

2. Every eight Digital Featurephone interface circuits, reduce the maximum mix of lines and trunks by eight lines or four trunks. Four Digital Featurephones can have a DSS AOM (Add-On-Module).

3. Up to 64 standard station instruments can have a DSS appearance on only two Digital Featurephones. A maximum of four Digital Featurephones can have up to 16 line appearances and all others can have a maximum of four line appearances. A maximum of 16 Digital Featurephones can have up to four control line appearances, and up to 16 more can have two control line appearances, with all others having only a single control line appearance.

4. Each Attendant Console interface circuit reduces the maximum mix of lines and trunks by eight lines or four trunks.

5. A BLDU (Busy Lamp Display Unit) varies with the configuration of the system. A BLDU can be associated with each Attendant Console and/or can be an independent monitoring device at other locations where BLDU hundreds keys are used to select the display.

6. DTMF Receivers: Two cards per system; four DTMF receiver circuits per card. When the DTMF receiver card is placed in the reserved card slot, the maximum mix of lines and trunks is not affected, otherwise eight lines or four trunks are lost. One DTMF circuit is required for each of the two attendants.

7. Recorder announcer access circuits: 64 maximum (requires a dedicated E&M trunk).

8. Conference circuits (eight-party): A maximum of two conference circuit cards can be used in the system. For each conference circuit that is placed in a card slot, other than the one reserved for it in a cabinet, the maximum mix of lines and trunks is reduced by eight lines or four trunks. Otherwise, there is no reduction in the maximum mix of lines and trunks.

9. Dictation access circuits: No limit. No software limit exists for the number of dictation access circuits in the system. However, each assigned dictation access card reduces the number of available lines by eight or the number of available trunks by four.

10. Display telephone sets (calling/called number): The limit of sets is the total number of stations in the switch. Stations equipped with a calling number display cannot have the message-waiting feature.

11. The Analog Featurephone and Digital Featurephone maximums cannot occur simultaneously. The system supports a maximum of 128 for a mixture of both types of phone.

Table 10.10 Universal PCB Cards Maximum System Capacities (Voice)

CARD TYPE	CARD NO.	CIRCUITS PER CARD	MAXIMUM CARDS IN SYSTEM
Line Card	FB-17254-A	8	32
Off-Premises Line Card	FB-17250-A	8	23
Conference (8-Party)	FB-51279-A	1	2
Attendant Interface	FB-17208-A	2	2
Dictation/Paging/Code Calling	FB-17210-A	2/1/1	9
PDTMF Receiver	FB-17203-A	4	2
Two-Wire Trunk (Two-Way Loop)	FB-17202-A	4	16
Two-Wire Trunk (Two-Way E&M)	FB-17201-A	4	16
Two-Wire Trunk (One-Way Incoming Loop)	FB-51280-A	4	16
Four-Wire Trunk (Two-Way E&M)	FB-51267-A	4	9
T1 Access/Span Interface	FB-17277-A or FB-15277-I A		1
T1 Frame Detector	FB-15278-A		1
T1 Line Compensator	FB-15280-A		1
T1 Buffer	FB-17192		1
T1 Supervisory Circuit	FB-20718-I A		1
PCM Release Link Trunk	FB-17251 -A	4	4
S1 Dual Modem and Loop Interface	FB-17209-A	2	8
Control Interface Processor	FB-17225-A	8	16
Voice Control Interface Processor	FB-17235-A	8	16
Data Voice Control Interface Processor	FB-17236-A	4 Voice/4 Data	16
VPLC2	FB-17246-A	8	8
OMNI Automatic Identification of Outward Dialing	FB-17276-A	4	1

Table 10.11 Voice - Data Circuit Ratios (Without a T1 Span)

8-Circuit Line Cards	Voice Lines	Trunk Cards	Trunk Circuits	Voice Packet Line Card 2 (VPLC2)	Data Ports
0	0	0	0	15	120
2	16	1	4	14	112
6	48	1	4	13	104
9	72	2	8	12	96
12	96	3	12	11	88
15	120	4	16	10	80
18	144	4	16	9	72
20	160	4	16	8	64
21	168	4	16	7	56
22	176	4	16	6	48
23	184	4	16	5	40
22	176	5	20	4	32
26	208	5	20	3	24
27	216	6	24	2	16

Table 10.12 Voice - Data Circuit Ratios (With a T1 Span)

8 Circuit Line Cards	Voice Circuits	Trunk Cards	Trunk Circuits	VPLC2 Cards	Data Ports
0	0	0	0	11	88
2	16	1	4	10	80
6	48	1	4	9	72
9	72	2	8	8	64
10	80	2	8	7	56
12	96	3	12	6	48
16	128	3	12	5	40
18	144	3	12	4	32
19	152	4	16	3	24
22	176	4	16	2	16

ORDERING THE VOICE SYSTEM

11.0 The hardware editing sheets (FM-41460-A) used for this release must be completed to order SVR 5210. Ordering Guide (FM-41 640) provides instructions for completing the editing sheets. The system can be ordered as either a standard or custom hardware configuration.

Standard Hardware Configuration

11.1 The standard configuration is available in two sizes.

- 48 lines/4 trunks
- 96 lines/4 trunks

Normally a standard system is shipped within three weeks from receipt of the order. Any additional equipment needed must be ordered piecemeal.

Custom Hardware Configuration

11.2 When ordering a custom system, order all peripheral cards and software required. A custom configuration is shipped within eight weeks from receipt of the order.

In addition to starndard or custom hardware, software is also ordered as either standard or custom configuration. Whenever possible, order a customized data base.

When ordering a custom system, fill out the frame layout worksheet that is included in the Ordering Guide.

Submit the frame layout worksheet along with the completed editing sheets when ordering the system.

Use the following steps and flowchart (Figure 11 .1) when completing the frame layout worksheet:

1. Determine quantity of each type of universal card required.

2. Determine quantity of universal card slots needed.

3. Fill in the frame layout sheet with the following:

   a. If used, place T1 span cards.
   b. Place trunk cards using preferred location first.
   c. Fill remainder of double width card slots.
   d. Place single height cards evenly among equipped time groups.
   e. Place line cards.
   f. Place CIP cards next to line card.
   g. Add any remaining single height cards.

   NOTE: Use pencil when filling in this form.

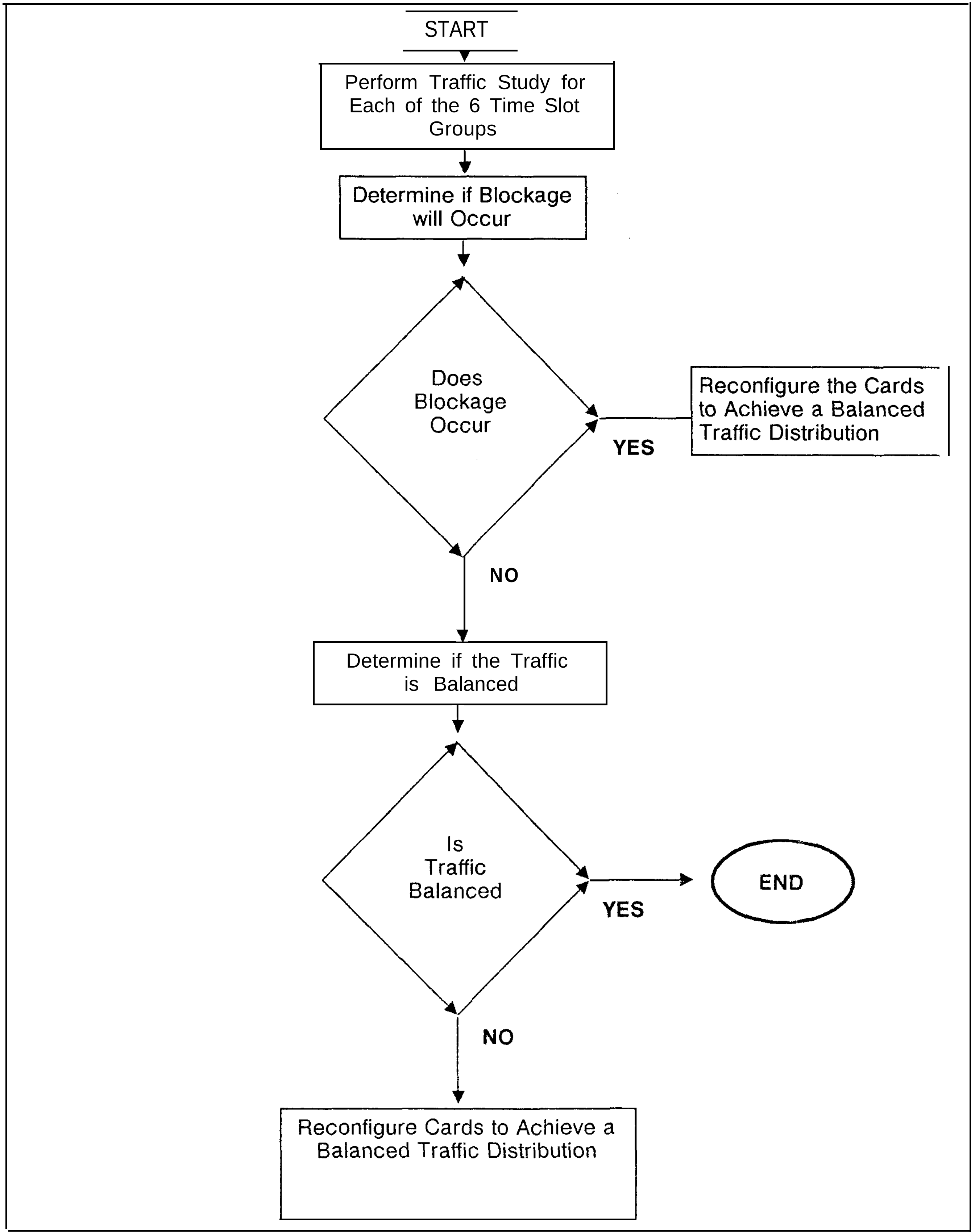


Figure 11.1 Flowchart for Balancing System Traffic

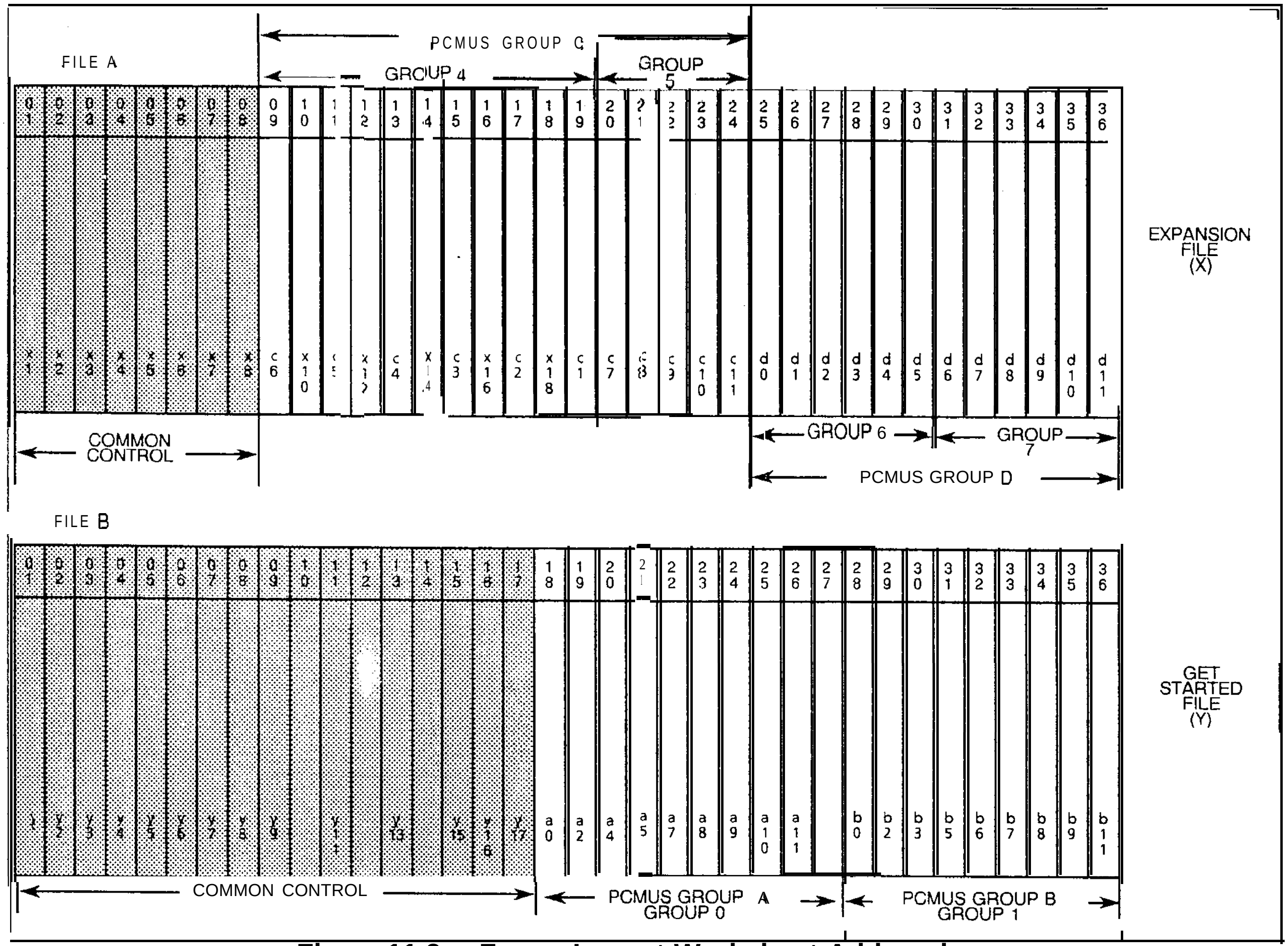


Figure 11.2 Frame Layout Worksheet Addressing

NOTES: The following infirmation should be considered when configuring a system.

- Refer to the card description section of this document, section 3.0, for information on cards required to support features to be implemented.
- Figure 11.2 provides information for determining card slots required.
- Provide a minimum of 20% growth for the system.
- If group D is required, then an extra PCMFS and PCMI must be ordered. Put a PCMFS (FB-17189) card in slot 06 and a PCMI (FB-17187) card in slot 08 of the Expansion File.
- T1 span cards go into the T1 span card slots in between C1-C6, located to the right of the PCMUS.
- Preferred locations for double width cards are:
  - T1 span location when the Tl-span option is not used.
  - Al 1, which provides an overhang for a double width card as it is a heat vent
  - B11 and D11, which provide overhang into the cabinet
- Use single width location for double width cards only after using all double width locations.
- The ATTI2 has no overhanging parts, but due to its heat output requires an empty slot next to it for ventilation.
- When possible the PLCC and CIP cards required to support the Analog Featurephone should be put into card slots next to one another. Circuits for a given IFP should also match on both cards. Assign circuit in the following order: 4, 0, 5, 1, 6, 2, 7, 3. Circuits 0, 2, 4, and 6 of the DVCIP card are used for voice, while circuits 1, 3, 5, and 7 are used for data.
- Traffic study must be done for each time slot group. Tables 11.2 and 11.3 are provided to help in determining traffic
- If the traffic study indicates that blockage will occur, the card must be relocated.
- If the traffic study indicates a large imbalance in CCS per group, it is recommended o reposition cards. In some cases, one group may have been deliberately configured with a low amount of CCS traffic, which is acceptable.
- It may be necessary to try several locations for the various cards before a proper balance is achieved.

Table 11.1 Card Parameters

PART NO.	MNEMONIC	WIDTH	GUAR TS(s)	CKTs ON CARD
FB-17201 -A	PEMT	Double	0*	4
FB-17202-A	PCOT	Double	0	4
FB-17203-A	PDTMF	Single	4	4
FB-17208-A	ATT12	Double	0	2
FB-17209-A	SIDML	Double	0	2
FB-1721 O-A	PADIC	Double	0	
FB-17225-A	CIP	Single	0	8
FB-17226-A	VPLC (VPL0)	Single	0	8
	VPLC (VPL1)	Single	0	2
FB-17426-A	VPLC2 (VPL0)	Single	0	8
	VPLC2 (VPL1)	Single	0	2
FB-17227-A	PBE/T	Single	0	1
FB-17228-A	PR	Single	0	1
FB-17229-A	ADMP-A	Single	0	1
FB-17230-A	ADMP-B	Single	0	1
FB-17231 -A	UCB/DCP	Single	0	1
FB-17235-A	VCIP	Single	0	8
FB-17250-A	POPS	Double	0	8
FB-17251 -A	RLT	H	4	4
FB-17254-A	PLCC	Single	0 +	8
FB-51267-A	PFWTA	Double	0"	4
FB-51279-A	PCONF	Single	0	1
FB-51280-A	PILT	Double	0	4

NOTE: Guaranteed TS(s) (Time Slot(s)) required on this card when:

*CKT(s) control RA(s)
**CKT(s) used with ACD/CAS Agent Instrument.
+ CKT assigned for M-O-H or attendant line circuit
+ + If all 24 T1 trunks are enabled the T1 span uses all 24 TSs. If not, some TSs can be used for other equipment.

Table 11.2 Available CCSs

AVAILABLE CCS			
TIME SLOTS	P = 01	P = 02	
24	529	570	
23	500	540	
22	472	510	
21	443	480	
20	415	451	
19	388	422	
18	360	393	

Table 11.3 Group Traffic

GROUP TRAFFIC AT P = 0.01						
GROUP	TRUNK CCS	LINE CCS	ATTN CCS	OFFERED CCS	AVAILABLE CCS	AVAILABLE TIME SLOTS
A						
B						
C_1						
C_2						
D_1						
D_2						

Required Forms

11.3 When the frame layout worksheet from the Hardware Ordering Guide (FM-41480) has been completed, the hardware editing sheets are to be filled out. The editing sheets contain a listing of all software and hardware components of the system. When completed, a copy of the editing sheets and a copy of the frame layout workskeets are to be submitted.

NOTE: Use the frame layout worksheet, found in the SI Hardware Ordering Guide FM-41450, when ordering a system.

DATA EQUIPMENT/ ENGINEERING

12.0 The data system can handle a maximum of 120 data ports. This section provides a general configuration overview, detailed configuration rules and limitations, and procedures for engineering the data system.

The following parameters describe the customer requirements to configure or add on a data switch:

APM = T1 + T2 - T3

where T1 = number of asynchronous data terminal ports requested (maximum speed = 19.2 Kbps) (VPLO or VP20)

T2 = number of asynchronous host ports requested (maximum speed = 19.2 Kbps) (VPLO or VP20)

T3 = number of Digital Featurephones with APM requested (VPLO or VP20)

SPM = S1 + s2

where S1 = number of X.25 ports < = 19.2 Kbps (using RS-232-C interface) (VPLO or VP20)

S2 = number of X.25 ports > = 9.6-64 Kbps (using V.35 interface) (VPL1 or VP21)

Computing Hardware Quantities

12.1 To compute the required hardware quantities, first determine the number of data ports needed. There are five types of ports:

- Terminals
  - Asynchronous (with or without modems)
  - X.25 (with or without modems)
- Host ports
  - Asynchronous (with or without modems)
  - X.25 (with or without modems)
- Number of modems in each modem pool (if required)
- Data network connections
- Network interface connections

Compute the hardware quantities as follows:

1. Compute the total of asynchronous terminals (with and without modems), plus asynchronous host ports (with and without modems), plus asynchronous modems in modem pools. This total is the number of APMs or DPF/APM required.

2. Determine any other DFP/APM required.

3. The X.25 terminals and host ports must be split into two groups, depending upon line speed. Those which operate at 19.2 kbps or less fall into the first group, which uses a SPM with an RS-232-C connector. Those which operate above 9.6 to 64 Kbps are handled separately because they require an SPM with a V.35 connector.

4. Total the X.25 terminals, X.25 host ports, and public data network connections operating at 19.2 kbps or less. This gives the total number of SPMs with RS-232-C connectors required. These will be connected to VPLC type 0 line cards.

5. Total the X.25 terminals, X.25 host ports, and public data network connections operating at line speeds between 9.6 and 64 Kbps. This gives the total number of SPMs with V.35 connectors required. These will be connected to VPLC type 1 line cards.

6. To determine the required number of VPLC type 0 line cards, add the number of APMs (from step 1) and the number of SPMs with RS-232-C connectors. Divide this by 8, rounding up if the answer is a fraction.

7. To determine the number of VPLC2 cards, add the number of DFP (with and without APM). Divide this number by 8, rounding up if the answer is a fraction.

8. To determine the required number of VPLC type 1 line cards, divide the number of SPMs with V.35 connectors by 2, rounding up if the answer is a fraction.

   NOTE: Since these VPLC type 1 cards control the system's high-speed links, it is often advisable to order and configure an extra or back-up VPLC type 1 card for links to public data networks. (Remember, each VPLC type 1 has only two usable circuits.)

9. Determine the number of NIC cards.

After completing step 9, list the quantities of hardware required:

1. ___APM
2. DFP/APM
3. SPM with RS-232-C connectors
4. SAPM with V.35 connectors
5. VPLC type VPLO
6. VPLC type VPL1
7. VPLC2 type VP20
8. VPLC2 type VP21
9. ADMP card (2-card pair) per PD-200 Data System
10. UCB cards
11. PR
12. PBE
13. PBT
14. NIC - one per T1 card set

Local Packet Bus Configuration

12.1.1 Refer to Figures 12.1 through 12.6 for possible LPB (Local Packet Bus) configurations. These figures also provides placement of PR, PBE, PBT, and UCB functioning as bus terminators.

Figures 12.2 and 12.3 represent possible LPB configurations. The recommended arrangement for most applications is to use a complete file (A & B, C & D) for the LPB. Put voice cards in the unused PCMUSs. This will make the best use of both the data and voice bus capacity of the file. Put LPB 0 in the Get Started File whenever possible (see Figures 12.4, 12.5, 12.6).

NOTES:

- Placement of bus extender and terminators can be reversed.
- In the following configurations, the UCB is configured to function as a bus terminator card (see Figure 12.7).

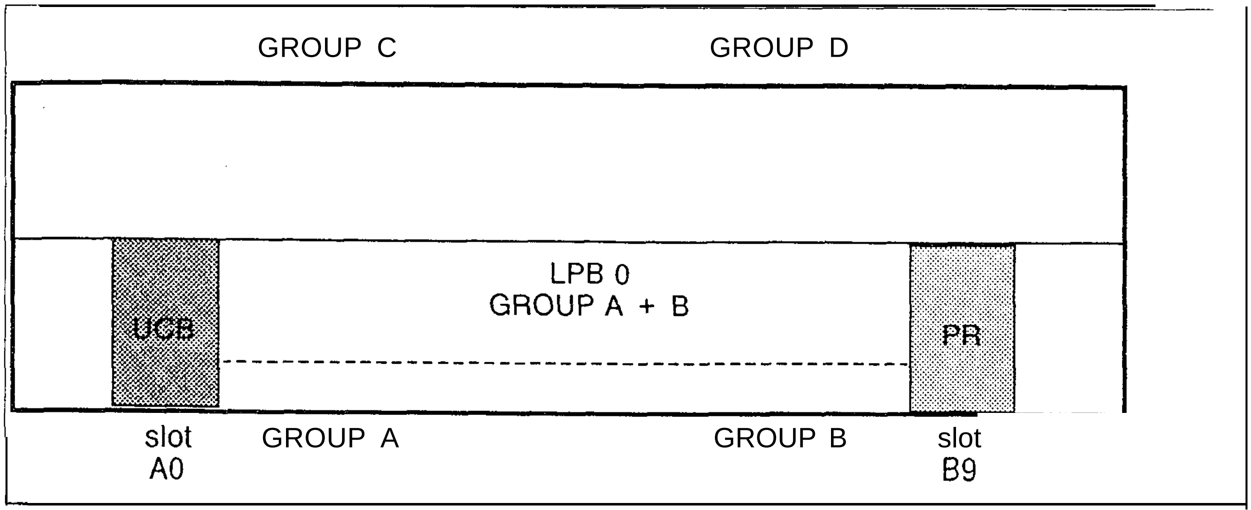


Figure 12.1 Example of Single Local Packet Buses in Groups A & B

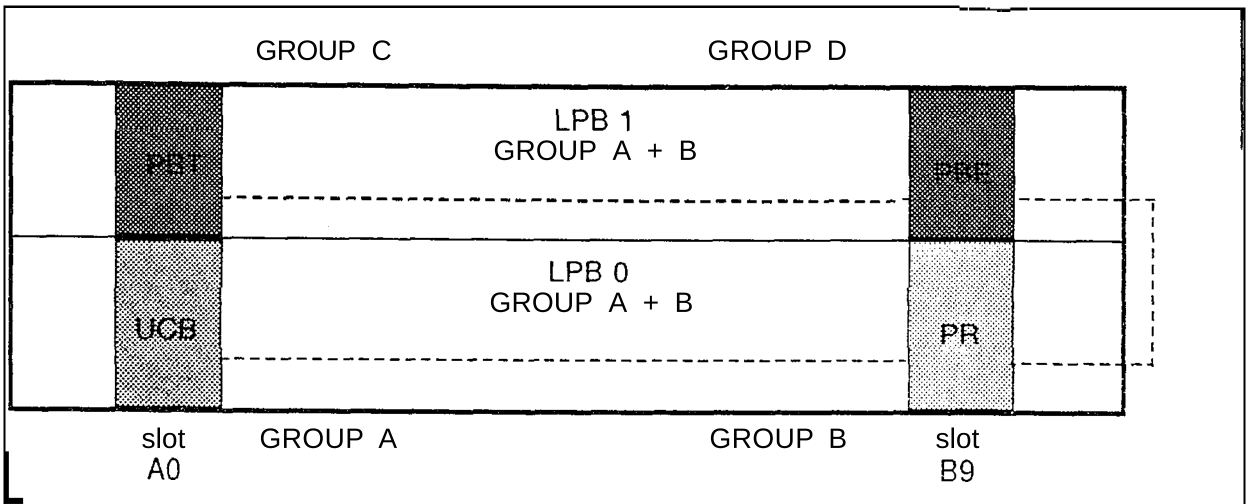


Figure 12.2 Example of Two Local Packet Buses (LPB 0, LPB 1) in Groups A & B, C & D

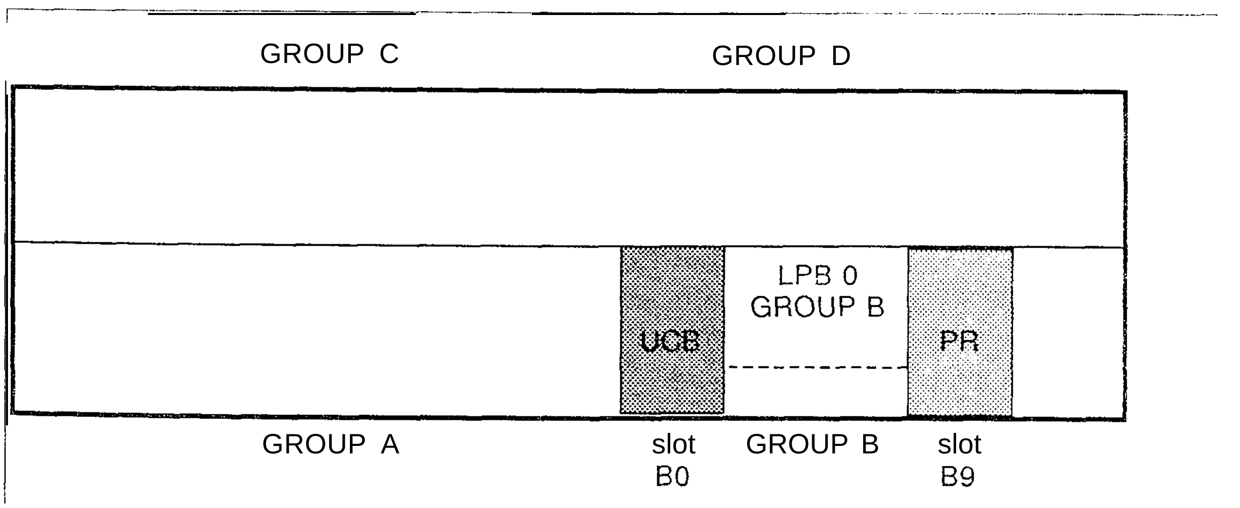


Figure 12.3 Example of Single Local Packet Buses in Group B

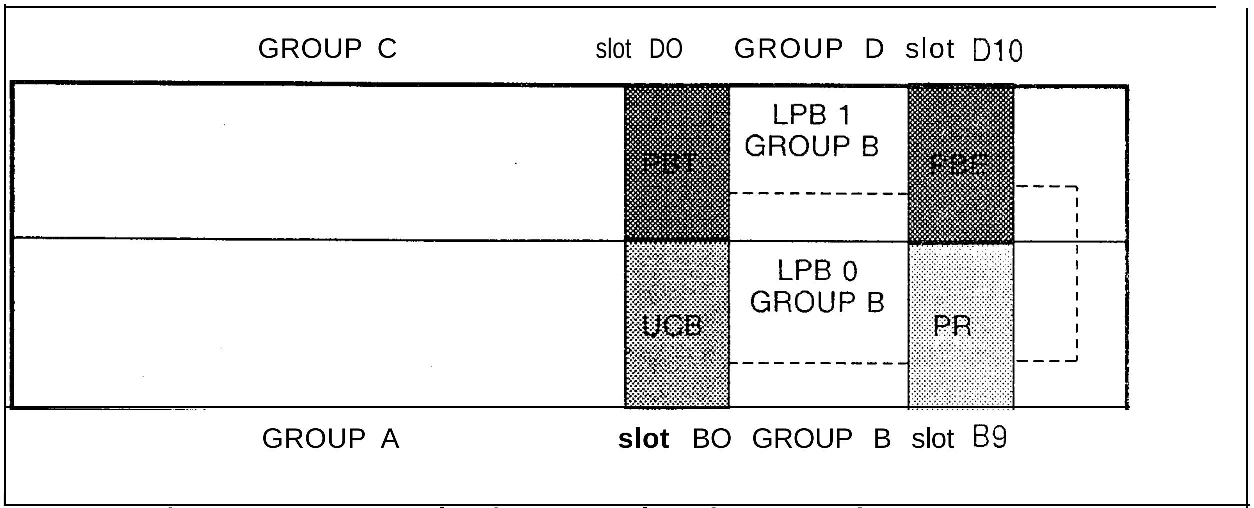


Figure 12.4 Example of Two Local Packet Buses in Groups B & D

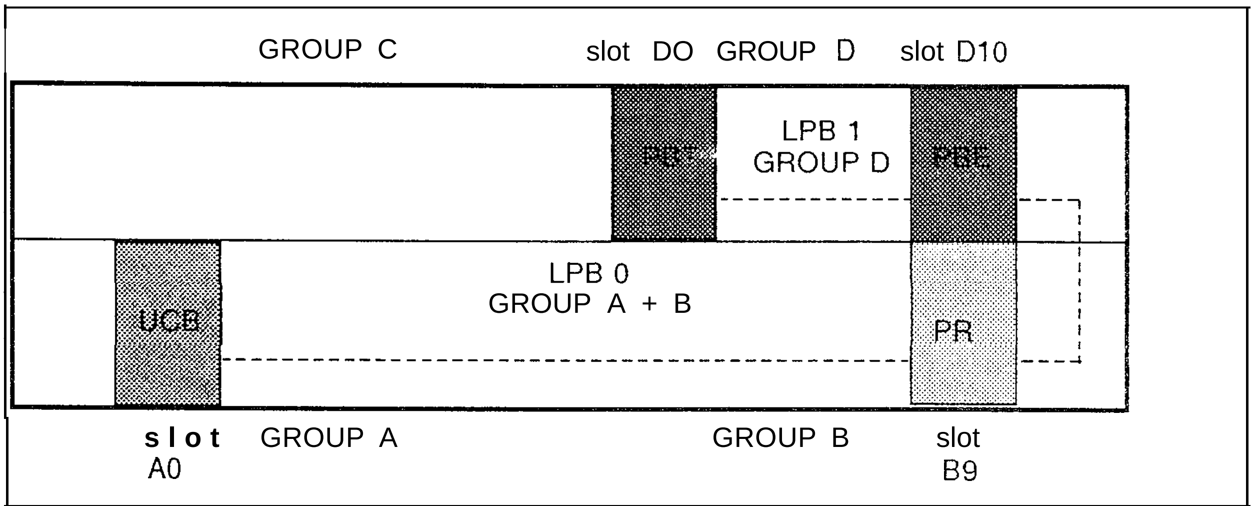


Figure 12.5 Example of Two Local Packet Buses (LPB 0, LPB 1) in Groups A& B and D

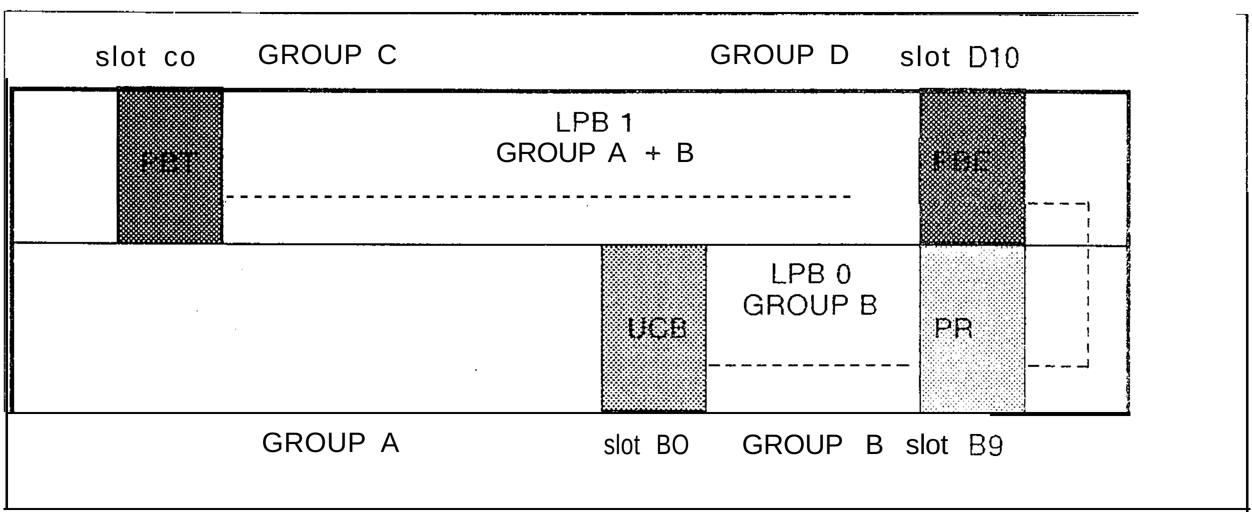


Figure 12.6 Example of Two Local Packet Buses (LPB 0, LPB 1) in Groups B and C & D

To connect the LPB into one bus in the top file of the PEC, add jumper wires between slots X24 and X25, pins 36-40 and 86-90.

To connect the LPB into one bus in the bottom file of the PEC, add jumper wires between slots Y26 and Y28, pins 36-40 and 86-90.

Example: X24 pin 36 to X25 pin 36
X24 pin 37 to X25 pin 37

X24 pin 40 to X25 pin 40
X24 pin 86 to X25 pin 86

X24 pin 90 to X25 pin 90

Frame Image Card Placement

12.1.2 Once the LPB location has been determined, complete the frame image as the following flowchart indicates.

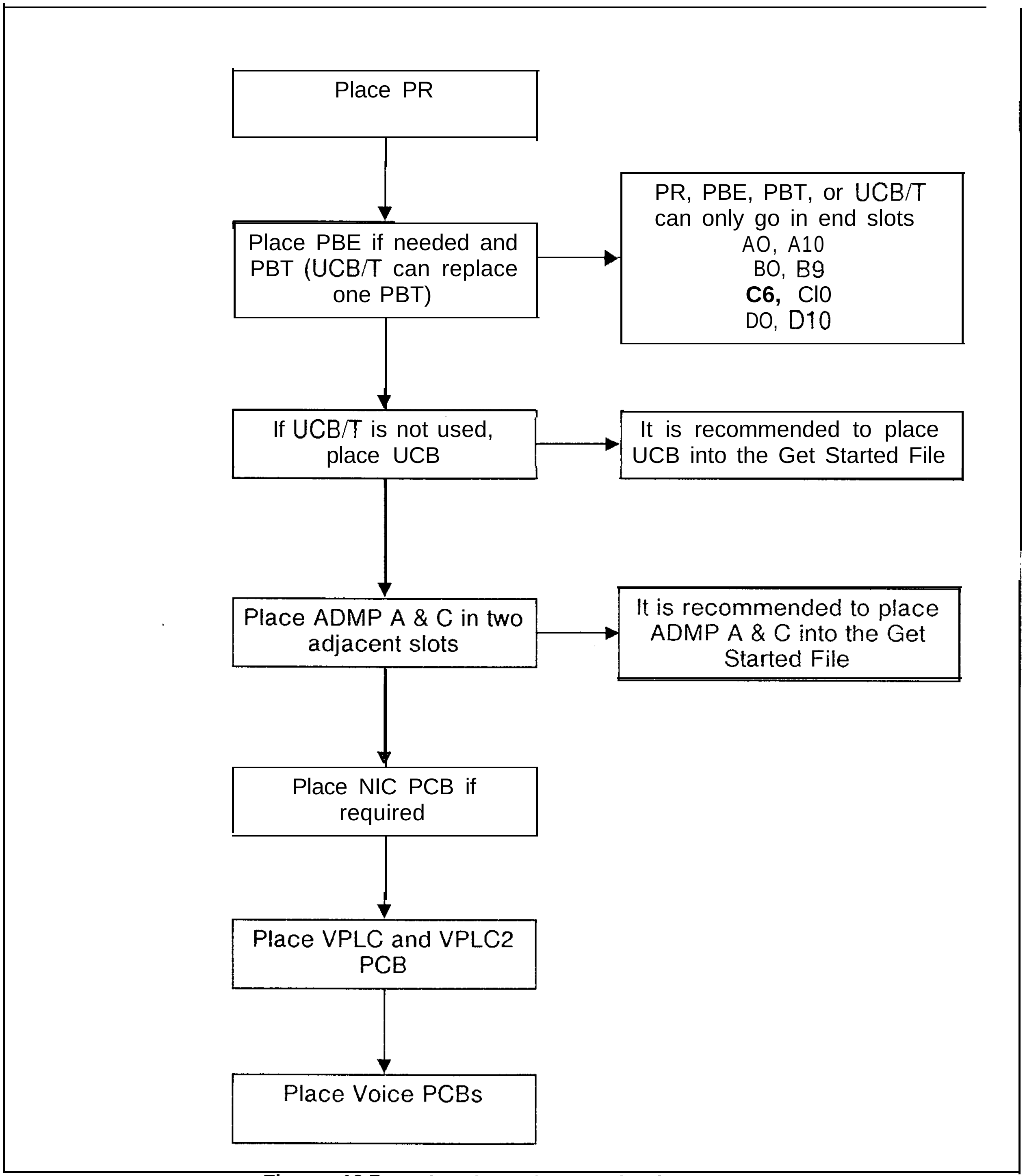


Figure 12.7 Flowchart for Card Placement

Table 12.1 provides a worksheet for determining card slots and cards required to support the data option.

Table 12.1 Worksheet for Data Option

ITEM	DESCRIPTION	SYMBOL	QTY
	CUSTOMER REQUIREMENTS:		
1	Number of asynchronous data terminals	T1	
2	Number of asynchronous host ports	T2	
3	Number of DFP/APM	T3	
4	Number of low-speed (RS-232-C) X.25 lines (<19.2 kbps)	S1	
5	Number of high-speed(V.35) X.25 lines (>9.6-64 Kbps)	S2	
	CALCULATIONS:		
6	Total number of APMs required: = T1 + T2 - T3 = add items 1 and 2 and subtract item 3	APM	
7	Total number of DFP/APM: = T3 = item 3	DFP/APM	
8	Total number of low-speed SPMs required: = S1 = item 4	SPM Low	
9	Total number of high-speed SPMs required: = s2 = item 5	SPM High	
10	Number of ADMP card sets required:	ADMP	1
11	Number of UCB cards required:	UCB	1
12	Number of VPLC type 0 cards required: = add items 6 and 8 and divide by 8 = APM + SPM low) /8 rounded up to the next whole number	VPLO	
13	Number of VPLC type 1 cards required: = divide item 9 by 2 = SPM high/2 rounded to the next whole number	VPL1	

Table 12.1 Worksheet for Data Option (Continued)

ITEM	DESCRIPTION	SYMBOL	QTY
	CALCULATIONS contuined:		
14	Number of VPLC2 cards required: = divide item 7 by 8 = DFP/APM/8 rounded up to the next whole number	VPLC2 (VP20 or VP21)	
15	Number of NIC cards required: = NIC	NIC	
16	Number of Ports required: = add items 1, 2, 3, 4, 5, 10, 11 and 15 **NOTE:** Depends upon configuration.	PLA	
17	Number of Packet Router cards required:	PR	1
18	Number of slots used: = add 2 times item 10 + items 11, 12, 13, 14, 15 + item 17 = ADMP times 2 + UCB + VPLO + VPL1 + VP20 + VP21 + PR + NIC	SLOT	
19	Number of packet bus extender cards required: = 0 if item 16 is < or = 64 = 1 if 16 > 64 or if data cards occupy both files **NOTE:** Depends upon configuration.	PBE	
20	Number of packet bus terminator cards required: If 1 LPB is configured the UCB is normally used to terminate the bus When a second LPB is used a PBT will terminate the bus.	PBT	
21	Number of PCMUS slots required: = add items 18, 19, and 20 = SLOT + PBE + PBT	PCMUS	

Voice Packet Line Card Configuration

12.2 Table 12.2 provides configuration rules that apply to both VPLC (FB 17226-A) and VPLC2 (FB 17246-A).

Table 12.2 Voice Packet Line Card Type 0 and Type 1 Configuration Rules

TYPE OF VOICE PACKET LINE CARD	NUMBER OF PORTS	TYPE OF VOICE PACKET LINE CARD
VPLC type VPLO	8	(1) APM (2) SPM running at less than or equal to 19.2 kbps
VPLC type VPLO	2	(1) APM (2) SPM running at less than or equal to 64 kbps
VPLC2 type VP20	8	(1) DFP/APM (2) APM (3) SPM running at less than or equal to 19.2 kbps
VPLC2 type VP21	2	(1) DFP/APM (2) APM (3) SPM running at less than or equal to 64 kbps

NOTE: Both VPLC type VPLO and VPL1 are the same physical hardware. The type of VPLC card required is selected via a setting in the data base when adding the card to the system. No hardware changes are required. The VPLC2 also comes in type VP20 with 8 ports and VP21 with 2 ports.

System Capacities

12.3 Table 12.3 provides slot requirements and PLA (Packet Line Address) requirements for data card and remote devices.

Table 12.3 PCB/Device Quantity per System

CARD OR DEVICE	QUANTITY	NO. OF PCMUS SLOTS
ADMP A & C FB-17229-A FB 17230-B0A	1 per system	1 each (2 total)
UCB/DCP FB-17231 -A	1 per system	1
Packet Router FB-17228-BOA	1 per system	1
PBE/T FB-17227-A	This # depends on the packet bus configuration.	1
VPLC (type VPLO) FB-17226-A	1 per 8 ports	1
VPLC (type VPL1) FB-17226-A	1 per 2 ports	1
VPLC2 (type VP20) FB-17246-A	1 per 8 voice + data ports	1
VPLC2 (type VP21) FB-17246-A	1 per 2 voice + data ports	1
APM	1 per asynchronous port	0
SPM	1 per X.25 port	0
NIC FB-17242-A	16 per system	1
DFP/APM	1 per asynchronous port (also with voice requirements)	0

Power Requirement 12.4 Table 12.4 provides power requirement for data cards and remote devices.

Table 12.4 Physical Location and Power Requirements of Cards

TYPE	ESTIMATED POWER REQUIREMENTS			
	+ 5VDC	+ 12VDC	-12VDC	-48VDC
VPLC/VPLC2	0.834 Amp	-		
UCB (see Note)	12.62 Amp		0.038 Amp	-
ADMP (2 cards)	4.48 Amp	0.13 Amp	0.056 Amp	-
PR	2.2 Amp		0.038 Amp	-
PBE/T (see Note)	-			
RPTR	1.5 Amp			
NIC	2.0 Amp	-	-	-
APM	Not powered from the switch			
SPM	Not powered from the switch			

NOTE: When these cards are equipped with terminators, they will require an additional 0.08 ampere at + 5VDC.

Power Supply Limits **12.5** The power supply limits of the system can be found in the TL-130000-1001, paragraph 2.12.

System Recommendation 12.6 The following card placement is recommended for data cards.

- Place the ADMP A & C cards in the Get Started File. The ADMP A& C cards must be in two adjacent slots.
- Place the UCB in the Get Started File.
- Configure the UCB as a UCB/T (bus terminating) by placing it in an end PCMUS.
- Place the PR in the Get Started File.

Local Packet Bus Configuration Rules

12.7 The general configuration rules for the data system can be characterized as falling into one or more of the following categories:

- PHY (Physical Limitation) = limited by physical space or number of slots.
- POW (Power/Thermal) - limited by available power or thermal considerations.
- SYS (System) = system design limitation
- MEM (Memory) = available memory limitation
- PRO (Processor Power) = software processing power limitation
- CON (Configuration) = general configuration rule

Each of the following rules specifies within parentheses which of the above categories is causing the rule to be in effect:

- The PEC can support one packet router enhanced (PHY, POW).
- A packet router can directly support two local packet buses (SYS).
- Each UCB and ADMP requires one PLA. A VPLO requires eight PLAs, while VPL1 requires two PLAs. Packet routers and packet bus extender/terminators do not require PLAs (SYS). NICs require 1 PLA.
- Each UCB, NIC, VPLC, PR, and PBE/T card requires one PCMUS, while the ADMP cards require two adjacent slots, only the A side of which requires power (SYS, PHY).
- Each asynchronous port requires its own APM or DFP/APM (SYS).
- Each X.25 line which runs less than or equal to 19.2 Kbps can use one SPM with the RS-232-C interface. The V.35 interface (SYS) supports X.25 lines which runs up to 64 Kbps.
- The system supports a maximum of 127 DFP (PRO, MEM).
- A single UCB card can support up to 240 data ports (MEM, PRO).
- A UCB card can support 240 data ports (for future release) The UCB software is only required if data is present (SYS, PRO, MEM).

- A VPLC2 type VP20 card will support up to eight DFPs (with or without the APM), APMs, or SPMs running at not greater than 19.2 2Kbps in any combination. Alternatively a VPLC2 type VP21 card can support one SPM running at up to 64 Kbps and one other device (SPM at any speed or APM) (PRO, PHY).
- A PEC has two files (shelves) (PHY,SYS).
- The total number of VPLC, ADMP, and VCIP cards per local packet bus primary or extension is eight (POW, PRO).
- No data card can reside in the last PCMUS of group (CON).
- The maximum number of UCBs allowed per system is 1 (POW, PRO).
- Each active local packet bus must be terminated with either a UCB/T card or a PBT card (SYS).
- A packet router requires a PBE card to gain access to its second local packet bus (SYS).

General Rules

12.8 The following general rules apply when configuring a data system.

- The system contains 36 PCMUSs (universal slots) suitable for data use. Due to the size and organization of the port tables, slot Al 1, B11, Cl 1, D11 cannot be used by the data system.

  Slot A0 is not used by data cards.
- One PR (Packet Router) can be connected to a maximum of one LPB (Local Packet Bus).
- PR supports up to 64 PLA (Packet Line Addresses).
- To gain access to additional PLAs, the PRE must connect to a second LPB (LPB 1). This connection is through the PBE card.

  When it is set, a switch on the UCB allows the card to function as a terminator. If the switch is not set, the card does not function as a terminator. In that case, the card can be placed between a PR or PBE and PBT.

- The PR (Packet Router) routes mini-packets on up to two LBPs. The first LPB (LPB-0) on a PR is bounded by the PR and a PBT (Packet Bus Terminator), or UCB/DCP with Bus Terminator (UCB/BT). The second LPB (LPB-1) is bounded by a PBE (Packet Bus Extender) and a PBT.

  Twelve hundred mini-packets a second, per LPB, can be routed through the PRE without overflowing. When configuring the system, traffic calculations should be made accordingly. The bits-per-second rate of all asynchronous and synchronous devices should not exceed 600,000 bits per second. For example, if there are 11 SPMs operating at 64 Kbps each, then they should not be configured on the same LPB. The following paragraph provides traffic information for the data system.

Data Traffic Considerations

12.9 Data terminal output information is linked to the CPU (Central Processor Unit) by interface devices operating at different data rates. Compatibility between the interface device and PABX allows for uninterrupted transfer of data. Incompatible interface devices require protocol conversion equipment. Data rates with varying ranges are used to transmit data over a specified communications line. The type of data transmitted can fall into one or more of the following categories:

- Typed words
- Numbers
- Graphics
- Syntax
- Bit sequences
- Files

The primary considerations for transmission of any category depends upon the user's PABX configuration, type of link, and network interfaces. Protocol requirements control the physical and near real-time aspects of data transfer, and therefore must be the primary software consideration.

Programmable terminals and host CPUs transfer data formats in either a synchronous or asynchronous mode of transmission. The format of the data, as it appears on the communications link, may or may not arrive at its destination in the most economical manner. Thus, the efficiency of data transmission depends upon throughput rates, peak load periods, and the specific format.

Data Throughput Considerations

12.9.1 For each APM, SPM, or DFP/APM connected to the system, the PD-200 Data System supports either the combination of one voice communication and up to 19.2 Kbps asynchronous data communications, or one voice communication and up to 64 Kbps synchronous data communications. Throughput on each LWL (Local Wire Loop) has been sized to accomplish this. Each LWL working in half-duplex mode at 256 kilobytes can transfer 1,330 mini-packets per second in each direction. Table 12.5 shows the mini-packet equivalent for each throughput specification. The 64 kilobyte synchronous calculation includes the overhead for data control, while the overhead for voice and data control is included in the 19.2 kilobyte asynchronous calculation.

Table 12.5 Mini-Packet Equivalent for Each Throughput Specification

FUNCTION	MINI-PACKETS/SECOND
19.2 kilobytes asynchronous	300
64.0 kilobytes PCM (Voice)	1,000
64.0 kilobytes synchronous	1,200

Hardware on the VPLC strips off receive voice mini-packets and merges them into the voice switch (time-division multiplex) domain. Only non-transparent mini-packets reach the PTS (Packet Transport System). These mini-packets include voice control messages, data control messages, and X.25 packet messages. For each asynchronous port (or synchronous port at 19.2 Kbps or less), the mini-packet traffic is limited to 300 mini-packets per second. The VPLC is sized to handle eight ports, each running at 300 mini-packets per second. For high speed data service, the VPLC is limited to running 2 ports at a rate of 1,200 mini-packets per second (data rate of 64 Kbps).

Traffic Metering

12.9.2 To ensure that congestion does not occur within the VPLC, all MPP (Mini-Packet Protocol) end-point components in the system can restrict the rate at which mini-packet traffic is generated to a given destination. The restricting process is called metering. When a component such as a UCB/DCP card sends a message through a VPLC to an APM, the UCB card sends the message mini-packets at a rate of one mini-packet each 3.3 milliseconds (or less). Similarly, two high-speed SPM managers communicating can send mini-packets to each other at a rate of one mini-packet per millisecond.

If multiple components send messages to the same destination component simultaneously, a temporary overflow condition can occur. Mini-packet buffers in the VPLC and in the PRs are used to hold the overflow. If the overflow condition persists, the data system invokes an appropriate flow control mechanism.

Flow Control

12.9.3 Flow control is inherent in the design of the data system and exists at both the message and the mini-packet levels. It provides speed matching for communicating subscriber devices running at different data rates, and is also used to recover from internal data congestion situations. Both levels of flow control are invoked automatically as required.

Speed Matching Flow Control

12.9.4 All user data within the switch is sent by using the X.25 packet level procedure. The packet window rotation mechanism (number of unacknowledged packets which may be outstanding) regulates traffic between two users.

For example, the standard window for an APM is two X.25 packets. If a 64 Kbps X.25 host sends packets to a 300-baud terminal attached to an APM, flow control is used to provide speed matching as follows:

- A packet is received at 64 Kbp's from the X.25 host by the SPM.
- The packet is sent as a series of mini-packets metered at a rate of 300 mini-packets per second.
- The packet is reassembled at the APM. If it has been received error free, the user data is sent to the terminal at a rate of 300 baud.
- The X.25 packet level acknowledgment is sent to the SPM, and then on to the X.25 host only after the entire packet has been transmitted to the user by the APM.
- In this example, a second packet may be sent to the APM while the first is being serviced. It is queued by the APM. According to X.25 standards, the X.25 host can send no more packets to the APM because the window is full (two packets are unacknowledged).

In addition to the above rules, X.25 allows the user or the data switch to send RNR (Receiver Not Ready) packets to enable flow control without having to wait for a full window condition. The data system does not generate RNRs but accepts them when received from an X.25 subscriber.

Congestion Flow Control

12.9.5 Congestion can occur within a normally operating data system for two reasons:

- Due to the "bursty" nature of data traffic, a fully configured system may saturate the internal bus structure temporarily.
- Many end points within a data system may be communicating with a single end point (i.e., a group of APMs with X.25 calls established through the same SPM). It is statistically possible that if enough end points send simultaneously, the path to the destination end point will become congested.

The MPP protocol detects a congestion condition and recovers from it. The mechanism works as follows:

- Congestion is detected when mini-packets are lost in transmission. The MPP receivers do not acknowledge invalid sequences received.
- When an acknowledgment is not received, MPP sends a time-out and then schedules the message for retransmission.
- To prevent a second congestion from occurring, a pseudo-random timer is run at each sending MPP before resending the message. This is called adaptive backoff.
- Adaptive backoff is applied repetitively, with longer time-out periods, until messages are sent successfully. Adaptive backoff then gradually allows normal operation to resume. This action is called damping.
- If adaptive backoff cannot handle the congestion, logical links are marked down and calls are cleared until the traffic load within the switch goes under the congested level. This happens only if the data system is over-configured.

Configuration Limits

12.9.6 Table 12.6 shows the maximum throughput in full-duplex mini-packets per second for each MPP end point in the data system.

Table 12.6 Mini-Packets/Second for MPP End Points

MPP END POINT	MINI-PACKETS/SECOND
Administrative/Maintenance Processor	1,000
Universal Controller Board card	1,000
Synchronous Packet Manager (> 9.6 kbps)	1,200
Synchronous Packet Manager (< 9.6 kbps)	300
Asynchronous Packet Manager	300

Examine the figures in Table 12.6 versus those in Table 12.7, which shows the mini-packets/second capability of the VPLC and the PTS (Packet Transport System).

Table 12.7 Mini-Packets/Second Capability of the VPLC/PTS

VPLC/PTS	MINI-PACKETS/SECOND
VPLC (VPLO)	2,400 (8 X 300)
VPLC (VPL1)	2,400 (2 X 1,200)
LPB (Local Packet Bus)	12,000
PR (Packet Router)	24,000 (2 X 12,000)

Observe that each LPB (Local Packet Bus) is limited to 12,000 mini-packets/second. This limit must be taken into account when planning installation of a fully configured system. This prevents one LPB or one PR (Packet Router) from becoming congested. It is normal for a data system to use an average bandwidth far less than the maximum bandwidth available. The throughput of the LPB and PR takes this into account. Thus, if the subscriber devices burst data or run at a rate of one-half the maximum (or less), then any LPB can be selected for the connections. With maximum throughput devices, the limit is 40 asynchronous devices or 9 synchronous devices per LPB.

Data Capacity Considerations

12.9.7 The data switch can handle a maximum of 120 data ports operating at 9,600 bits per second (9.6 Kbps). There are 16 X.25 lines at rates of 56 or 64 Kbps (80 percent utilization) using a multiple mix of packet lengths, with a maximum of 128 bytes/packet for line handling. Packet rates are 64 packets/second. The remaining lines may be at any allowable speed(s).

During average busy hours, call setup and takedown delay is limited to ≤3 seconds for 99 percent of all calls. Call setup and takedown rates (Table 12.8) translate an average 15 CCS per port for a 2-minute call holding time.

Table 12.8 Call Setup and Takedown Rates

NUMBER OF DATA PORTS	PEAK CALL SETUP AND TAKEDOWN RATE (CALLS/SECOND)	MAXIMUM NO. OF CALLS DURING A BUSY MINUTE RATE (CALLS/MINUTE)
64	3	15
120	4	24

Data Traffic Per Port Calculation

12.9.8 The following equations can be used to calculate the number of ports which can be configured per LPB. These equations assume a 25 percent overhead for internal control traffic. Asynchronous and synchronous devices of varying speeds may be mixed on each LPB. The curves in Figure 12.8 were calculated by using the following formulas:

1. The number of asynchronous devices per LPB is calculated as follows:

$$N(a) = \frac{\frac{12{,}000\text{MP}}{\text{Sec}} \times \frac{10 \quad \text{Bits}}{\text{Byte}} \times \frac{8 \text{ Bytes}}{\text{MP}}}{1.25 \times \frac{\text{R Bits}}{\text{Sec}} \times U}$$

$$N(a) = \frac{768{,}000}{R \times "}$$

where:

N(a) = number of asynchronous devices per LPB
R = baud rate of devices
U = average bandwidth utilization by asynchronous devices

and:

Maximum N(a) is 64 ports

2. The number of synchronous devices per LPB is calculated as follows:

$$N(s) = \frac{\frac{12{,}000\,MP}{Sec} \times \frac{8\ Bits}{Byte} \times \frac{8\ Bytes}{MP}}{1.25 \times \frac{R\ Bits}{Sec} \times U}$$

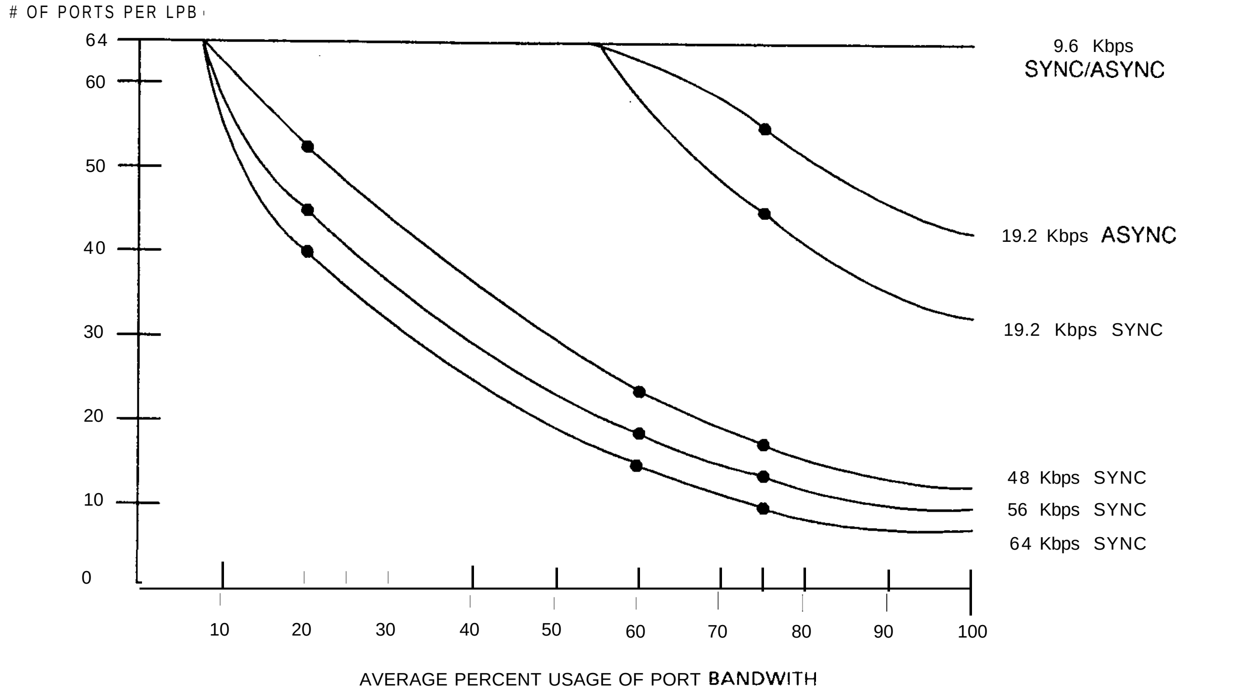
$$N(s) = \frac{614{,}000}{R \times U}$$

where:

N(s)	= number of synchronous devices per LPB
R	= baud rate of devices
U	= average bandwidth utilization by synchronous devices

and:

Maximum N(s) is 64 ports



NOTE: MAXIMUM NUMBERS MAY BE LIMITED BY AVAILABLE CARD SLOTS.

Figure 12.8 Number of Ports per Local Packet Bus Versus Bandwidth Utilization

ORDERING THE DATA SYSTEM

13.0 The PD-200 data option must be ordered as a custom system. When ordering the data system, first complete a frame layout worksheet. The frame layout worksheet is located in the Hardware Ordering Guide (FB-41640). After the layout sheet has been completed, fill out the hardware editing sheets (FM-41460). Submit copies of the completed sheets. Information for completing the layout sheet is given in Figure 13.1. Notes that provide an explanation of information in Figure 13.1 follow:

NOTES:

1. Determine number of DFP/APM, APM, SPM low-speed and SPM high-speed connections.

   a. Determine number of VPLO cards and/or VP20 cards used for DFP/APMs, APMs, and low-speed SPM connections.

   b. Determine number of VPL1 cards and/or VP21 cards used for SPM high-speed connections. Since the VPL1 or VP21 cards control the system's high-speed data links, it is recommended to configure and order a back-up card for connection to the PDN (Public Data Network).

   c. There is a system limit of 8 VPLC (any type) per file and16 VPLC (any type) per system maximum

2. Maximum number of LPBs for this release is two. Each LPB can support up to 12,000 mini-packets per second.

3. ADMP A and C, UCB, PR, and PBE/T

4. See voice ordering section for further details.

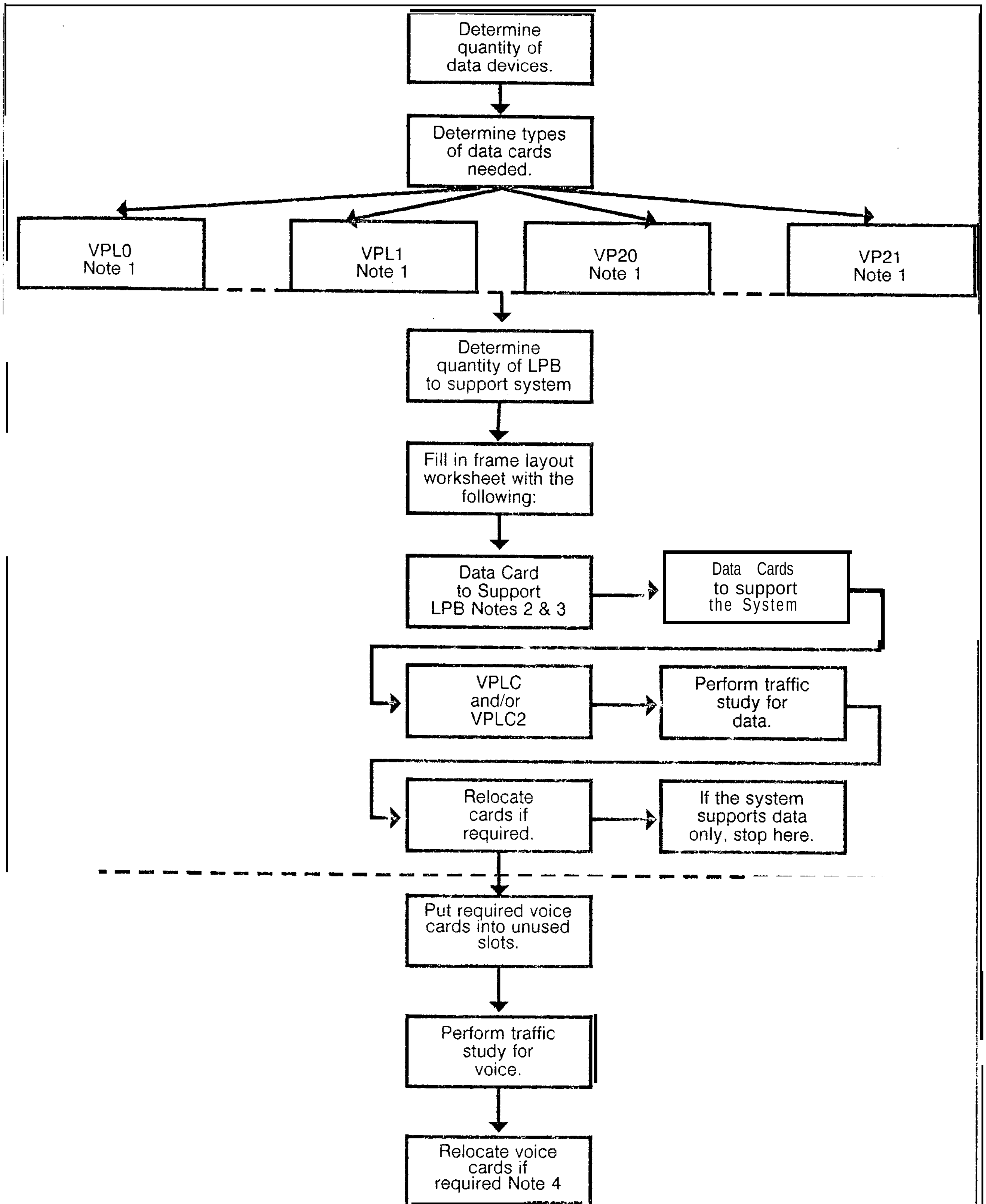


Figure 13.1 Flowchart for Configuring the System

PERIPHERAL EQUIPMENT

14.0 The total configuration of the OMNI SI system consists of the system switching equipment, system peripheral equipment, and system software. The system switching equipment configuration and descriptions are presented in sections 2.0 through 13.0. System software configuration and descriptions are presented in sections 34.0 through 36.0. System peripheral equipment configurations and descriptions are presented in sections 15.0 through 33.0 as follows.

- Section 15.0 Attendant Console
- Section 16.0 BLDU (Busy Lamp Display Unit)
- Section 17.0 Telephones
- Section 18.0 KEDU (Key Entry Display Unit)
- Section 19.0 Hotel/Health Care Printer
- Section 200. Maintenance Terminals
- Section 21 .O MDR (Message Detail Recording) Printer
- Section 22.0 Type 200 Digital Test Set
- Section 23.0 Paging and Dictation Interface
- Section 24.0 MOH (Music-On-Hold)
- Section 25.0 Recorder Announcer
- Section 26.0 Conference Hardware for Silent Monitor

THIS PAGE IS INTENTIONALLY LEFT BLANK.

ATTENDANT CONSOLE

15.0 The Attendant Console (Figure 15.1) is a compact, microprocessor-controlled, desk-top console unit. It provides control and flexibility for processing calls through the system while on-line (in use). The user can place the console in other control modes during nightime, maintenance, or unattended periods.

The system supports a maximum of two Attendant Consoles. The Attendant Consoles are remotely located from the equipment cabinets. The Attendant Console contains compact electronics and all controls and indicators needed for complete monitoring and control of calls through the system. Both incoming and outgoing calls can be answered, extended, camped on, put on hold, or released from the console by depressing various pushbuttons.

Each Attendant Console uses five EPROM (Erasable Programmable Read-Only Memory) chips that contain the generic software that determines the functional operation of the Attendant Console.

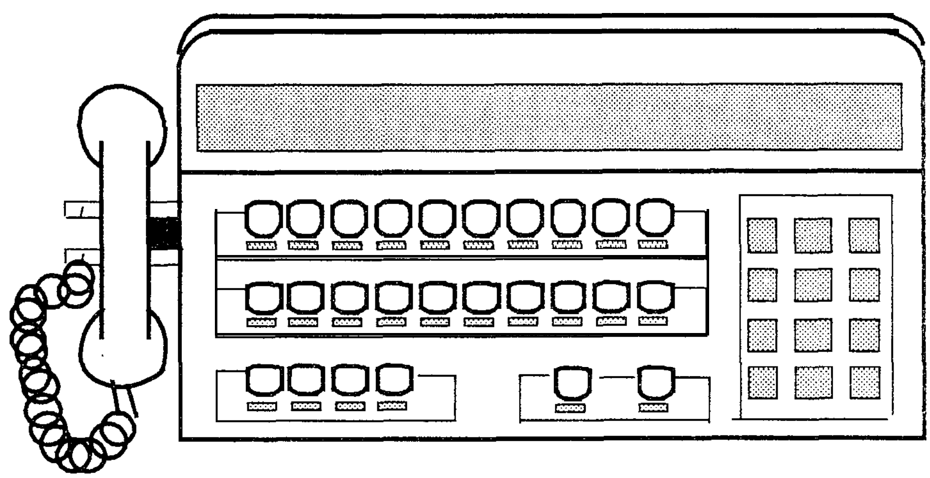


Figure 15.1 Attendant Console

Attendant Console System Interface

15.1 The Attendant Console interfaces with the system via the PLCC and the ATT12 cards, located in PCMUS (Pulse Code Modulation Universal Slots). It is recommended to place these cards in adjacent card slots for easier installation and maintenance.

- PLCC (line circuit card) FB-17254-A provides the analog channel for both DTMF signaling and voice communication between the Attendant Console and the system. The analog channel is connected to the system via tip and ring connections to one port of the line card.

- ATT12 (Attendant Interface) FB-17208-A provides the digital data link channel connection to the system. The data link is a full-duplex, 1,200 baud, current loop.

NOTE: The ATTI2 and PLCC cards are described on Table 3.6. Attendant Console interface cabling is described in TL-130300-l 001.

Attendant Console Hardware

15.2 The Attendant Console includes the following features:

- 26-pushbutton keyboard
- 12 key dial pad
- Call-waiting queue lamp
- Optional headset/handset and associated jacks
- Connect to BLDU (Busy Lamp Display Unit) (Optional)
- 32-alphanumeric character LED display for displaying class of service, equipment identity, type of call, dialed digits, time of day, and system alarm information

Light Emitting Diode

15.3 An alphanumeric, 32-character display LED (Light Emitting Diode), located across the top of the Attendant Console, provides call information to the user including the type of call, class-of-service marks, and equipment identity. The type of call indication is present as long as the attendant is servicing the call and during attendant recalls. Possible call types include FX and Tie. System fault alarm information is displayed in the type of call position if the ALARM pushbutton is depressed.

Headsets

15.4 The Attendant Console has two sets of headset/handset jacks: a rear jack for supervisory monitoring and a side jack for attendant use.

The headset/handset must be plugged in for the Attendant Console to operate in the on-line mode. Should the headset/handset be unplugged, the Attendant Console defaults to the Night 1 mode, and is considered busy by the operating system software.

Diagnostic self tests can be initiated while in the Night 1 mode.

General Information for Attendant Console Software and Operation

15.5 For dual Attendant Console operation, attendant-seeking traffic is distributed evenly between the two attendants. When the headset/handset is removed from the jack, the Attendant Console is considered unstaffed and the call is diverted to a second Attendant Console (the Attendant Console with the headset/handset removed is placed in the N1 mode of operation). Night answer modes Nl and N2 are data base programmable to provide UNA (Universal Night Answer), PNA (Predetermined Night Answer), or a combination of both night answer services. Removing the headset/handset turns off all lamps on the console and sends an unstaffed message to the system. All calls to this console are then cleared.

Attendant Call Distribution

15.6 Calls to the Attendant Consoles are distributed evenly, but the number of loops used for incoming calls is a programmable system feature. Either 1, 2, 3, or 4 loops may be programmed to receive incoming calls. Do not to enable the fourth loop since this could prevent the attendant from making an outgoing call during an emergency.

Regardless of the selected number of loops programmed for incoming calls, console 1 receives the first call on LOOP 1, console 2 receives the next call on LOOP 1, and so on, until all available consoles have a call on LOOP 1. Additional incoming calls either ring into LOOP 2 on each available console (if programmed), following the same sequence, or they are placed in the call waiting queue until an incoming loop becomes idle on any console. The system preferentially fills preceding loop levels before either moving into the next higher loop level or placing a call in Call Waiting Queue. When programmed for overflow, the universal answer mode is also activated when the second call-waiting queue is reached.

In a single console operation, calls can automatically ring into LOOP 1 through LOOP 4 sequentially, depending upon the system programming. All calls will remain in the call waiting queue until **a** loop becomes idle.

An attendant releases' an active loop by depressing the POS' RLSE, HOLD, or CAMP-ON pushbutton, or another LOOP pushbutton. Upon release, the next call in the call waiting queue is presented to the attendant.

For dual Attendant Console operation, attendant-seeking traffic is distributed evenly between the Attendant Console positions. Except for priority calls, all attendant-seeking traffic is queued on a FIFO (First-In, First-Out) basis. Only one new call is sent to the attendant at a time. The attendant can release from the active loop by depressing the POS RLSE, HOLD, or CAMP-ON pushbutton, or another LOOP pushbutton. When the button is released, the next call in the call waiting queue will be sent to the attendant.

Attendant Console Bower

15.7 A 24-VAC transformer, connected to a local 115-VAC power source or a-48 VDC power supply, powers the Attendant Console.

See TL-130300-1001 for a description of the Attendant Console power connections.

NOTE: Input to the Attendant Console should not exceed -54 volts DC.

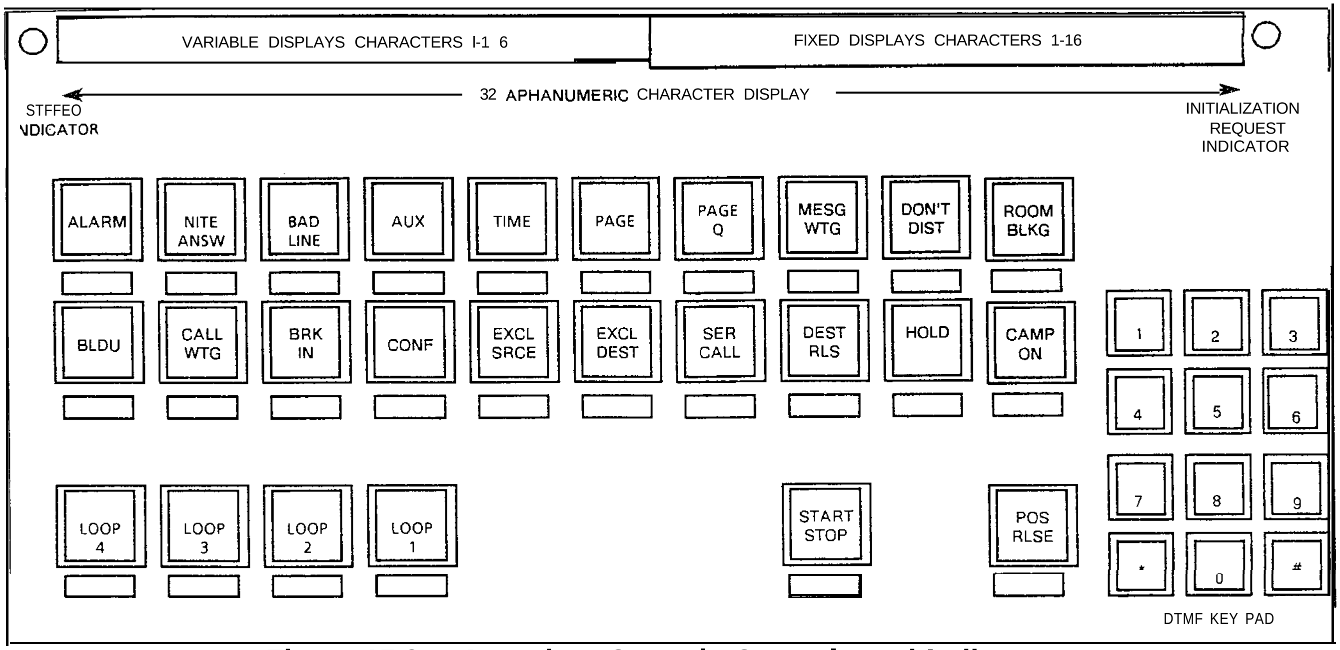


Figure 15.2 Attendant Console Controls and Indicators

Attendant Console Controls and Indicators

15.8 (Figure 15.8) The Attendant Console controls and indicators are listed below.

ALARM LED, Lights to indicate a fuse alarm or power failure, and flashes to indicate trouble with a particular line or trunk or group of lines and/or trunks.

AUX (2ND PAGE QUEUE). Allows placement of an incoming call in a page-queue hold condition.

BAD LINE. Allows the posting of a bad-line report on the maintenance teletypewriter if the pushbutton is depressed when a complete connection is present.

BLDU (BUSY LAMP DISPLAY UNIT). Allows selection of the hundreds group for display on the busy lamp line field.

BRK IN (BREAK IN). Allows break-in on two-way conversations.

CALL WTG (CALL WAITING) LED. Indicates that the first level of call waiting has occurred. The LED flashing indicates that the second level of call waiting has occurred. Depressing the pushbutton during either of these conditions will distribute a call to an Attendant Console that is in the night answer mode.

CAMP-ON. Allows placement of an incoming call in a waiting condition if the called station is busy. When camp-on is used, the system automatically extends the calling party to the called station when the called station becomes idle.

CLASS OF SERVICE DISPLAYS. Indicates the displayable class of service (0 through 15) and the n-displayable class of service (00-15) of the incoming call when a LOOP pushbutton is depressed to answer an incoming call from a station or Tie line. In a no-answer situation, the class of service display of the called station appears after the LOOP pushbutton is depressed.

CONF (CONFERENCE). Allows addition of stations, trunks, or the attendant to the first conference circuit. LED indicates the status of either conference circuit.

DEST RLS (DESTINATION RELEASE). Allows release of a called station or trunk if a dialing error is made or a busy signal is received.

DISPLAY FIELD. Displays two 16-character fields (fixed right: variable left). A four-digit display of the trunk or line identity appears in the left portion of the display-characters field. When the LOOP pushbutton is depressed to answer a call from a station or a Tie line, this four-digit display gives the number of the calling station or calling trunk. The number of the called station in camp-on, busy, and no-answer situations is displayed when the LOOP pushbutton is depressed. A display of predetermined messages that appear in the fixed portion (right) of the display characters field (i.e., INFORMATION, FX, NO ANSWER, etc.).

NOTE: The left 16 characters display variable data: line data, class-of-service data, etc. The right 16 characters display fixed, predetermined messages and input from the keypad.

DON'T DIST (DO-NOT-DISTURB). Allows blockage of all calls to a station when requested. This pushbutton is reserved for the Hotel/Health Care feature package.

ELECTRONIC RINGER VOLUME CONTROL. Is a slide bar located on the right rear of the Attendant Console and used only to increase/decrease the ringer volume.

EXCL DEST (EXCLUDE DESTINATION). Allows private consultation with the called party after the call is completed. The pushbutton can also be used to control the three-way connection between the called station, the attendant, and the outside calling party.

EXCL SRCE (EXCLUDE SOURCE). Allows private consultation with the calling party. This pushbutton can also be used to control the three-way connection between the called station, the attendant, and the outside calling party.

FUSE HOLDER AND FUSE. Is a circuit protection device located on the left rear of the Attendant Console.

HOLD. Used to place a call in a waiting condition (on hold) until the attendant can return to the call.

INFORMATION. Indicates that a station desires assistance of the attendant and has dialed 0. The associated LOOP LED flashes at 30 IPM. Once the loop button is pressed, the LCD displays the instrument number and COS of the caller.

NOTE: When the attendant recall-on-hold has timed out, the original display flashes at 60 IPM. The associated LOOP LED winks at 60 IPM.

LOCAL. Indicates an incoming CO call. The associated LOOP LED flashes at 120 IPM. Once the loop button is pressed, the LCD displays the trunk type and number of the incoming call.

LOOP 1 through LOOP 4 LEDs. Indicate status of the loops.

MESG WTG (MESSAGE WAITING). Allows the user to cause a telephone-associated LED to flash an indication to a station that a message is waiting. This pushbutton generally is used with a feature enhancement.

NITE ANSW (NIGHT ANSWER). Activates the night answer mode level (entered as a single digit on the keypad) of operation, or marks the Attendant Console as being unstaffed. The level of night answer mode is displayed on the type of call display. If the night answer pushbutton is depressed when the LED is on, the night answer mode is returned to the day mode.

ON/OFF TOGGLE SWITCH. Located at the left rear and used to turn the Attendant Console on or off. When the Attendant Console does not respond properly, the power can be turned off and on again to reinitialize the attendant console.

PAGE. Allows direct access to the paging system as long as the pushbutton is depressed.

PAGE Q (PAGE QUEUE). Allows placement of an incoming call in a page-queue hold condition.

RLS (Release) Pushbutton - This pushbutton allows the user to release from a call on an active line. The RLS LED lights steadily during a line 1 call or when the LINE 2 pushbutton is depressed. The RLS LED is extinguished when the RLS pushbutton is depressed.

REQUEST FOR INITIALIZATION LED. Flashes when the Attendant Console is initialized, and goes off when initilization sequence completes.

ROOM BLKG (ROOM BLOCKING). Allows blockage of incoming calls to all stations and diverts these calls to 120 IPM tone. This pushbutton is reserved with the Hotel/Health Care feature package.

SER CALL (SERIES CALL). Depressing this pushbutton after extending an incoming call automatically causes the trunk to recall the attendant when the station hangs up so the trunk can be extended to the next station in the series call sequence. Alternatively, the attendant can call or be called from a station (or have a station diverted to the Attendant Console due to an ACOF restriction on an accessed trunk group) and extend a trunk to the station using this pushbutton, reinitiating the automatic recall sequence to notify the attendant that the trunk is available for the next user. A lit LED indicates action confirmation.

STAFFED LED. Lights only if the position is staffed.

START/STOP. Activates the keypad and provides dial tone, and in the STOP mode, removes tick tone on outgoing calls.

TIME OF DAY (TOD). Allows the attendant to override any variable display with the time-of-day display in hours and minutes.

NOTE: See TL-130300-1001 for a description of Attendant Console operations.

BLDU (Busy Lamp Display Unit)

15.9 A BLDU can be used with the Attendant Console and accessed from the Attendant Console to display busy line/trunk groups. The BLDU can be located with the Attendant Console or can stand alone.

See TL-I 30300-l 001 for a description of installation procedures.

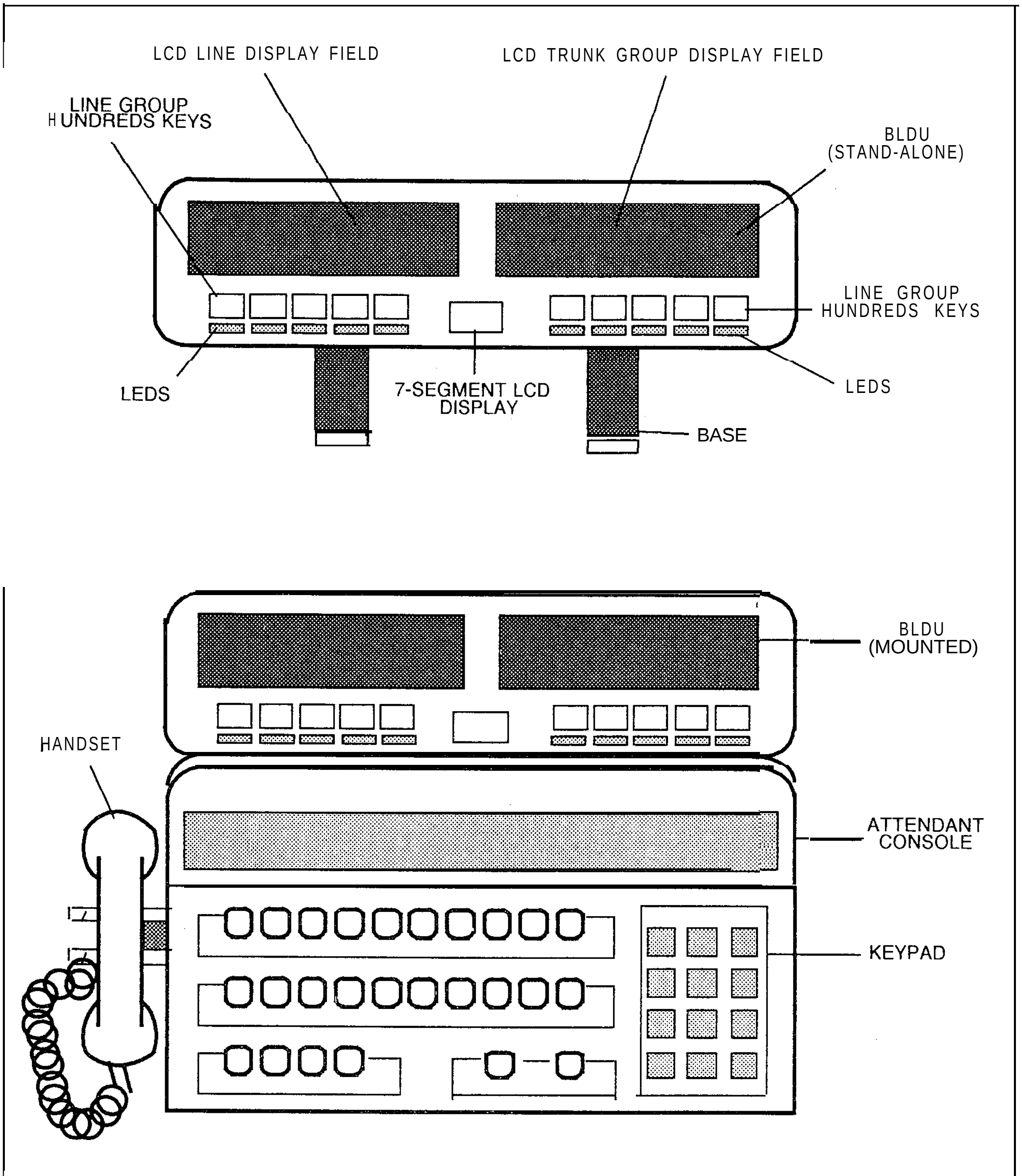


Figure 16.1 BLDU (Stand-Alone and Mounted)

The BLDU can be operated as a stand-alone unit or can be positioned on the Attendant Console so that the BLDU display is directly above the Attendant Console display (Figure 16.1). A detailed view of the BLDU showing the LCD line and trunk group fields and the LCD line group selected display is illustrated in (Figure 16.2). This view also shows the position of the ten line group selection keys and the small rectangular red LED indicators located immediately below each key. Refer to Table 16.1 for BLDU key functions.

The line and trunk group display fields use LCD elements to block out the appropriate characters. If a trunk group is restricted, an R is placed in view. Alternatively, if all trunks in a trunk group are busy, a B is visible. If one or more trunks within a trunk group are not busy (idle), the BLDU blanks the busy display for that trunk group.

The line field display shows the status of a group of 100 lines. Only the last two digits of the station numbers are shown so that any hundreds group can be displayed. The seven-segment line group display window shows which hundreds group is being displayed in the line field.

The line group hundreds keys on the BLDU that allow the selection of 10 different hundreds groups are defined in the data base. The LED indicates which pushbutton was operated. The remote control feature allows the attendant to select any hundreds group. The line group two-digit display will show this selection and the active LED will be extinguished.

NOTE: Activation displays of hundreds groups not programmed on one of the BLDU keys can be made only from an Attendant Console associated with the BLDU.

BLDU System Interface

16.2 The BLDU interfaces with the system via the ATT12 (Attendant Interface) FB-17208-A card located in a PCMUS slot. This same card supports the Attendant Console data link.

- This card provides the digital data link channel connection to the system.
- The data link is full-duplex, 1,200 baud, 20 ma current loop in serial form.
- The BLDU can connect to one port on an ATTI2 card, and an Attendant Console can connect to the other ATTI2 port. Up to four BLDUs can connect in series to the same ATTI2 port.

NOTE: The ATTI2 is described in Table 3.6.

Data link cable and power cable length limitations exist that limit the distance that a BLDU can be located from the system. Refer to TL-130300-1001 for cable length specifications and a description of BLDU installation procedures.

Data Link **16.3** Once in service, any interruptions of service at the BLDU, even momentarily unplugging the data link connector, will inititate an automatic recovery process. The maintenance terminal can also control the data link. The maintenance terminal can place the data link in service or out of service. If remote applications require, the standard type of data link used is compatible with readily available equipment designed for this application. Since this type of installation depends on the particular site, the specific design is performed on a per-site basis by special engineering personnel.

A BLDU only receives status data when its circuit is software selected.

Switches **16.4** BLDU switches include the following:

- DATA-LINK, loop-around switch. The last series-connected BLDU must have its loop-around switch set to the loop position.

- OFF-LINE switch. This switch takes the BLDU off-line and allows diagnostics resident in the BLDU to be executed. A modem can be used for remote operation of a BLDU. In this case, one data link channel is dedicated to the remote BLDU

Assign each BLDU a separate address (1 through 4). These addresses are assigned via two manually positioned switches in the BLDU. These switches are located under the BLDU cover. They are set to 00 to assign address 1 to a BLDU, 01 for BLDU address 2, 10 for BLDU address 3, and 11 for BLDU address 4. These switch-selectable addresses correlate to software addresses used by the system CPU during BLDU input/output routines. By altering the switch address, BLDUs can be interchanged in the system.

BLDU Power

16.5 The BLDU is powered from either a 24-VAC transformer connected to a 115 VAC power source or from a -48 VDC power supply. See TL-130300-1001 for power connections for the BLDU. TL-130100-1001 describes the operations that can be executed from the BLDU.

Initialization (I)

16.6 Once the connections to the BLDU have been completed, turning the power on activates the initialization process. The BLDU indicates this condition in the line group display; when the initialization process is complete, the trunk group status display is active and the station status display is ready for service. Either local key control or remote operation from an Attendant Console is required to establish the hundreds group to be displayed.

BLDUDisplay and Key Funtions

16.7 See Table 16.1 and Figure 16.2 for BLDU display fields and key functions.

Table **16.1 BLDU Display and Key Functions**

KEY OR DISPLAY	TYPE	DESCRIPTION
Top-Left Field	LCD	Indicates individual station activity.
Top-Right Field	LCD	Indicates trunk activity (B = Busy, R = Restricted).
Line Group (Seven-Segment Display)	LCD	Indicates selected hundreds group test status.
Line Group Selection Keys/Indicators	Key & LED	Selects hundreds groups. Indicates hundreds group selected by a key.
ON/OFF (Located on Left-Rear)	Toggle Switch	Controls power supplied to the BLDU.
Fuse Holder and Fuse (Located on Left-Rear)	W/3AG. 1.5A Fuse	Provides circuit protection.
Data Link Connections (2) (Located on Rear)	Ports	Primary port in and port out to provide series circuit for additional unrts up to a total of four.

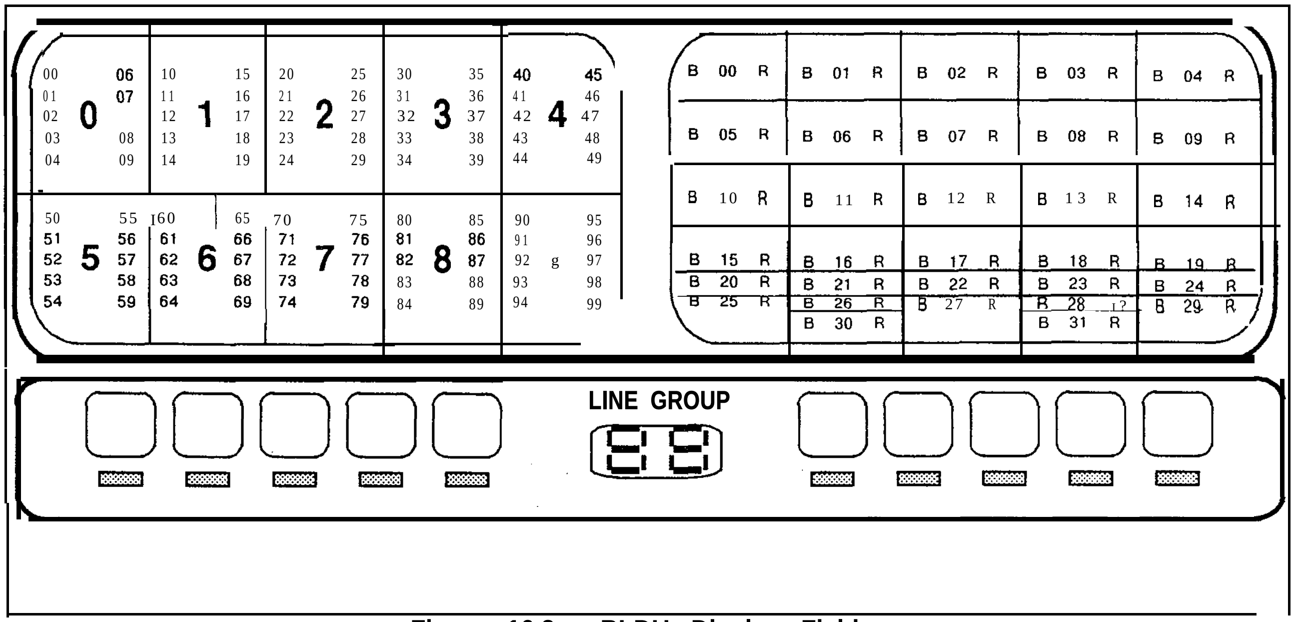


Figure 16.2 BLDU Display Fields

17.0 This section describes the various types of telephones that the OMNI SI supports.

17.1 The system supports all standard DP (Dial Pulse) telephones, DTMF (Dual Tone Multi-frequency) telephones, Display Telephones, and Stand-Alone Feature Telephones. All of these instruments interface to the system via standard tip and ring connections to the FB-17254-A PLCC (PCM Line Interface) card.

17.2 The system can interface with external equipment via standard lines and long-loop lines. The long-loop lines can be loaded or non-loaded. See TL-130300-1001 for line interfacing. Cards are described in section 3.0 of this document.

17.3 The Display Telephone (Figure 17.1), a compact, desk-top console unit, contains a 12-pushbutton DTMF keypad, a four-character liquid-crystal diode display, a handset, and a hookswitch assembly.

The Display Telephone interfaces with the system via a the FB-17254-A PLCC (PCM Line Interface)card located in a PCMUS slot. This card provides the analog voice channel (line circuit).

The telephone displays in its LCD either the calling or the called number. Customer needs determined the instrument's configuration. The configuration is defined in data base through the non-displayable class of service. Either entry directs the system to transmit DTMF signals over the tip and ring lines before applying ringing voltage. The calling number display provides the user with the station number of the calling party on internal calls and offers a blank display on outside calls. The called number display mode is most effectively used in a secretarial or screening type of operation. Calls diverted to the Display Telephone indicate the station that was called.

The Display Telephone can be powered by either a 24 VAC transformer or -48 VDC transformer. Power and tip and ring connections to the Display Telephone are described in TL-130300-1001.

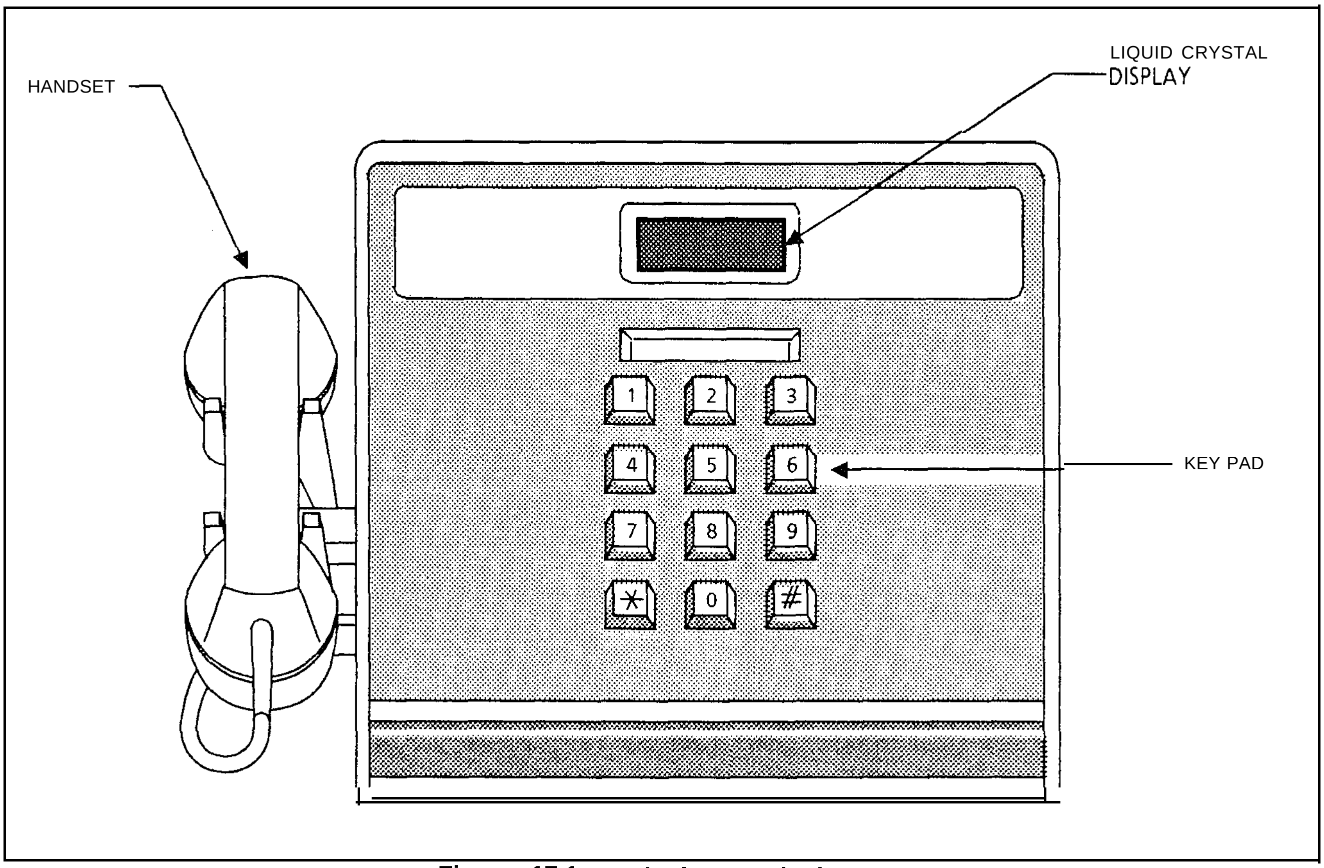


Figure 17.1 Display Telephone

Stand -Alone Featurephone

17.4 The Stand-Alone Featurephone, a compact microprocessor-controlled, desk-top console unit, contains a 12-pushbutton DTMF keypad, a 16- or 24-character liquid crystal diode display, a handset, a hookswitch assembly, and 8 or 16 feature pushbuttons. It can be connected to most PABX systems.

The Stand-Alone Featurephone interfaces with the system via the PLCC, located in a PCMUS. The PLCC card provides the analog voice channel (line circuit).

Power to the Stand-Alone Featurephone is provided by either a 24 VAC transformer or -48 VDC transformer, which is supplied by the system equipment cabinet.

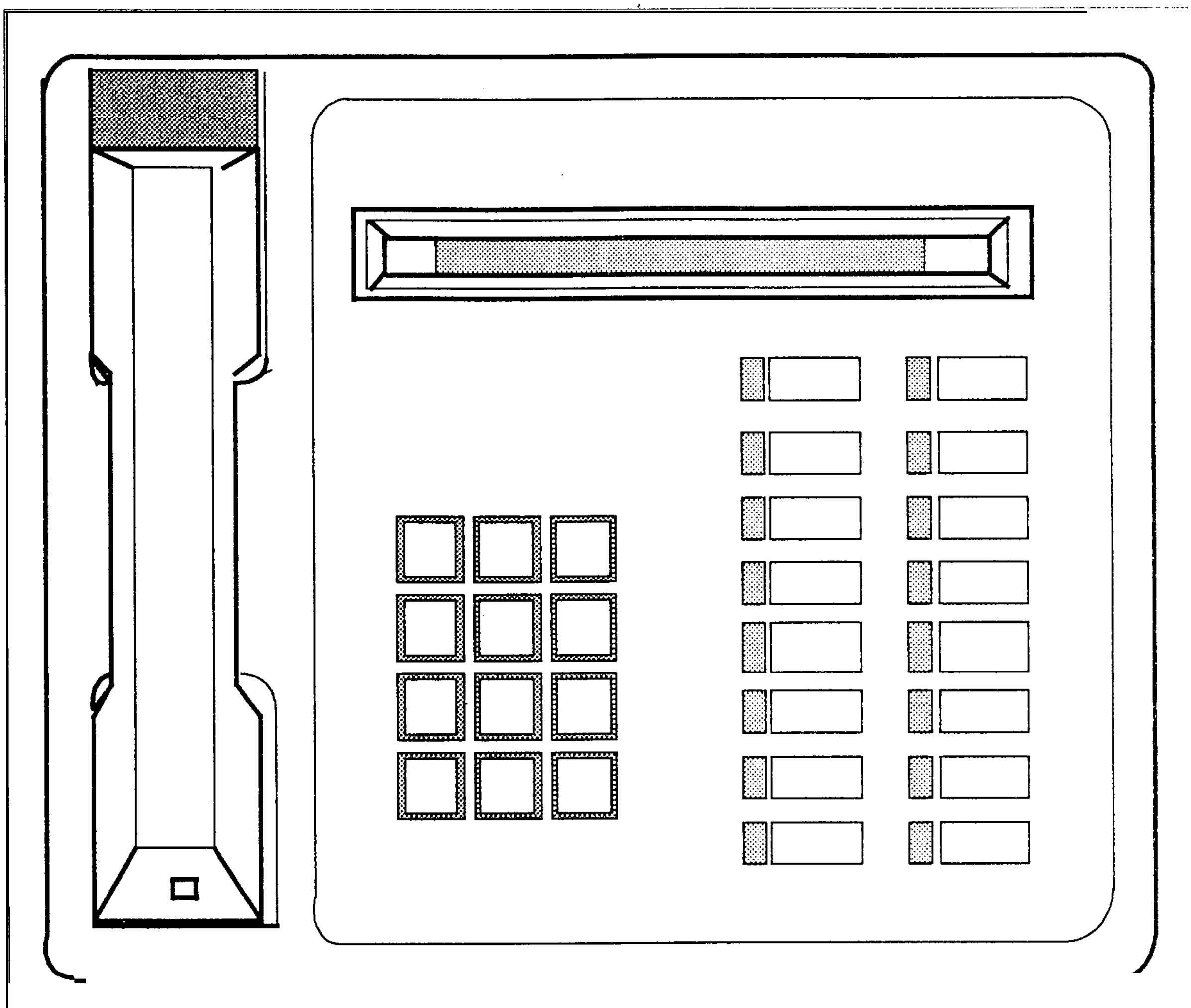


Figure 17.2 Stand-alone Featurephone

IFP (Integrated Featurephone)

17.5 The IFP, available in an analog version or a digital version, is a multi-featured telephone instrument. The IFP only operates when used with the OMNI PABX. The Featurephone is approximately 4.5 inches high, 10 inches deep, and 10 inches wide. It weighs approximately 3 pounds and comes in antique white with a wood-grain facemat.

IFPs are available with either 8 or 16 feature pushbuttons and associated LED indicators. An optional AOM (Add-On-Module), connecting directly to the IFP, provides an additional 30 feature pushbuttons and LEDs. Microprocessor control of the instrument allows the user to select the functions for any pushbutton/LED by simple programming steps. More than 47 features are available to choose from, which allows the instrument to be programmed to meet specific user needs.

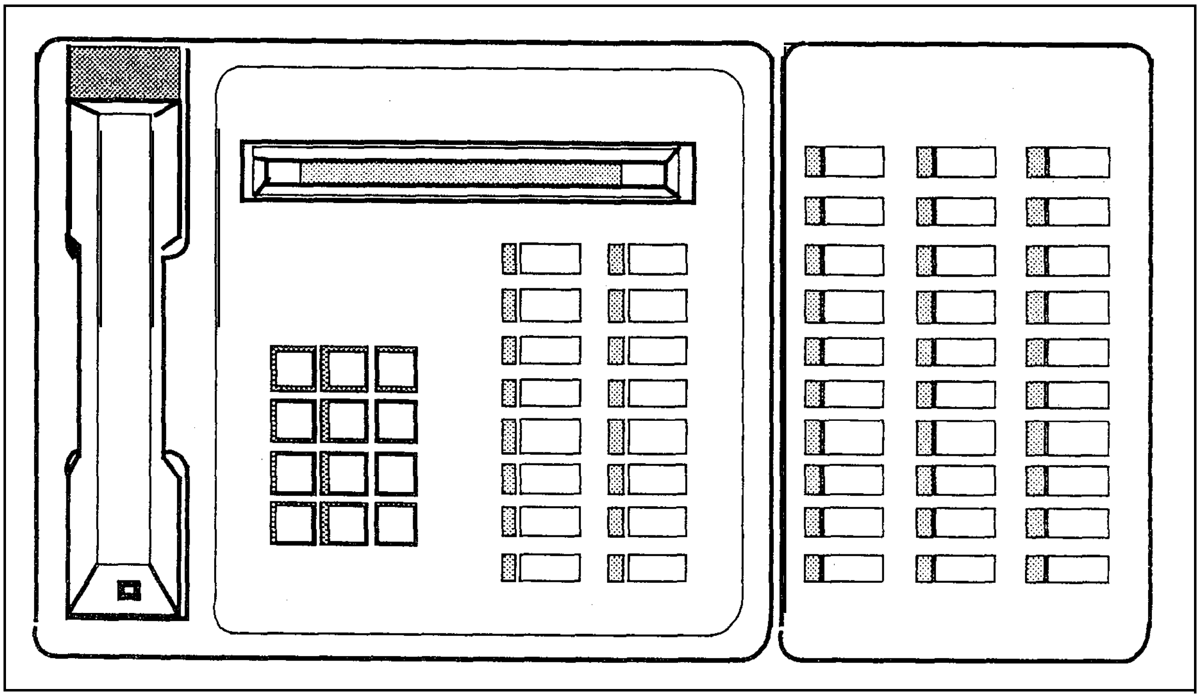


Figure 17.3 Integrated Featurephone with DSS

The IFP contains a loudspeaker and an amplifier and volume control. Whenever the monitor or optional speakerphone is active, the volume control and loudspeaker allow the user to listen to the line without lifting the handset off-hook. The loudspeaker produces audible alerting ringing tones of different cadences, with the volume and pitch being controlled by the IFP user's programming instructions. Audible beeps provide confirmation signals during programming operations.

An LCD, contained in the IFP, presents informative messages from the system to the user in addition to displaying user-activated functions. Uppercase alpha characters plus numerics and symbols are used, and underscoring is provided for prompting during programming. The 8-button instrument provides a 16-character display, and the larger housing of the 16-button instrument allows a 24-character display. An IFP with 16 pushbuttons and a 24-character display is shown in Figure 17.3 along with the optional DSS (Direct Station Selection) AOM (Add-On Module).

The IFP has access to most of the system services offered to a standard telephone instrument. The following COS (Class-of-Service) markings are inherent for the IFP and should not be assigned from the system:

- Calling Waiting Terminating
- Calling Number Display
- Called Number Display
- Call Forwarding

- The originating line selection preferences are based on a priority system:

  - If all three preferences are in effect, the prime line preference is the first choice.

  - If busy, the last line preference (selecting the last line used) is the second choice.

  - And if busy idle line preference is used, any remaining idle line present on the multi-line set is selected.

The Featurephone can support three different types of multiple-line appearances. They are standard, DSS (Direct Station Selection), and secondary appearance.

Standard Multiple Appearance

17.5.3 Standard multiple appearances associated with KTSs (Key Telephone Systems) have the following capabilities:

- All appearances can seize the idle line and use it in any normal fashion.

- Privacy is automatically invoked (although the prime control line can override the privacy), but the user can choose to defeat this function, thus allowing other appearances to join in the call.

- The line can be put on hold, which allows other appearances to pick up the call, or the line can be put on exclusive hold where only the invoker can retrieve the call.

DSS Appearances

17.5.4 DSS (Direct Station Selection) appearances have the following capabilities:

- A station with a DSS appearance of another station cannot control associated line seizing and holding activities.

- The DSS appearance allows line-status monitoring and single-pushbutton depression to call the appearance directly for call announce, intercom, or station-to-station calling.

- Directed call pickup can also be accomplished whenever the associated monitor LED indicates an unanswered incoming call. Outgoing calls cannot be made on a DSS appearance.

The following rules govern setting up DSS appearances. Only four IFP (Integrated Featurephones) can have the optional AOM (Add-On-Module) operating in the DSS mode.

- Up to 64 standard station instruments may have two DSS appearances on Featurephones (or AOMs). For additional information, see section 15.0.

- All Featurephone lines can have up to four line/DDS appearances on other IFPs (or AOMs)
- 16 IFP lines can have up to eight line/DSS appearances on other IFPs (or AOMs), and
- Any IFP can also have up to four secondary appearances of any of its control lines.

Table 17.1 lists the IFP system capacity.

Secondary Appearances

17.5.5 Secondary appearances have the following capabilities:

- Any Featurephone control line which does not have appearances on any other set can have multiple appearances (four maximum) at that Featurephone.
- When using a control line, the Featurephone user can place that line appearance on hold, select a secondary appearance line, and place another call.
- If another incoming call occurs at this point, the next idle secondary appearance line indicates the call and audible alerting is provided.
- The secondary appearances act like rotary lines appearing on a KTS instrument, except that the Featurephone does this with only one directory number.

Integrated Featurephone Capacity and Parameters

17.56 The following limits determine the amount of IFPs that the system can support:

- System limits
- Line limits
- Line card limits

The system supports up to 16 different Featurephone groups for intercom, call announce, etc. IFP capacities are listed by category in Table 17.1.

Table 17.1 Integrated Featurephone System Capacity Summary

CHARACTERISTIC	MAXIMUM NUMBER
System Limits	
Featurephones	128
Total appearances (line and DSS) (Refer to Note.)	512
Featurephone groups (i.e., intercom, call announcements, etc.)	15
Line Limit	
Total appearances of any one directory number (line and DSS) (Refer to Note.)	8
Line Card Limit (CIP, VCIP, and DVCIP)	
The following apply when considering all line appearances on all Featurephones connected to a line card of the types listed.	
The number of non-control appearances plus 5.47 times the number of control appearances.	≤ 236
Appearances of any one directory number	≤ 4
NOTE: Do not count secondary appearances in these totals.	

Analog Featurephone

17.6 The Analog Featurephone interfaces with the system via the PLCC and the CIP located in PCMUS slots. Place these cards in adjacent card slots for easier installation and maintenance.

- The PLCC (line circuit card) FB-17254-A provides the analog channel for both DTMF signaling and voice communications between the Analog Featurephone and the system. The analog channel connects to the system via tip and ring connections to one port of the line card.

NOTE: Once a PLCC is designated for Analog Featurephone use, only Analog Featurephones can go on that card.

- The CIP (Control Interface Processor) FB-17225-A provides the digital data link channel connection to the system and transports all control functions between the Analog Featurephone and the system.

When an Analog Featurephone goes off-hook, a digital message is sent to the system as a request for service message. Analog Featurephone station ringing initiated by the system sends the appropriate message to the Analog Featurephone. This directs the type of alerting cadence that the Analog Featurephone reproduces.

When dialing, the Analog Featurephone user hears the DTMF tones. The digital information is sent to the system via the data link and the system's DTMF receivers. These do not decode the information. If the Analog Featurephone user accesses a trunk requiring DTMF signaling, the system can cut through the analog voice path so that the dialed digital DTMF tones generated in the Analog Featurephone pass through the system to the trunk. Other applications may require the system to send either DTMF or DP signaling digits, which are an inherent system feature.

Power

17.6.1 Power for the Analog Featurephone comes from a 24 VAC transformer (connected to a local 115 VAC power source), a -48 VDC source (such as power pack/charger assembly), or the system power distribution point.

Digital Featurephone

17.7 The Digital Featurephone has many of the same features as the Analog Featurephone, such as user programmable functions. Unlike the Analog Featurephone the Digital Featurephone connects directly to the system by a digital data link. This connection incorporates the Digital Featurephone to the distributed microprocessor system, allowing for a continuous real-time control and response communications.

NOTE: A CODEC chip on the PLCC card converts analog to digital for the Analog Featurephone. The CODEC chip for the Digital Featurephone is in the phone itself.

- Microprocessor-controlled, desk-top telephone instrument.
- Available with either 8 or 16 pushbuttons and associated LEDs. The AOM (Add-On Module) provides 30 additional pushbuttons.)
- Available with 16- or 24- character liquid crystal display.
- Available in voice only or voice with data configurations. Digital Featurephones with data option include an RS-232-C jack in the rear of phone.
- 47 programmable voice features available to user

- Interfaces with the system via universal slots. For voice only, the VCIP FB-17235 provides the control link and voice channel interface between the system and digital phone (one tip and ring pair required). The DVCIP FB-17236, which is used for voice/data applications, circuit switches data via CD-100. The VPLC2 FB-17246-A, which is used for voice/data, switches voice and data via PD-200.

- Powered by either a 24 VAC transformer or -48 VDC

- System can support up to 127 Digital Featurephones

- Peripheral equipment Digital Featurephones voice and data

  -Interfaces with the system via universal slots. The VPLC/VPLC2 provides the voice/data mini-packet link interface between the system and the Digital Featurephones.

  -Powered by either a 24 VAC transformer or -48 VDC

  -Provides RS-232-C jack interface for data terminal/printer devices

  -The AOM (Add On Module) for the Digital Featurephone and Analog Featurephone are different (the AOM designed for the Analog Featurephone is not to be used with the Digital Featurephone as it could damage the phone). Check that the tag on the back of the instrument specifies RS-232-C.

- Peripheral equipment Digital Featurephone /APM

  -Microprocessor-controlled, desk-top telephone instrument

  -Available with either 8 or 16 pushbuttons and associated LEDs

  -AOM (Add On Module) provides 30 optional pushbuttons

  -Available in 16 or 24 character liquid crystal display

  -47 programmable features available to user (use feature code 41 for data button)

-Data button provides status indicator. Top LED is link light. Bottom LED is call light and indicates incoming calls.

-Supports simultaneous voice and data calls.

-Configured as single-line or multi-line sets.

-Equipped with internal APM.

-Interfaces with the system via universal slots. The VPLC/VPLC2 provides the voice/data mini-packet link interface between system and Digital Featurephone.

- Powered by either a 24 VAC transformer or -48 VDC supplied by an external source.

- Provides RS-232-C jack interface for data terminal/printer devices operating at speeds up to 19.2 Kbps.

The Digital Featurephone provides voice communication and, optionally, data communication using a single pair of telephone wires.

- Voice-only communication interfaces with the system via the FB-17235-A VCIP (Voice Control Interface Processor) card located in PCMUS.

- Voice and CD-100 (circuit switched data) communication interfaces with the system via the FB-17236-A DVCIP (Data Voice Control Interface Processor) card located in PCMUS.

- Voice and PD-200 (packet switched data) communication interfaces with the system via the FB-17246-A VPLC2 card (Voice Packet Line Card 2) located in PCMUS.

When the Digital Featurephone connects to the DVCIP, any data device connected to the Digital Featurephone inter-connects to the system's PCM bus. The data circuit in the Digital Featurephone provides an integral asynchronous terminal interface which interfaces data terminal equipment with the system. As an option, the Digital Featurephone can be strapped to interface data communication equipment instead.

When used with PD-200 data, the data circuit in the Digital Featurephone provides an internal APM (Asynchronous Packet Manager) to interface data terminal equipment with the system. Types of Digital Featurephone available are summarized in Table 17.2.

NOTE: Three types of mini-packets are transmitted between the DFP and SID (Software Identification). A single twisted pair multiplexes these mini-packets :

- Voice (transparent) mini-packets. These mini-packets carry only PCM (Pulse Code Modulation) voice samples between the CODEC in the telephone and the PCM bus in the switch.

- Voice control (non-transparent) mini-packets. These mini-packets carry instrument control messages between the ICP (Instrument Control Processing) software in the telephone and the telephone control software in the VCIP and DVCIP cards.

- Data (non-transparent mini-packets). These mini-packets carry user data between the integral packet manager and some other data end point (another Digital Featurephone or an APM). This end-to-end data transfer takes place only when a data call is established.

Table 17.2 Types of Digital Featurephones

PART NUMBER	CHARACTERISTIC	TYPE	CARD INTERFACE
HE-867831-WT14	8-button	Voice only	VCIP
HE-867834-WT14	8-button 16-character display Speaker telephone	Voice only	VCIP
HE-868832-WT14	16-button 24-character display Speaker telephone	Voice only	VCIP
HE-867832-WT14	8-button	Voice and data	DVCIP (circuit switched data) VPLC2 (packet switched data)
HE-867835-WT14	8-button 16-character display	Voice and data	DVCIP (circuit switched data) VPLC2 (packet switched data)
HE-867836-WT14	8-button 16-character display Speaker telephone	Voice and data	DVCIP (circuit switched data) VPLC2 (packet switched data)
HE-868834-WT14	16-button 24-character display Speaker telephone	Voice and data	DVCIP (circuit switched data) VPLC2 (packet switched data)

Figure 17.4 shows the Digital Featurephone and the VCIP (Voice Control Interface Processor) card, the DVCIP (Data Voice Control Interface Processor) card, and the VPLC2 (Voice Packet Line Card 2) card. The Digital Featurephone contains a CTU (Communications Terminal Unit) microprocessor chip, a MPRT (Mini-packet Receiver/Transmitter) chip, a VPAD (Voice/Packet Assembler-Disassembler) chip, a combination chip that combines the functions of a CODEC (Coder/Decoder) and a filter, and a TKDI (Timing and Keyboard/Display Interface) chip.

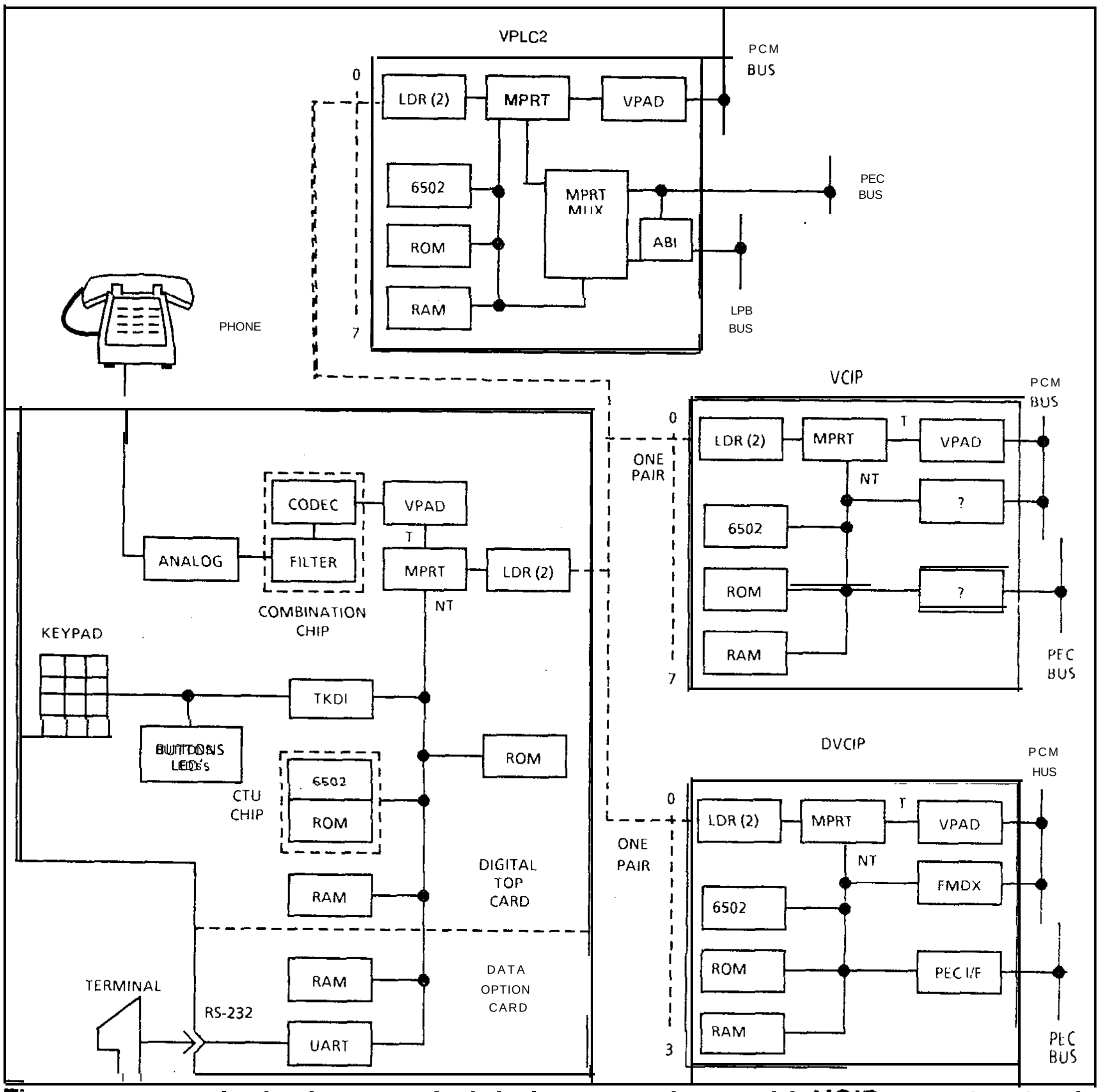


Figure 17.4 Block Diagram of Digital Featurephone with VCIP or DVCIP Card

The CTU has a 6502 microprocessor with 2 kilobytes of ROM (Read-Only Memory), 64 kilobytes of RAM (Random Access Memory), and DTMF tone generation circuitry based on ROM-stored patterns. The CTU controls all telephone functions and packet manager functions (with data option card equipped).

The VCIP card receives mini-packets from the eight digital phones connected to it by twisted paris. The VCIP card, by its eight MPRT chips, segregates the mini-packets into transparent and nontransparent mini-packets. Nontransparent mini-packets are presented to the 6502 microprocessor for processing. Transparent mini-packets are broken down into eight 64 Kbps PCM streams that are multiplexed into the allocated time slots on the PCM bus. Multiplexing each PCM stream to the correct PCM bus time slot is gated from the OMNI SI switch via three channel-select leads and a board-select lead on the PCM bus. For each new voice call, the OMNI SI switch reallocates a new time slot. This allocation remains constant for the duration of the call.

Converse functions performed by the VCIP card include initiating control mini-packets for transmission to the telephones and extracting PCM bit streams from the PCM bus. The VCIP card puts the PCM stream into transparent mini-packets by its PADs and sends them to the transparent ports of its MPRTs for transmission to the telephone. In both instances, the VCIP card determines which of the eight packet devices is the destination. The card presents both transparent and nontransparent mini-packets to the corresponding MPRT for multiplexing onto the destination twisted pair. The three channel-select leads on the PCM bus determine the destination device for transparent mini-packets. The destination device for nontransparent mini-packets is determined from the context currently being processed by the telephone control software. Each context corresponds to one of the eight telephones which, in turn, corresponds to one-of-eight channel identifications in the interface messages to or from the common control.

Data Voice Control interface Processor Card Operation

17.7.2 The Digital Featurephone can accommodate the CD-l 00 or the PD-200 data option. When equipped with either data option, the Digital Featurephone provides an RS-232-C connector that accepts a plug-in ASCII terminal and extra memory to take care of the data buffering. The data option card contains a UART (Universal Asynchronous Receiver/Transmitter) that terminates the RS-232-C interface data leads. The data phone supports simultaneous voice and data calls on a single twisted pair connection to the OMNI SI switch. Digital Featurephones equipped with the data option require a different interface card. The DVCIP (Data and Voice Control Interface Processor) is used in support of the CD-100 data option. The DVCIP card provides the capacity of four phones, each connected by an individual twisted pair. The DVCIP separates the transparent mini-packets received from the phone, converts them back to PCM form, and presents them to the PCM bus on four even channels (0, 2, 4, and 6) only for that card position. The nontransparent mini-packets are also separated, but into voice-control mini-packets and data mini-packets.

The voice-control software locally processes the voice-control mini-packets. This software is essentially identical to that used in the VCIP card. When messages associated with a voice call to or from the phone are processed by the control software, the interface messages to the OMNI SI switch are tagged with the even channel identification of the odd-even channel pair. Conversely, when messages associated with a data call to or from the phone are processed by the control software, the interface messages to the OMNI SI switch are tagged with the odd channel identification of the odd-even channel pair. This operation is true if the phone is in the data call setup mode or when a call terminating from the odd channel is being processed. In either case, the interface message communication with the OMNI SI switch is by the OMNI SI's PEC bus. User data mini-packets are framed for transmission on the OMNI SI's PCM bus, then are sent unchanged on the four odd channels (1, 3, 5, and 7) for that card position. Conversely, the PCM streams switched by the system to the four odd channels are interpreted as framed data. Finally, the DVCIP card removes the framing bits and transmits the mini-packets to the phone on the corresponding twisted pair.

The distinction between voice calls and data calls is transparent to the system. Each Digital Featurephone with the data option has two distinct adjacent equipment numbers and two distinct unrelated directory numbers, one for voice and one for data. The phone and its data option use adjacent channels on the same DVCIP card. The OMNI SI regards them as two individual Analog Featurephones, and makes no distinction between odd and even channel operation. If a voice phone user were to inadvertently dial a data transmission or vice-versa, the result would be transmitted as meaningless noise to either party.

Voice Packet Line Card Operation

17.7.3 When the PD-200 data option is used, the Digital Featurephone can be supported by either a VPLC or VPLC2 card (see paragraph 33.0).

Agent Instrument (PACET)

17.8 The Agent Instrument shown in Figure 17.5 is a compact microprocessor-controlled desk-top console unit. It is programmed to allow the device to operate with other Agent Instruments within a centralized pool of attendants (CAS Main application) or service agents (ACD application). Within this paragraph, the term agent describes both CAS attendants and ACD agents. The Agent Instrument contains compact electronics and all controls and indicators for monitoring and controlling calls processed through the system configured for CAS Main or ACD applications. Incoming calls can be answered, extended, put on hold, or completely released from the Agent Instrument by depressing various pushbuttons on the Agent Instrument. Outgoing calls can be initiated from the Agent Instrument.

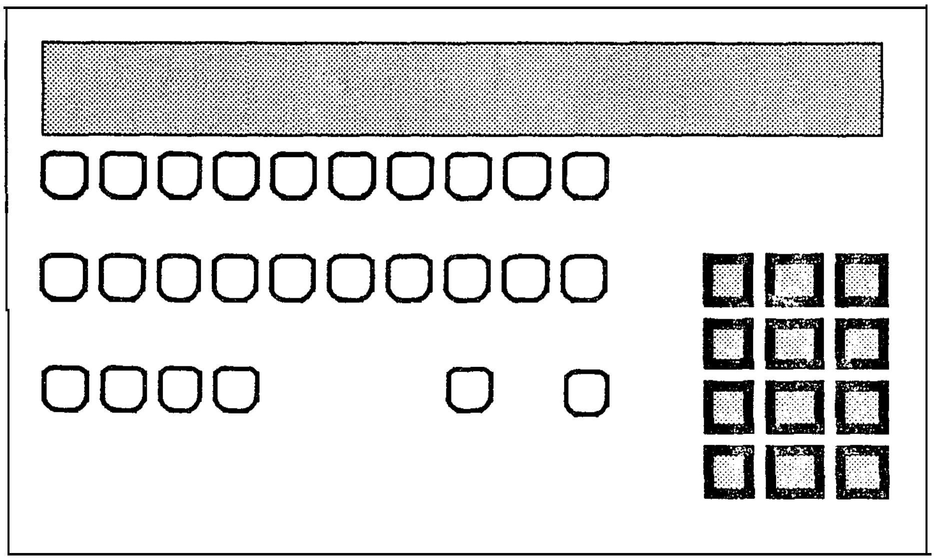


Figure 17.5 Agent Instrument

The Agent Instrument has headset or handset jacks located at the rear and on one side of the console. Supervisory headsetfhandset monitoring use the rear jacks, while agent's headset or handset use the side jacks. The agent's headset or handset must be plugged in for the Agent Instrument to operate in the on-line mode. When the user removes the headset or handset, the Agent Instrument defaults to the test diagnostic mode.

A 32-character alphanumeric display across the top of the Agent Instrument provides the agent with a visible description of the incoming call. (Refer to Figure 17.6. for a view of the Agent Instrument console.) The call description indication is present as long as the attendant is servicing the call and during agent recalls. The type-of-call display provides the agent with source and destination call information.

The Agent Instrument includes the following features:

- 26-pushbutton keyboard
- 12-key dial pad
- Call waiting level indicator
- Headset/handset jacks.
- Internal self-test diagnostics.
- 32-alphanumeric character LED display for displaying source of call information

The Agent Instruments connect by cable to the cabinet. Refer to TL-130300-1001 for a detailed description of Agent Instrument installation procedures and cable lengths.

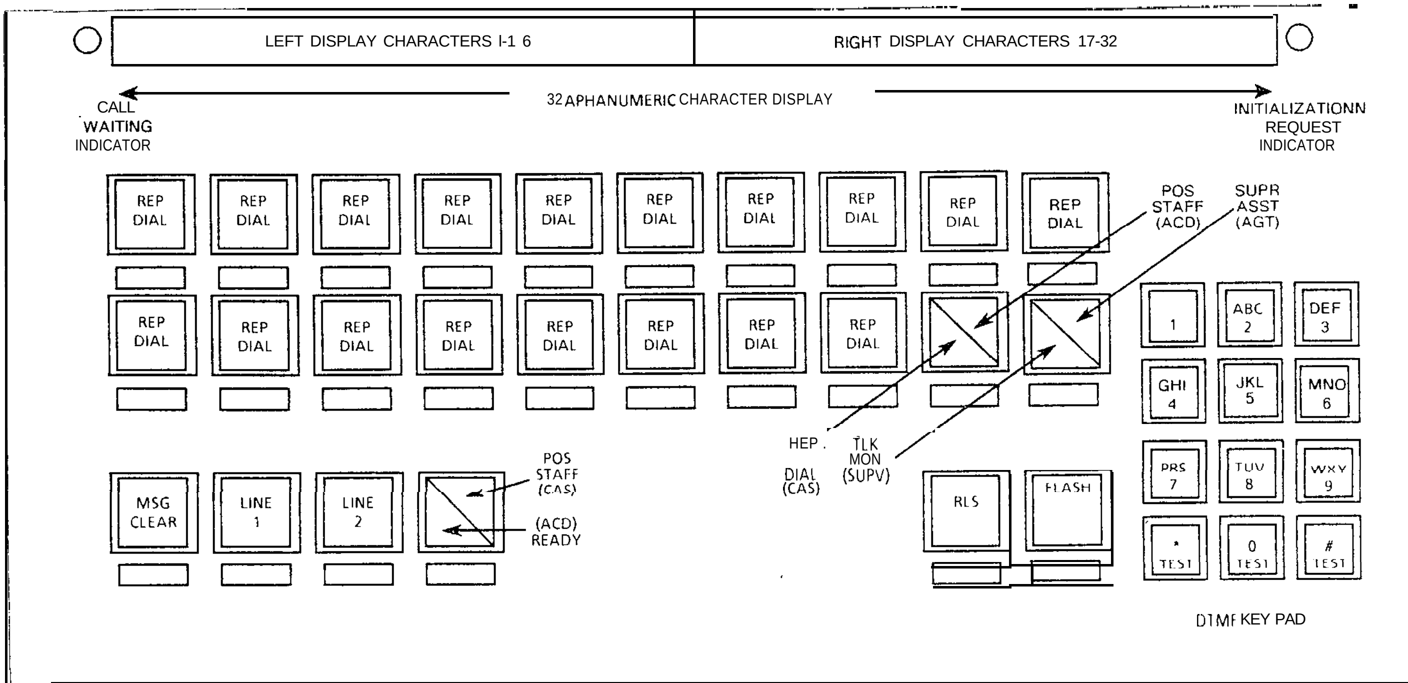


Figure 17.6 Agent Instrument Controls and Indicators

Agent Instrument Interface

17.8.1 One analog voice channel and one full-duplex, digital, data link channel allow each Agent Instrument to communicate with the system. An Agent Instrument can be optionally equipped with a second analog channel to support additional features as explained in Agent Instrument features in TL-130000-1001.

The analog channel handles both DTMF signaling and voice communication between the Agent Instrument and the system. The analog channels are cable connected to the system via two pairs of tip and ring connections to two circuits on an FB-17254-A line circuit card. The line circuit card, which the Agent Instrument connects to, is located in a universal card slot in the PEC equipment file.

The digital data link channel is cable-connected to the system via a dual-port FB-17209-A SIDML (Dual Modem Loop Interface-2) card. The Agent Instrument connects to this card, which is located in a universal card slot in the PEC equipment file. The data link is a full-duplex, 1,200-baud current loop.

See paragraph 3.6.17 for a description of the SIDML card and paragraph 3.6.12 for a description of the line circuit card. TL-1300300-l 001 describes Agent Instrument cabling.

Agent Instrument ControlL And initialization

17.8.2 Each Agent Instrument has five EPROM (Erasable Programmable Read-Only-Memory) chips. These EPROMs contain generic software that allows the Agent Instrument to function in the CAS Main, ACD, or supervisory modes. If the system includes two or more Agent Instruments, at least one of the instruments will be equipped for supervisory functions. There may be up to eight supervisory functions in a fully configured system.

During system startup, the system sends initialization data to all Agent Instruments for all repertory dial pushbuttons, source messages, special messages, and system dependent timing values. The mode of operation (CAS Main, ACD, or supervisory) is also set during initialization.

When an Agent Instrument receives a command from the system, it verifies the format, test, and checksum, and responds with a positive acknowledgment (ACK) if the command is correct, or with a negative acknowledgment (NAK) if the command is in error. A NAK causes the command in error to be retransmitted.

If the supervisor moves an agent between agent groups, the Agent Instrument for that agent is automatically reinitialized with the new agent group's repertory dial pushbutton assignments and mode of operation. This facilitates movement between CAS Main and/or ACD agent groups without the physical replacement for the agent's instrument or relocation of the agent.

An on-line, self-test diagnostic, resident in Agent Instrument PROM, routinely checks the generic software program. When the Agent Instrument is in the unstaffed mode, the user can request specific off-line, stand-alone maintenance diagnostics from the console keypad. These diagnostics test all the LED and alphanumeric displays, test the display console parameters, and show the contents of various internal registers.

Agent Instrument Power

17.8.3 Power to the Agentlinstrument comes form either a 24 VAC transformer connected to a local 115-AC power source or from VDC power supply. Battery input to the Agent Instrument should not exceed -54 VDC. Agent Instrument power connections are described in TL-130300-1001.

displayed. Manual dialing causes the digits, as entered via the keypad, to be displayed.

EMER (Emergency) Pushbutton - Depressing the EMER pushbutton signals the supervisor that an agent requiring supervisory action on an incoming line 1 call . At the supervisory position, this pushbutton allows the user to silence the alerting tone and display the agent ID and type of assistance required. (Refer to MUTE.) After depressed the EMER pushbutton, the EMER LED flashes at 120 IPM this acknowledges system acceptance and the LED extinguishes after the Line 1 call is released.

FLASH Pushbutton - The FLASH pushbutton allows the user to gain access to a branch location while processing a line 1 call, or the user can access standard PABX features on line 1 or 2. Depressing the FLASH pushbutton causes the FLASH LED to light steadily until instrument action is complete or the pushbutton is released.

FUSE - A fuse located on the left rear of the Agent Instrument provides circuit protection.

KEDU (KEY ENTRY DISPLAY UNIT)

18.0 The KEDU (Key Entry Display Unit) (Figure 18.1) provides control and display of Hotel/Motel features including

- Automatic wake-up service
- Room message waiting (indicator lamp) service
- Room restriction to prevent unauthorized calling
- Do not disturb service
- Room status (maid service)
- Message units metering
- Month/day/hour/minute display

A maximum of two KEDUs can connect to the system, with one assigned as the master with extended control over the features.

- The master KEDU can update the system's real-time clock, adjusting for daylight saving time changes.

- The master KEDU can output individual or total memory dumps of all features on the Hotel/Health Care printer for a hard-copy record.

- The master KEDU controls memory-clearing functions.

NOTE: Predetermined entries in the data base enable any of the six features, plus the time functions, to be allowed or disallowed for both KEDUs.

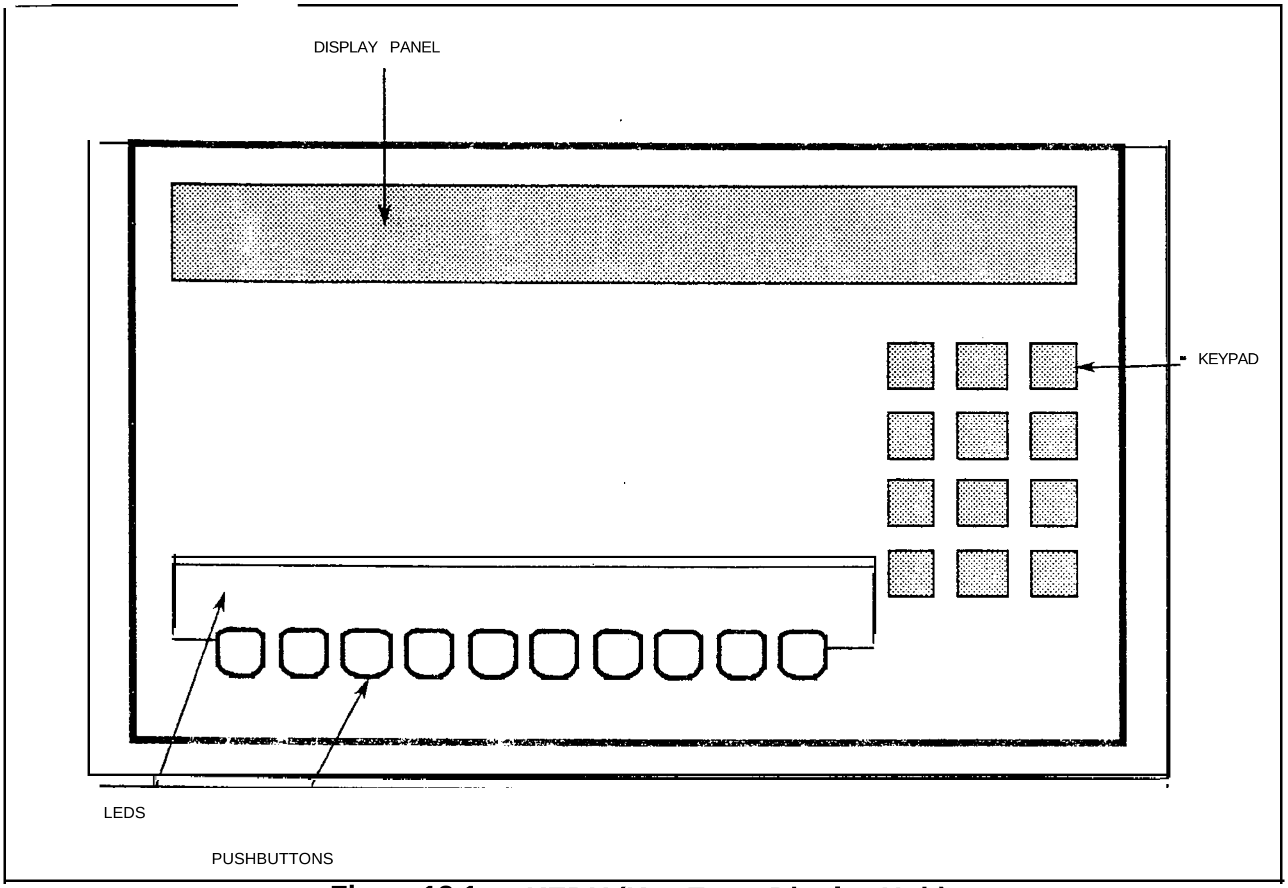


Figure 18.1 KEDU (Key Entry Display Unit)

KEDU/ System Interface

18.1 The KEDU interfaces with the system via the SIDML (SI Dual Modem and Current Loop) FB-17209-A card located in a PCMUS slot.

- The SIDML provides full-duplex, 110, 300 or 1,200 baud. 20ma, current loop data link. There are two in/out ports per card, and each port support a KEDU or a printer
- The system supports a maximum of two FB-17209-A SIDML cards and is limited to a maximum of two KEDUs or printers

NOTE: The SIDML card is described in paragraph 3.6.17

The KEDU is powered by either a remote -46 VDC system or a local power-pack assembly.

Controls, Indicators, and Displays

18.2 The KEDU controls, indicators, and displays (Figure 18.2) are listed below:

ALM RESET. (Alarm Reset). Allows the alarm to be reset.

CLEAR. Allows the removal of data or time information from the KEDU memory.

DATA-RESPONSE CODE. When DONT DIST, MESG METER, MESG WTG, ROOM RSTR, or ROOM STAT functions are requested, the LED lights, and the display indicates the respective data and/or response code.

DONT DIST. (Do Not Disturb). Allows a room number to be entered to block that room from all calls. An administrative station or the Attendant Console can activate, deactivate, or override the DO NOT DISTURB feature.

ENTER/XCUTE (Enter/Execute). Allows removal of data if depressed after the CLEAR pushbutton, or allows entry of new data after the data is entered via the KEYPAD, and executes the new data when depressed a second time for the following features:

WAKE UP	MESG WTG
MESG METER	ROOM STAT
ROOM RSTR	DONT DIST

HOUR-MINUTE. Lights the LED and indicates real or wake-up times, respectively, when the TIME or WAKE-UP function is requested.

KEYPAD. Allows the entry of the appropriate numerical data required for programming and executing the various functions.

MESG METER (Message Meter). Allows the entry of a room number for recording outgoing calls. Recording can be pegged per completed call or on a measured rate of service based on the trunk group. This information is programmed into the data base for the specific switch and location.

MESG WTG (Message Waiting). Allows the entry of a room number by flashing a lamp on the room telephone to advise a guest of a message waiting at the desk.

MONTH-DAY. Lights the LED and indicates the month (1 through 12) and day (1 through 31) when the TIME function is requested.

ROOM NUMBER. Lights the LED and indicates the room number (3 or 4 digits) when DON DIST, MESG METER, MESG WTG, ROOM RSTR, ROOM STAT, or WAKE-UP functions are being executed.

ROOM RSTR (Room Restriction). Allows the entry of a room number to be restricted from DOD (Direct Outside Dialing).

ROOM STAT (Room Status). Allows the status of a room to be displayed as follows: available or unavailable, occupied or unoccupied; if maid service is needed, in progress or completed.

SECURITY LOCK. Prevents unauthorized use of the KEDU.

TIME. Allows the display or update of month, day, hour, minute, and year.

WAKE-UP. Allows the entry of an automatic wake-up call for a specific room and time, up to 24 hours in advance.

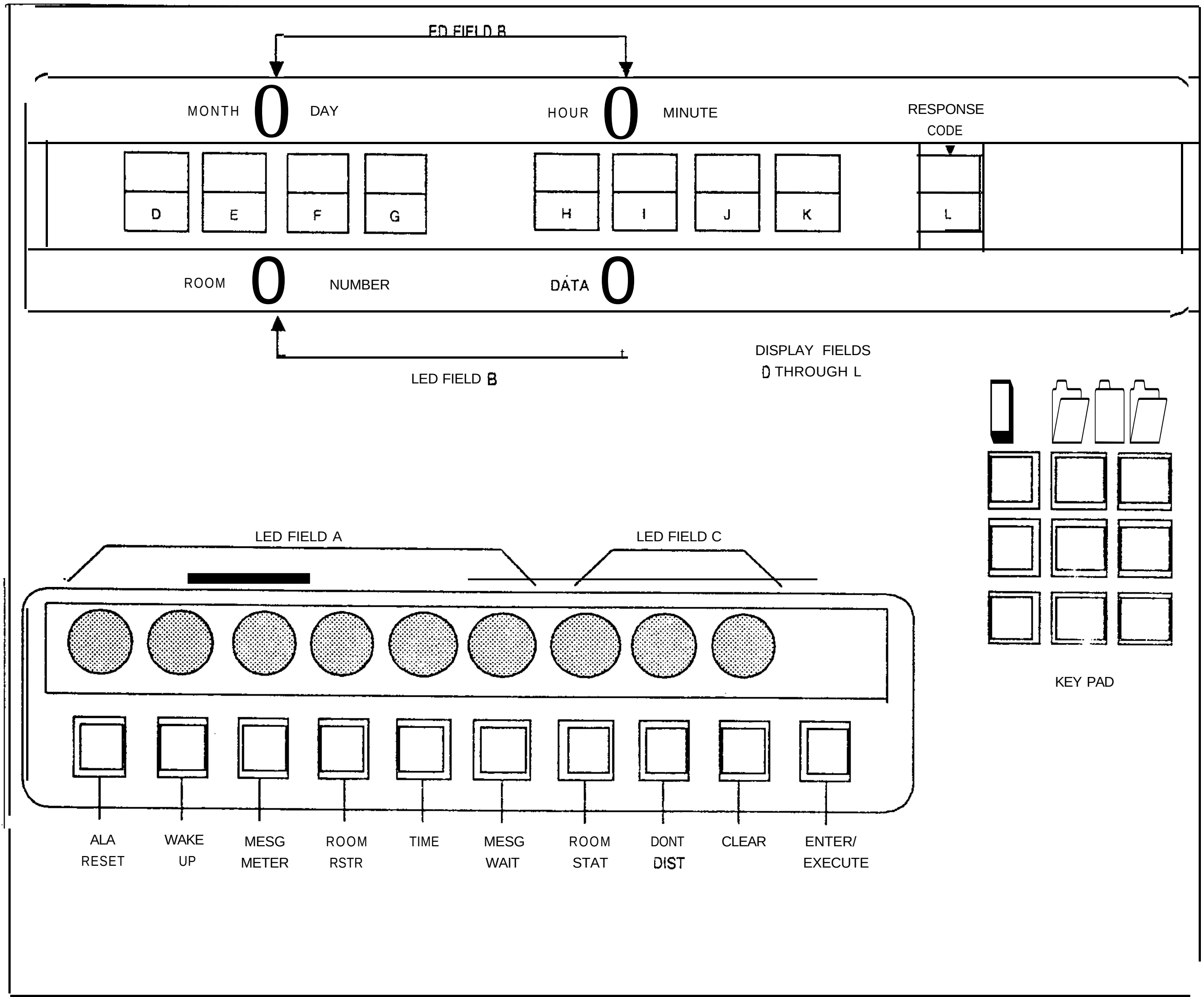


Figure 18.2 KEDU Button Function

THIS PAGE IS INTENTIONALLY LEFT BLANK.

H/HC (HOTEL/HEALTH CARE) PRINTER

19.0 The H/HC printer provides hard-copy records of the H/HC services. Any combination of outputs from the message metering, room status, do not disturb, room restriction, message waiting, and wake-up features can be obtained on the H/HC printer. Each of these six features can be selectively controlled to print whenever the feature is activated and/or deactivated (from any authorized control point), or whenever a print command is issued from a KEDU.

The system can support up to two H/HC printers. This provides hard copy for related services and activities . Examples of the printer applications are:

- Cleaning service operations where room status print-outs provide records of maid service activities (required, in progress, and complete).
- Front desk where a print-out is required to advise when maid service is finished and a room is available.
- Accounting department where a continuous audit record of all message metering is required. The checkout/billing area may only want this information on request (from a KEDU) for guest billing/receipt purposes.

Typically, a narrow 40-column printer manufactured by Axiom (Figure 19.1) is used for the H/HC printer applications; however, a wide 80-column format can also be used. Data properties of either 110 or 1,200 baud, odd/even parity, one- or two-stop bits, and seven- or eight-bit words can be specified. This allows almost any printer to be used as long as it is 20 ma.

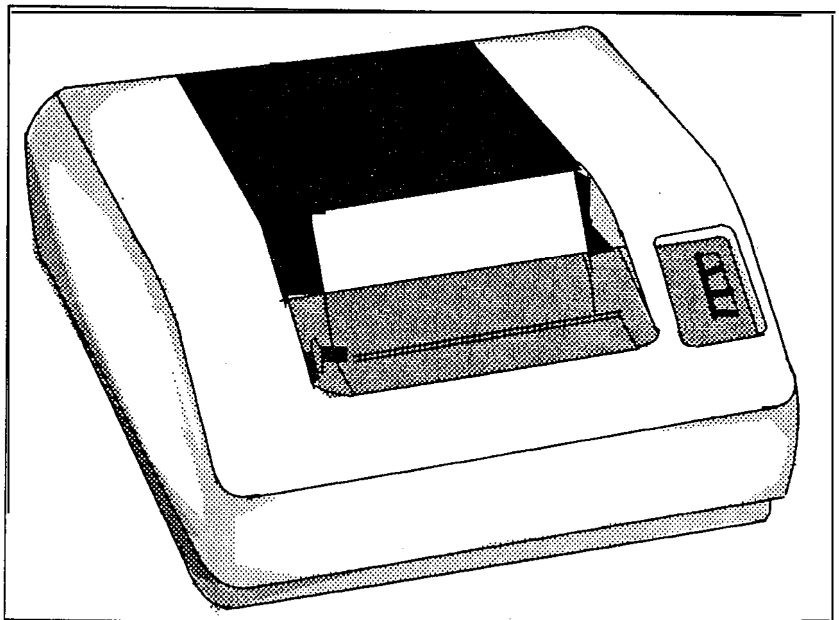


Figure 19.1 Axiom Printer Used with (or without) KEDU in Hotel/Health Care Applications

H/HC Printer/System Interface

19.1 The **printer interfaces** with the system via the SIDML card, located in a PCMUS slot.

- The card provides a 20ma transmit circuit to the H/HC printer.
- A single simplex data pair from the card supports the printer.
- The printer is slaved to the system.

NOTE: For description of the SIDML card, see paragraph 3.6.17.

Power for the H/HC printer typically comes from a local 115 VAC power outlet .

MAINTENANCE TERMINAL

20.0 The maintenance terminal records system traffic data and system faults and provides the means for performing data base changes and requesting maintenance procedures and diagnostics. The system uses a TTY printer or video display terminal as a maintenance terminal. The maintenance terminal can be equipped with a printer when a hard-copy print-out is required.

Maintenance Terminal/System Interface

20.1 The maintenance terminal interfaces with the system via one port of the FB-20992 NSDC (Narrow Serial Device Controller) card, located in slot 6 of the Get Started File.

NOTE: The NSDC card is described in paragraph 3.4.7

The NSDC card is a programmable serial device controller with two full-duplex ports, designated port 0 and port 1. Each NSDC port can be manually configured for a 20-ma, 12 volt, current loop or RS-232-C operation mode. Under software control, each NSDC port can be individually programmed for baud rate. The NSDC port is addressed via the CPU address bus and data is transferred between the maintenance terminal NSDC port and system CPU via the CPU data bus.

If the maintenance terminal is a TTY, the associated port on the NSDC card is configured for 20-ma current loop operation. If the maintenance terminal is a printer or equivalent EIA (Electronic Industries Association) compatible device, the associated port on the NSDC card is configured for partial RS-232-C operation. Modems are available for remote TTY or terminal operation.

Maintenance terminal power connections, cabling, and installation procedures are described in TL-130300-1001.

The following characteristics are needed to interface a CRT or TTY to the system:

- EIA RS-232-C communication
- ASCII character set.
- Full- or half-duplex operation
- Baud rate of 110, 300, or 1,200
- Display format of 24 by 80 characters
- Character mode transmission

Terminal Types and Mode of Operation

20.2 The user is required to identify the terminal(s) to be used for the ports of the interface card. The terminal types are entered into the data base via CPG or Recent Change. The terminal types are listed in Table 20.1.

Table 20.6 Terminal Types

TTY CODE 1	TERMINALS
0 = ADDS	ADDS Regent 60/Viewpoint 60
1 = DECV	DEC VT100 Series TDM 36
2 = XT300	GTE XT300
3 = LS31	Lear Siegler ADM31, ADM32
4 = TELE	Televideo 900 Series
5 = ASGN-1	Customer defined-1
6 = ASGN-2	Customer defined-2
7 = TTY	Hard copy terminals or “DUMB” CRTs

Other parameters defined in the data base tables that can be displayed or changed via Recent Change include the following:

- Transmission rates
- Security lock time-out
- Input time-out
- Echo
- Printer equipped

MDR (MESSAGE DETAIL RECORDING) HARDWARE

21.0 The MDR feature in the system is implemented via software and common system hardware. The software associated with the MDR feature uses common or shared memory.

MDR Printer/ System Interface

21.1 The MDR printer interfaces with the system via one port of the FB-20992 NSDC (Narrow Serial Device Controller) card located in slot 6 of the Get Started File.

NOTE: The NSDC card is described in paragraph 3.4.7.

The system can only support one NSDC card (two ports). The MDR output records can be transferred to external equipment through either serial port 0 or serial port 1. Manual switches on the NSDC card allow these two ports to be selected individually for RS-232-C mode or 20-ma current loop mode. Under software control, each port can be individually programmed for baud rate, character length, parity checking, synchronous or asynchronous operation, number of stop bits, and interrupt capability.

- Port 0
  - interfaces to a local or remotely located TTY or similar terminal which provides a hard copy of MDR records
  - controlling software output does not react to control signals from the TTY
  - MDR output is provided independently of the condition of the receiving TTY

With the use of port 0, the system works on a real-time basis (i.e., the call records are printed as they become available). Header data is output whenever a system start occurs, when a change in date or time occurs, every 64/32 call records or upon command from the administrative facilities (64 call records for single-line output or 32 for double-line output). The call record is output whenever an assembled call record is available and is in single format. The data base programmable output rates are 110, 300, and 1200 baud.

- Port 1
  - typically configured for RS-232-C operation
  - interfaces to a minicomputer or remote polling device
  - MDR output format is the GTE-modified EBCDIC (Extended Binary Coded Decimal Interchange Code) shown in Table 21.1.

NSDC card port 1 output is programmed to transmit data at speeds fast enough to avoid loss of data under the heaviest traffic conditions. The baud rates are 110, 300 and 1200. The baud rate and other options are programmable in the office-dependent data base.

When selecting the speed for the output device from MDR port 0 or MDR port 1, the amount of traffic being handled by the system must be considered. The MDR file is capable of buffering a fixed amount of call record blocks, after which the additional call information is completely lost. Therefore, the worst case peak traffic must be considered, along with work group and trunk group screen options. These factors will determine the probability of losing a call record block.

The MDR is capable of buffering 12 call record blocks (132 calls). As each call record is output by the MDR, an additional call record block can be generated and the data saved for output. If, however, 12 call record blocks are already in use at the time another call record block is generated, the data in the block will be lost for output to that port. The speed of the TTY or other output receiving device must therefore be chosen so that call record blocks are not lost. Cabling for the MDR is described in TL-130300-1001. Figure 21.1 shows a typical MDR configuration.

NOTE: When a call block is lost, no indication of the loss is provided, nor is there any means to recover the data.

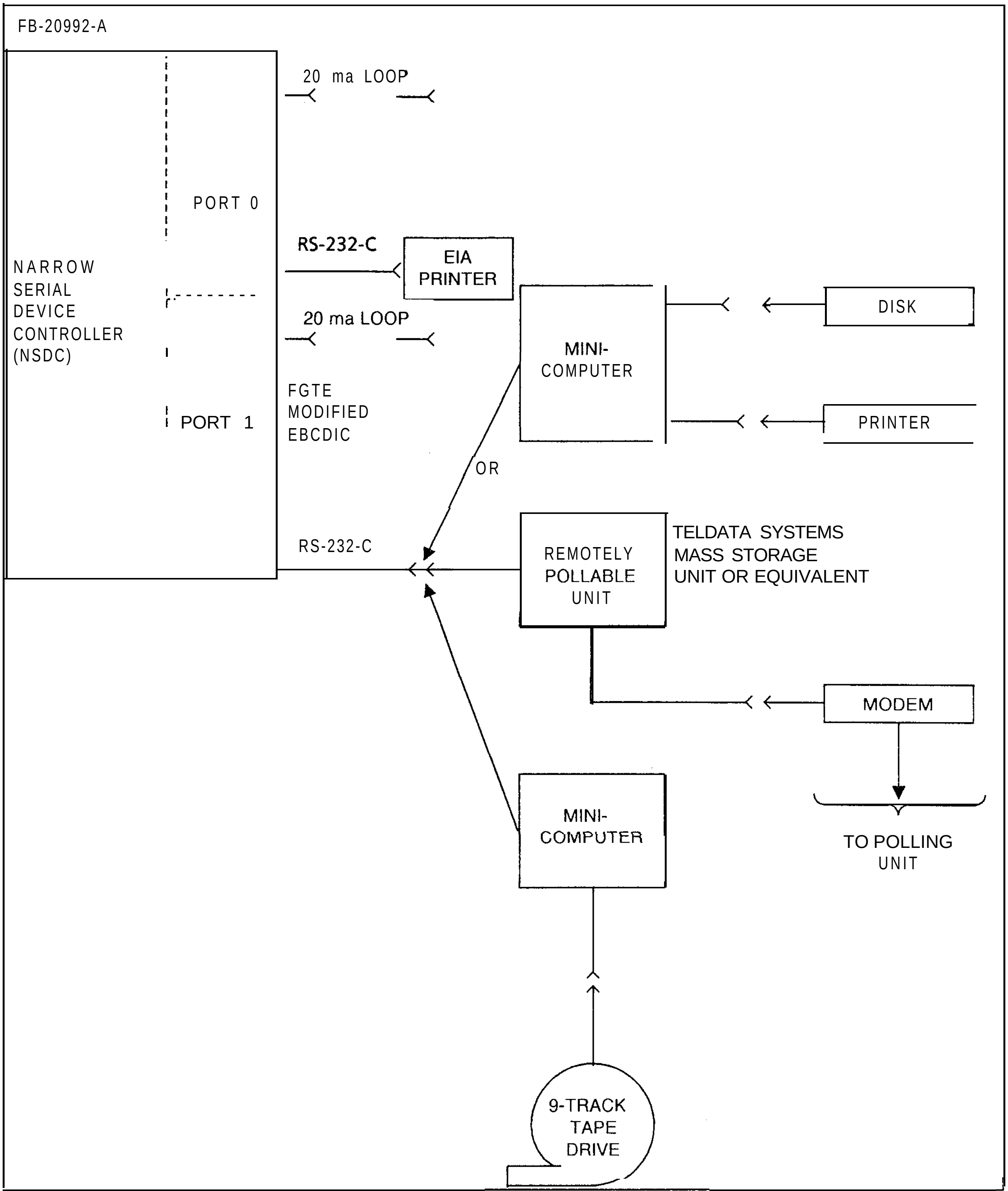


Figure 21.1 MDR Typical Configuration

Table 21.1 Fujitsu GTE-Modified EBCDIC

DECIMAL CHARACTER	HEXADECIMAL EQUIVALENT	BINARY EQUIVALENT	MDR USE
0	0	0000	The number zero (See Note 1)
1	1	0001	The number one
2	2	0010	The number two
3	3	0011	The number three
4	4	0100	The number four
5	5	0101	The number five
6	6	0110	The number six
7	7	0111	The number seven
8	8	1000	The number eight
9	9	1001	The number nine
10	A	1010	The number 10 (dialed digit zero) (See Note 1)
BLANK /11	B	1011	A blank space/identifying character/ the number 11 (See Note 2)
ALPHA a/l 2	C	1100	identifying character/the number 12 (See Note 2)
BETA b/13	D	1101	Identifying character/the number 13 (See Note 2)
14	E	1110	The number 14 (See Note 2)
15	F	1111	The number 15 (See Note 2)

NOTES:

1. The hexadecimal zero is always used for the decimal zero. He-xadecimal A is used as zero in calling and called numbers.

2. Hexadecimals A, E, and F are always used for the decimals 10, 14, and 1 5, respectively. Hexadecimals B, C, and D may have three different uses; however, their meaning is fully defined by the position occupied in the record.

The MDR feature is an option that provides the means for recording call information pertaining to incoming trunk calls, outgoing trunk calls, and trunk-to-trunk calls.

Hardware associated with the MDR feature provides the following:

- Interface with the program and data base memory
- Interface with the buffer memory
- Interface the input/output to external equipment

THIS PAGE IS INTENTIONALLY LEFT BLANK.

TYPE 200 DIGITAL SYSTEM TEST SET

22.0 The Type 200 System Test Set is a compact briefcase-type test instrument. The set provides a man-machine interface for performing system maintenance to include verifying microprocessor operation and isolating hardware/software malfunctions.

Type 200 Test Set/System Interface

22.4 The Type 200 Test Set interfaces with the system via the FB-17188 TPI2 (Test Panel Interface) card.

NOTES:

- The TPI2 card is described in paragraph 3.4.9.
- Use of the test set with a system having the PD-200 Data System is not recommended.

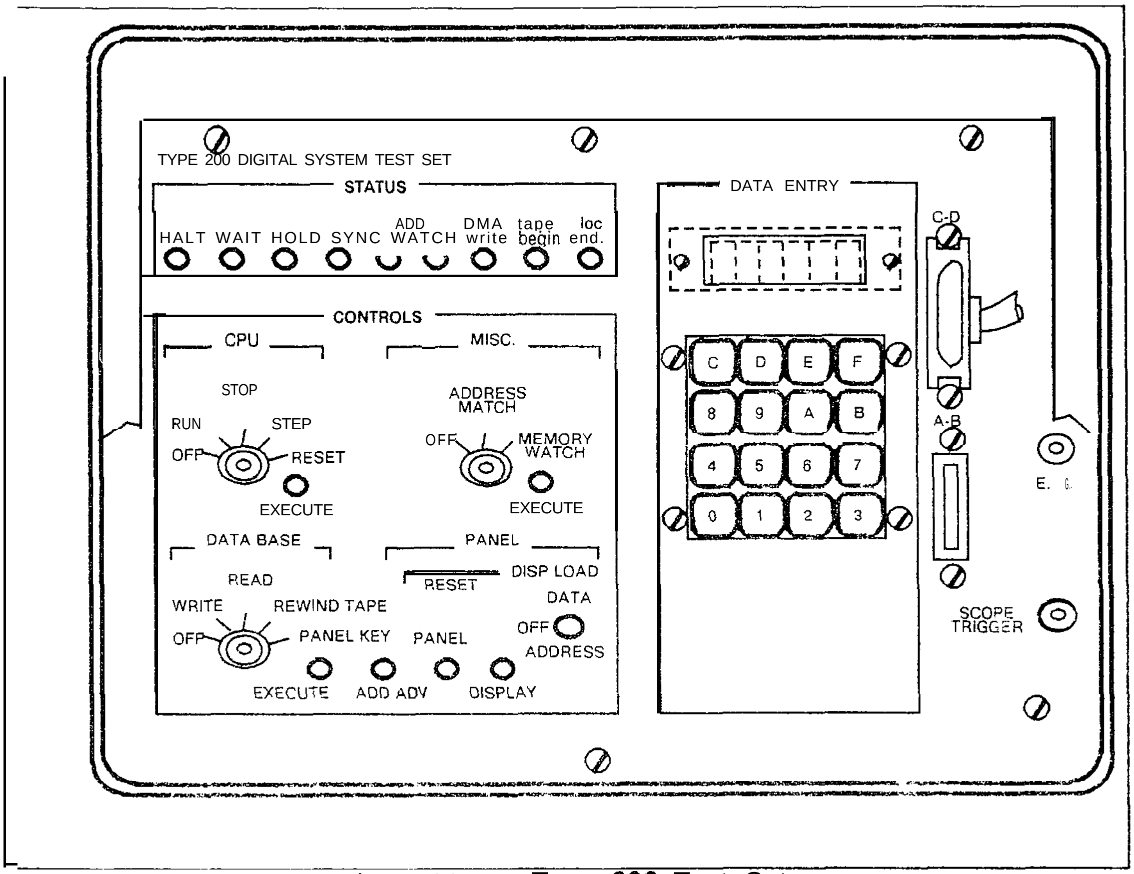


Figure 22.1 Type 200 Test Set

THIS PAGE IS INTENTIONALLY LEFT BLANK.

PAGING/DICTATION EQUIPMENT

23.0 Paging and dictation equipment is interfaced with the FB-17210-A PADIC (Public Address and Dictation) card, located in a PCMUS slot.

- The card has two dictation circuits and one paging circuit.
- The number of cards the system will support is unlimited when used for dictation.

NOTE: The PADIC card is described in paragraph 3.6.

The PADIC card allows for paging and/or code calling to four areas. These four areas are data base programmable into eleven different zones.

Interfacing to the customer-provided paging/dictation equipment is shown on the system functional block diagram in TL-130300-1001.

THIS PAGE IS INTENTIONALLY LEFT BLANK.

MUSIC-ON-HOLD

24.0 The music-on-hold hardware provides music to calling parties for the duration of time the call is placed in the hold queue.

The hardware (consisting of a music source, FM radio, packaged music source, etc.) interfaces with the system via the RABR and the PLCC cards, located in PCMUS slots.

- The FB-20996 RABR (Recorder Announcer Buildout Resistor) provides capacitive coupling to the PLCC tip and ring input.
- The FB-17254-A PLCC (line circuit card) provides data base programming and specifies the group, universal card slot position, and circuit number for the line interface card. This card is used to connect the system to the music source.

NOTE: The RABR and PLCC cards are described in section 3.0.

Calls placed on hold will be connected to the music source via this line circuit rather than to the normal quiet termination.

There is no limitation to the number of lines or trunks that can be connected to the music-on-hold feature.

NOTE: To prevent crosstalk, some systems require implementation of a line circuit that is marked out of service (see the line IS/OS table of the CPG).

THIS PAGE IS INTENTIONALLY LEFT BLANK.

RECORDER ANNOUNCER

25.0 The recorder announcer supports CAS Main or ACD operations. It announces to a caller "All Agents Busy" or an optional advertisement-type message. The system supports a maximum of five recorder announcers. Typically the fifth recorder announcer is used for night answer. A maximum of two recorder announcers per trunk group is allowed.

The recorder annnouncer interfaces with the system via a two-wire E&M trunk PCB, FB-17201, located in a PCMUS slot. An FB-20996 RABR (Recorder Announcer Buildout Resistor) card houses resistors to be connected in series with the tip and ring connections of the recorder announcer. The resistors inhibit conversation between subscribers connected to a particular announcement.

NOTE: An FGBS Automatic Intercept Recorder Model INT or equivalent may be used.

Hardware

25.1 To connect the recorder announcer to the system, the following hardware is required:

- One E&M trunk circuit interface per recorder announcer. The E&M trunk circuits are located on the FB-17201-PCM Two-Wire E&M Tie Trunk card. Each E&M trunk card contains four trunk circuits. The E&M trunk card is located in a universal card slot.

- One recorder-announcer device for each announcement used in the system. An FGBS Automatic Intercept Recorder Model INT or equivalent device may be used. The recorder-announcers are housed in an enclosure external to the system, i.e., miscellaneous equipment rack.

- An FB-20996 RABR card is mounted in a dedicated card slot on the back of the Power Monitor Transfer file, at slot position 8. This card contains eight resistors that support the interface to four recorder announcers. Two resistors, in each of four resistor pairs, are placed in series with the tip and ring connections to a recorder announcer. The resistors inhibit conversation between subscribers connected to the particular announcement.

NOTES:

- The RABR card also contains a coupling capacitor for the music-on-hold feature as described in this paragraph and in TL-130300-1001.

- There is a system limit of 64 trunks that can be cut in to a recorder announcer. Applicable trunks are all cut in at the start of the recording to prevent entering into the middle of a message.

A system configured for CAS Main/ACD can support five recorder announcers. A fully configured (operational CAS Main/ACD call) system can support two recorder announcers per trunk group. Often, only two recorder announcers (maximum) are used for CAS Main/ACD call functions in a system.

The recorder announcers are housed in an enclosure that is external to the centrally located cabinets, i.e., miscellaneous equipment rack, and are cable-connected to the system. , Recorder announcer cabling and installation procedures are described in TL-130300-1001.

CONFERENCE HARDWARE FOR SILENT MONITOR

26.0 For sites that have up to eight CAS Main/ACD supervisors, a minimum of one eight-party conference card is required to implement the silent monitor feature. One conference card will allow one supervisor to monitor any one agent. If more supervisors must monitor various agents concurrently, an additional conference card must be provided for each concurrent monitor. If nonblocking access to the feature is desired, assign the same number of conference cards as the number of supervisors in the system. The conference cards used for silent monitor are totally independent of the regular system conference feature and do not affect the two-conference-card-per-system limit.

THIS PAGE IS INTENTIONALLY LEFT BLANK.

THIS PAGE IS INTENTIONALLY LEFT BLANK.

RELEASE LINK TRUNKS

28.0 RLTs (Release Link Trunks) are used to minimize the number of interconnecting trunks required to connect a main system with branch systems for CAS operation and to provide the control and features required by this integral configuration. Both the CAS main and the branch must have the special capability of using RLTs. Additional hardware and revised data base distribution are required to incorporate RLTs into the system. However, RLTs are not tied up for the duration of the CAS call but are available when needed. They allow the interchange of control functions between two systems without the need for any additional interconnections and appear transparent to the users.

Incoming calls to the branch PABX are identified by differential audible tones heard by both end users and through seizure of the RLT associated with the main system. An audible ring is returned from the PABX station to the telephone instrument. The CO trunk-to-station connection at the branch is still set up and ringing continues at the station. The attendant “holds” the caller at the branch, and connects the call. The RLT releases to idle condition for RLT-2WGS (Wire Ground Start). (The branch end RLT-2WGS should provide open tip and ring leads to the main PABX end during the idle state).

When the call is answered, the branch end RLT-2WGS stops ringing within 100 msec. of receiving the answer signal. Ringing is tripped during both the silent and ringing intervals. The branch end PABX detects the answer signal from the RLT-2WGS and passes it to the connecting circuit (i.e., PABX-CO trunk) to cause CO ringing to trip and cut through the transmission path for talking. When an answer is detected, the branch PABX times for 300 msec. and then transmits, via the RLT-2WGS, through the appropriate call information tone to the CAS attendant, followed by cut-through of the transmission path between the calling party and the RLT-2WGS. After answer and until disconnect, the branch end RLT-2WGS provide up to 425 ohms resistance to ground on the tip lead, and -48 VDC loop battery through up to 425 ohms resistance on the ring lead.

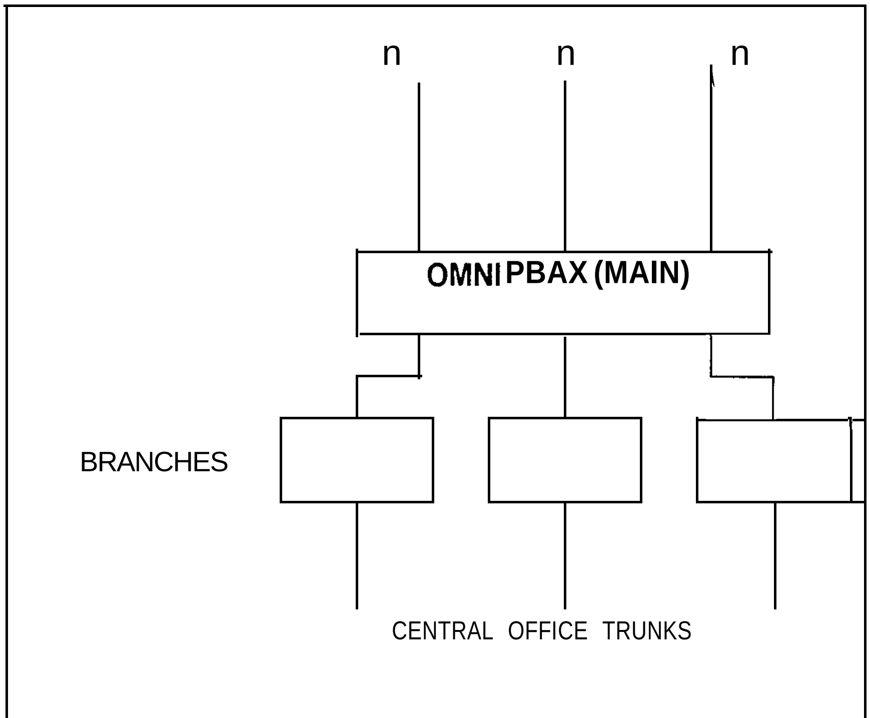


Figure 28.1 Release Link Trunks

If a call is directed to an idle station but the station is not answered within the selected time-out interval, the system recalls the attendant by means of a preselected idle RLT. On recalls, the appropriate audible signals are extended to the attendant to identify the type of call. The tone is heard only by the attendant because the system delays cut-through long enough to permit any required audible signal to be sent to the main location before the calling party is connected. If the station is not answered before the attendant answers, the seized PRLT signals the attendant with the appropriate audible tone. The call is then reconfigured to a three-way mode in which the local station continues to ring while the attendant and the trunk party converse over ringback tone. If the trunk party does not wish to wait for the selected line to answer, the attendant flashes off the original called number. The attendant then redirects the call by a second flash and redialing. Alternatively, the station can answer after the attendant has been recalled; the attendant then releases the PRLT and causes the trunk and line to remain connected while the PRLT is made available for reuse. The same procedure applies when the line rings for an additional recall period.

The functions of the PRLTs can be summarized as follows: The branch system uses the PRLT to send supervisory and information signals to the CAS Main, and receives supervisory and address signals from the CAS equipment to direct call switching at the branch location. Only the branch can seize the PRLT to initiate the sequence. The system can extend stations or tie trunks to outgoing trunks via the PRLT if this feature is required.

The system supports a maximum of 16 PRLTs.

Refer to the system functional block diagram found in TL-130300-1001. The PRLT trunks are connected into the system via an FB-17251 -A PRLT card. Each card can support four PRLTs.

Each PRLT can be dialed up for testing. The CAS branch system provides the cross-connecting arrangements so that telephones can be used for terminating the PRLTs at the branch location. This arrangement permits calls to be answered locally when the equipment at the main location fails, or the trunk facilities fail and no local console has been provided.

Each PRLT in the system has an assigned access number permitting the main location to make loop-around test calls to verify performance of both PRLT and main locations are being properly done. The main location accesses the branch system via a designated Tie trunk. The branch system is directed by dialing the PRLT access number to connect the Tie trunk to the selected PRLT for extension back to the main location. Two 100-msec bursts of tones signal the CAS attendant of a test call. Test calls can also be originated from the branch location by special stations class-marked in the data base as maintenance lines.

During the talking state, the RLT-2WGS circuit interprets a substained increase in the external DC resistance across the T and R leads from less than 1,300 ohms to greater than 20 kilohms for greater than 1 second as a disconnect indication. If the calling party abandons the call before answer supervision is received from the CAS Main location, the RLT-2WGS send a disconnect signal to the call distributor by opening the T and R leads.

The manner in which the call is completed varies according to the status of the local line selected. If the call is successfully terminated, immediate ringback tone is transmitted to the CAS Main and the CAS attendant operates the RELEASE pushbutton. The RLT interprets this as an on-hook condition, the system then reconfigures the call and connects the incoming trunk to the destination station.

The four-wire E&M RLT provides audio frequency transmission via the T, R and T1, R1 leads, while supervisory signals are transmitted over the M lead (outgoing) and the E lead (incoming). The T, R and T1, R1 leads do not carry DC signals. The CAS Branch PABX trunk circuit sends a seizure by replacing the ground potential on the M lead by -48 VDC battery potential. The seizure is received at the CAS Main location as a change of state on the E lead from open to ground. The E lead at one end of the interconnecting facility always indicates the state of the M lead at the other end of the facility. Disconnect at either end of the facility is signaled by connecting ground to the M lead at the disconnecting end. The other end of the facility receives the disconnect as a change of state on the E lead from ground to open. The four-wire E&M RLT reflects an idle state at the branch PABX as a ground on the M lead, causing the CAS Main to receive an open on the E lead. The CAS Branch PABX reflects an outgoing seizure condition by applying -48 VDC battery to the M lead (removing ground).

After sending a seizure, the branch PABX should interpret the appearance of ground on the E lead as answer supervision indication from the main location. Upon receipt of answer supervision, the branch PABX performs the following functions:

- Return answer supervision (trips ringing) to the CO trunk (if involved).

- Time for 300 msec, followed by transmittal of the appropriate call information tone to the centralized attendant, followed by cut-through of the voice transmission path between the calling party (i.e., CO trunk) and the attendant. During the talking state, the branch PABX should maintain -48 VDC battery on the M lead and continue to monitor ground on the E lead. During the talking state, the branch PABX interprets a momentary open of the E lead (i.e., 300 to 1,000 msec) as a flash. During the talking state, the branch PABX interprets an E lead open of greater than 1 second as a disconnect signal. If the calling party abandons the call before the attendant answers the call, the branch PABX replaces the -48 VDC battery on the M lead with ground, thereby transmitting a disconnect indication to the main location.

THIS PAGE, IS INTENTIONALLY LEFT BLANK.

T1-TYPE DIGITAL TRUNK LINK

29.0 The T1 digital trunk link allows the system to receive and transmit time-division-multiplexed T1 carrier data to and from a channel bank at the far end. A T1 digital data link can support 24 channels of multiplexed Tl-type data. The two-line (transmit and receive) T1 trunk link provides the same traffic capabilities as 24 individual analog trunk lines. The T1 digital trunk link (Figure 29.1) consists of the T1 interface cards at the system end, a channel bank at the far end, and a T1 span in between.

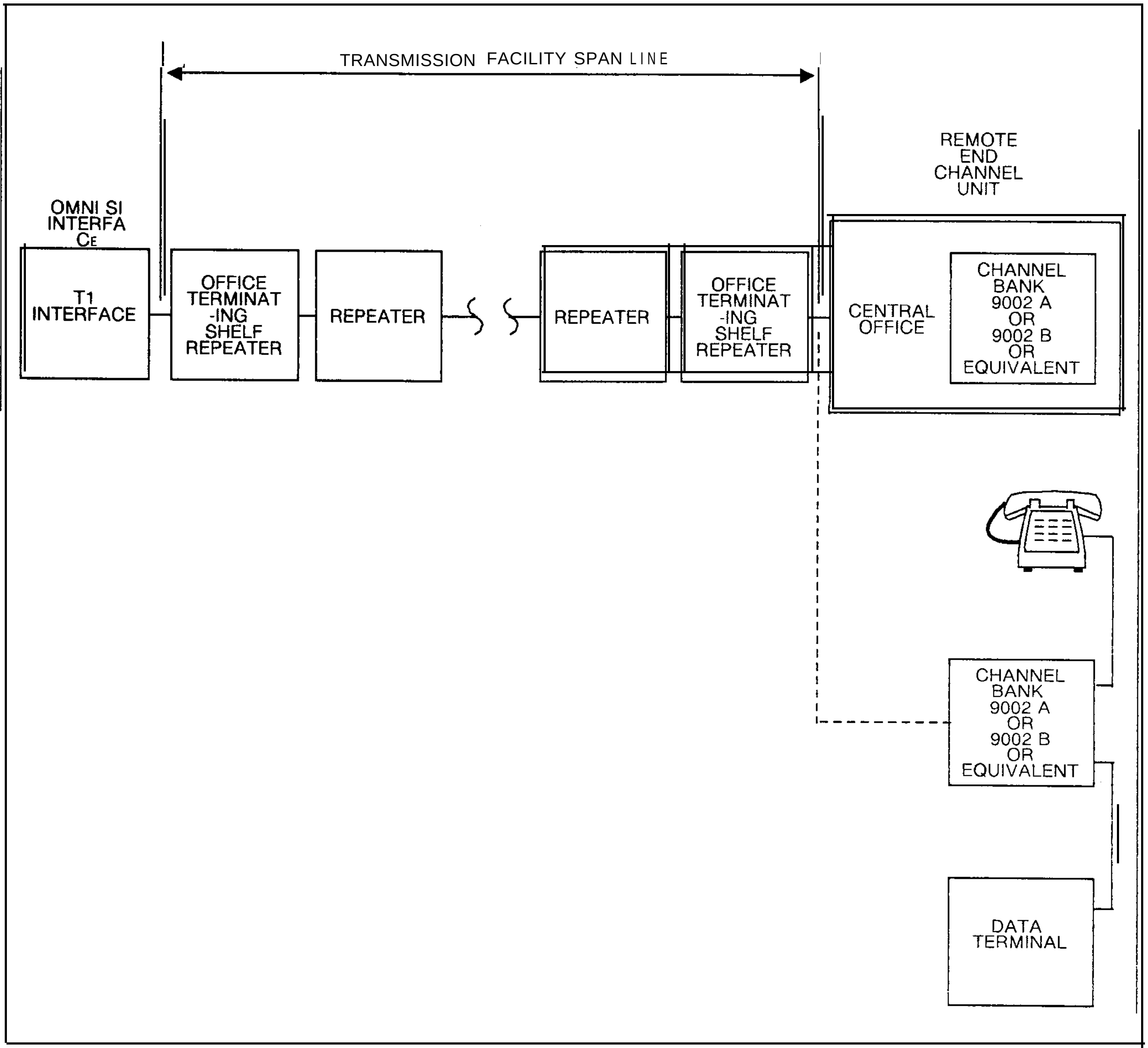


Figure 29.1 Tl-Type Digital Trunk Link

The T1 interface is compatible with WECO D2 and D3 signaling format channel banks or Siemens Transmissions Systems 9002A (24) or 9002B (48) channel banks at the far end. The far-end channel bank interfaces with loop-start, ground-start, or E&M trunks that can be connected to any standard telephone and most data terminals, to a system, or to a CO that supports T1-type communications. Two systems can be connected via a T1 link without the use of a channel bank at either end. The T1 interface is joined directly to the local terminating shelf repeater and does not require a channel bank or A/D to D/A converters in the system end (Figure 29.1).

Supervisory signaling information is transmitted every sixth frame. During the signaling frame, the least-significant digit is borrowed from the voice information coding and used to indicate the signaling condition of each channel. Alternate signaling frames are designated as A and B, respectively, called A and B signaling. When a customer-related interface is used, such as a FX ground start or FX loop start, the A signaling frame provides on-hook/off-hook information. The B signaling frame provides ringing information and special conditions.

CAUTION

Because the removal of a loop-start channel at the CO is not detected by the Tl-type interface it is recommended that the use of loop-stsrt channels for T1 interface be avoided.

The removal of a loop-start channel unit at the CO is not detected by the Tl-type interface cards. Therefore, it is recommended that use of the loop-start channel interface be avoided for T1 applications. Removal of a ground-start channel unit at the CO causes the GD (Ground-Detected) sense point, bit 7, in the FX trunk sense word to become true, making the associated trunk busy to the system. When this occurs, the associated trunk-busy lamp on the Attendant Console lights.

Removal of an E&M or FX ground-start loop dial channel unit at the far end causes an incoming seizure to the system. If the system data base is configured for the trunk circuit as a ringdown to the attendant, removal of the far-end channel unit causes an incoming call to the attendant's loop. If a channel unit at the CO channel is equipped with a busy key, operation of the busy key is equivalent to removing the channel unit from service. The system can support one 24-channel, Tl-type digital interface. If T1 access is used in the system, a minimum of 12 terminations per file are assigned for Tl-type carrier access. The physical location and component height of the T1 circuit cards restrict the use of three universal card slots. T1 software addressing restricts the use of universal card slots 4, 5, and 6. These card slots normally support 4 turnks each (12 total terminations). Additional T1-type terminations (up to 24 per file) are assigned in groups of 4. When 24 (maximum) T1 terminations are assigned, the use of two more universal card slots is also restricted because of software addressing limitations.

T1 Span

29.1 The T1 span is defined as the transmission facility between the T1 interface cards in the system and the channel bank at the far end. The T1 span (Figure 29.1) includes the following:

- An office terminating shelf repeater connected to the T1 interface in the system.
- An office terminating shelf repeater connected to the channel bank at the far end.
- Regenerative pulse repeaters strategically located at critical distances within the T1 span.
- A two-wire or four-wire interconnecting exchange cable between all units in the Tl span. Full duplex operation requires a four-wire exchange cable in the T1 span.

The T1 span supports T1 carrier transmission. A T1-type carrier is a 24-channel, time-division-multiplexed, PCM, method of transmitting digital or analog data and signaling data between two devices using standard non-loaded telephone pairs.

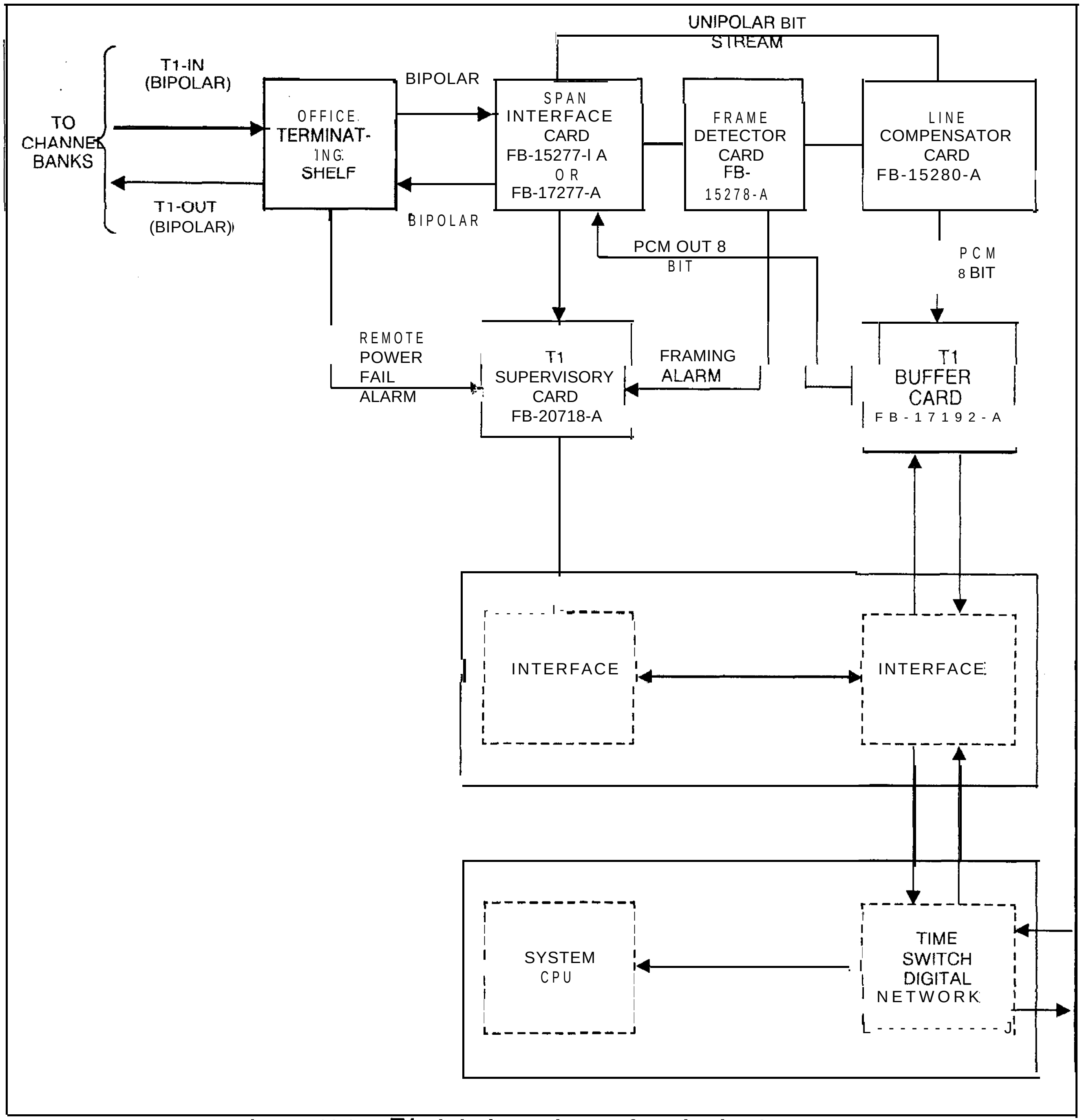


Figure 29.2 T1 Digital Trunk Interface in the System

NOTES:

- Critical distances between units in the T1 span are defined in TL-130300-1001.
- Refer to 342-922-106 for a description of the 9104A Channel Repeater Line Equipment.

T1 Interface

29.2 The Tl interface (Figure 29.2) is comparable to the CO-type Siemens Transmissions Systems 91258 FX Channel Unit or equivalent. Data base programming determines whether the T1 interface responds to ground-start or loop-start signaling. The interface can be either incoming, outgoing, or two-way operation. Since the digital interface is transparent, remote channel units can be either two-wire or four-wire.

NOTE: When the serving end offices use digital switching, synchronization of the Tl-type lines can be under end office control where the system functions as a slave. (Refer to T1-type Digital Trunk (paragraph 29.5) for the optional configuration). A/D (Analog to Digital) and D/A (Digital to Analog) conversions must be made at the other system when connecting a system configured for Tl carrier operation to another system equipped for Tl-type carrier lines and routing the data onto an analog trunk or line at the other system.

Refer to Figure 29.1 for incoming Tl data. The T1 interface receives 1.544-MHz, bipolar, T1 -type carrier information from the shelf repeater at the system end of the T1 span and performs the following functions.

- Converts the serial bipolar T1 incoming frames of data into unipolar serial format.
- Detects framing synchronization for the 24 time-division-multiplexed channels.
- Detects and removes control signaling and supervisory information from the PCM signal and forwards these signals to the system.
- Detects incoming data alarm conditions and forwards alarm signals to the system.
- Converts the serial unipolar data to eight-bit PCM data bytes that represent voice and tone samples.
- Buffers the eight-bit PCM voice and tone sample data bytes to circuits that transfer the samples at a 49.408-MHz rate to the systems time-switch network.

For outgoing Tl data, the T1 interface receives 49.408 MHz, eight-bit PCM data bytes from the system's time-switch network and performs the following functions:

- Provides buffering for the outgoing PCM data into the Tl interface.
- Adds signaling and supervisory information into the outgoing data.

- Converts and synchronizes the eight-bit PCM data bytes, representing voice and tone samples, into serial bipolar frames of data.

- Outputs the serial bipolar frames of data onto the Tl connection to the repeater located at the system end of the T1 span. The T1 interface uses time-division-mulitplexing methods to insert each frame of output data synchronously into its assigned channel time slot on the multiplexed 24-channel output to the T1 repeater.

The T1 interface in the system can operate in the master or slave clock mode. In the master mode, the system generates the master T1 clock signals for data transmission on the T1 digital trunk lines. In the slave mode, the clock in the system T1 interface is synchronized to a master clock contained in the T1 data stream received from another system device.

T1 Interface Cards

29.3 Each T1 type digital interface in the system comprises five cards which are installed into the system. When these cards are installed, three universal card slots in the associated equipment file cannot be used for other applications. The T1 interface cards are defined in Table 29.1.

Table 29.1 System T1 Interface Cards

CARDPARTNUMBER	CARD NAME	ACRONYM
FB-17277-A	Span Interface (for Master Operation)	SIL
FB-15277-I A	Span Interface (for Slave Operation)	SIL
FB-15278-A	Frame Detector	FDC
FB-15280-A	Line Compensator	LCM
FB-17192	T1 Buffer Card	T1B2
FB-20718-1 A	T1 -Type Supervisory Card	T1S

Span Interface Card

29.3.1 The FB-17277 or FB-15277-1A SIL (Span Interface Card) (Figure 29.3) receives and terminates the incoming bipolar signal, converts it to the incoming unipolar bit stream, and then extracts the clock frequency from the unipolar stream. The SIL card converts the outgoing unipolar stream into a bipolar signal that is compatible with T1-type span signaling. A strapping field is provided on the card for application configuration during installation as explained in TL-130300-1001. The span interface card provides the looping ability to test the digital cards for framing synchronization.

Each FB-15277-1A SIL card has two SINX outputs. Connect the SIL card to SINX 0. For cable lengths, refer to TL-130300-1001.

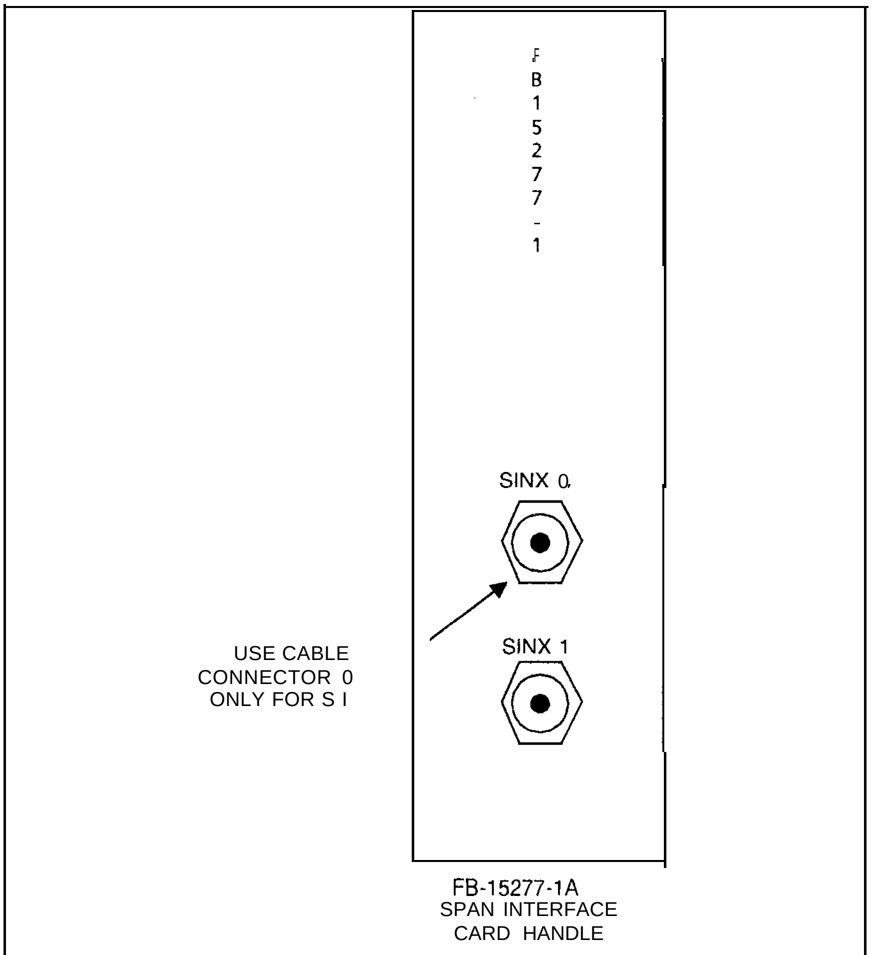


Figure 29.3 FB-15277-1A Span Interface Card Handle

Frame Detector Card

29.3.2 The FB-15278-A FDC (Frame Detector Card) performs the following functions:

- Monitors for errors in framing synchronization patterns.
- Provides a framing alarm signal to the Tl-Type Supervisory card when three or more bit-pattern errors are detected out of five incoming synchronization bits examined.
- Provides a signal to the SIL card that indicates a new frame of voice samples is about to arrive.
- Indicates when bit 1, the second most significant bit, is to arrive by providing a signal to the SIL card.
- Provides the supervisory frame signal that decodes channel A and B signaling.

Line Compensator Card

29.3.3 The FB-15280-A LCM (Line Compensator Module) card provides buffering to compensate for propagation delay variations due to changes in temperature of the span line. Two PCM frames are stored in the buffer on this card. This card receives the unipolar bit stream from the span interface card. This unipolar stream is received in serial format and converted to an eight-bit voice sample that is forwarded to the Tl-type buffer card in parallel format.

T1 Buffer Card

29.3.4 The FB-17192 Tl B2 (T1-Buffer) card provides a buffer between the incoming PCM data from the line compensator card and the digital time-switch network in the system, and also buffers the outgoing PCM data from the digital time-switch network to the Span Interface card.

The T1 B2 synchronizes and aligns the 24 PCM channels between the digital network and the T1 digital trunk interface.

T1 -Type Supervisory Card

29.3.5 The FB-20718-l A T1 S (T1 -Type Supervisory) card provides the supervisory signaling interface between the system and the T1-type interface. The T1S card contains buffers that retain the status of sense and control points (Figure 29.4).

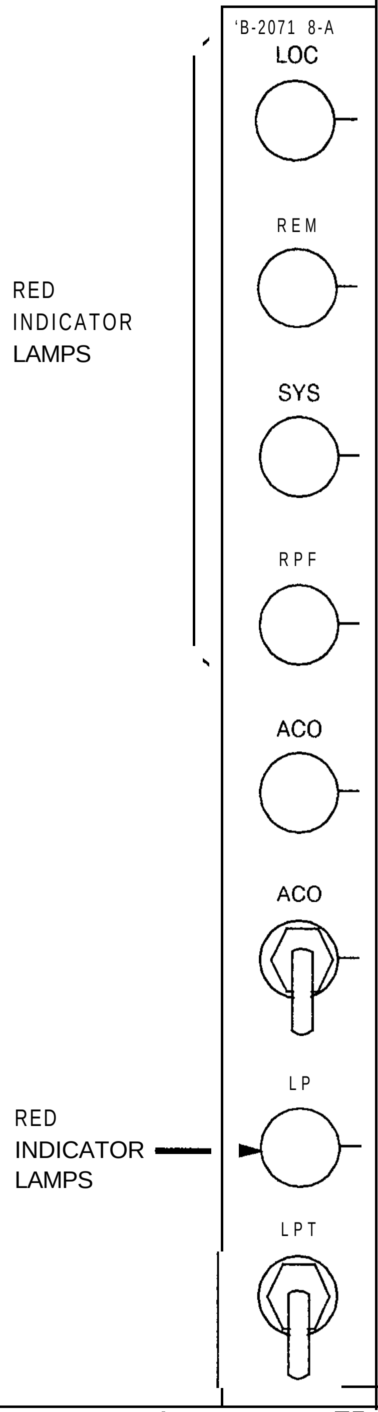
The T1S card has a program board that is strapped to condition the card to decode FX trunk signaling or E&M trunk signaling. Strapping is also provided to change from D2 to D3 signaling format and to provide a variable framing alarm delay time. Strapping option procedures are described in TL-130300-1001.

For alarm conditions, alarm detection processes, and associated troubleshooting procedures see TL-130200-1001.

Master Mode Operation

29.4 System master mode operation is used for applications where loss of a data frame is not critical, i.e., when the T1 data link handles analog (voice) transmission. For master mode operation, the FB-17277-A SIL card (instead of FB-15277-1A) is installed.

When the system operates in the master mode, the far-end channel banks must be conditioned for loop clock synchronization. If the far-end channel bank is a Siemens Transmissions Systems 9002A PCM channel bank, its Transmit Common Unit 9'1221 and Terminal Equipment Shelf 91220 must be modified for applications where the system functions in the master mode. A modification kit is supplied by FGBS. This modification kit provides the means whereby the FGBS timing signal recovered from the received bipolar signal is looped back to generate the transmit frequency rate. This allows the master clock in the system to provide looped clock synchronization. In the master-slave hierarchical timing structure, the system is the master clock and the channel bank at the far end is the slave.



LOCAL ALARM (LOC) LAMP THAT LIGHTS WHEN THE FRAMING OF THE INCOMING BIPOLAR SIGNAL IS LOST (INDICATES MISFRAMING; LOSS OF FRAMING SYNCHRONIZATION).

REMOTE ALARM (REM) LAMP THAT LIGHTS WHEN BIT POSITION 2 OF THE INCOMING BIPOLAR STREAM IS INHIBITED FOR A PERIOD OF 1.32 TO 1.44 SECONDS.

SYSTEM ALARM (SYS) LAMP THAT LIGHTS WHEN ANY ALARM CONDITION EXISTS, INCLUDING WHEN THE SYSTEM IS FULLY FRAME-SYNCHRONIZED BUT IN A LOOP MODE.

REMOTE POWER FAILURE ALARM (RPF) LAMP THAT LIGHTS WHEN A POWER FAILURE OCCURS IN THE OFFICE-TERMINATING SHELF.

ALARM CUTOFF (ACO) LAMP THAT INDICATES WHEN THE ALARM CUTOFF SWITCH IS ACTIVATED AND THE LOOP TEST SWITCH IS ACTIVATED.

A TWO-POSITION ALARM CUTOFF (ACO) SWITCH THAT IS ACTIVATED IN THE UP POSITION

LOOP SWITCH (LP) LAMP THAT INDICATES THE LOOP TEST AND ALARM CUTOFF SWITCHES ARE ACTIVATED.

A TWO-POSITION LOOP (LP) SWITCH THAT, WHEN IN THE UP POSITION, ALONG WITH THE ALARM CUTOFF SWITCH, ACTIVATES THE LOOP TEST.

Figure 29.4 FB-20718-1A T1-Type (T1S) Supervisory Card Handle

Slave Mode Operation

29.5 System slave mode operation is typically used when the T1 data link handles digital data transmissionthat must adhere to PCM toll communication standards. Present PCM toll communication standards require a maximum slip rate of one frame in 20 hours over a particular digital communication link. The system network clock card (FB-20771-1A) does not have the frequency stability to allow only one frame slip in 20 hours when the system is operating in the master mode. Therefore, it is necessary to make the system operate as the slave in the timing hierarchy when PCM toll communication standards must be met.

The system can be configured to slave its network clock to a far-end digital switch by using the FB-20922-A INCKS (Synchronized Intermediate Network Clock) card and the FB-15277-1A Tl-Type Span Interface Card (replacing the FB-17277 Span Interface Card). When slaved to another digital switch, the system derives its network timing from the incoming T1 -type bipolar digit stream. The FB-15277-l A Span Interface Card extracts a 1.544-MHz clock signal from the data on the Tl-type span.

The derived clock signal is cabled to the FB-17267-A NCWS card. The synchronizable clock card compares its output to the input frequency from the T1-type SID card and, by means of a phase lock loop and a voltage-controlled oscillator, synchronizes itself to the 1.544-MHz frequency of the T1 -type span bit stream. In this mode, the system is then slaved to the far-end digital switch, which acts as the controlling master clock.

The clock synchronization hardware on card FB-20922-A provides for monitoring four external 1.544-MHz clock signal inputs (SINXO to SINX3) on the card handle. One of the input clock signals is selected by the card and used for frequency synchronization. The clock card remains locked on an input until it determines that the clock signal has missed a pulse or died completely. When the SINX input that the clock card is locked onto fails, the card begins scanning the four inputs for a valid clock input. When it finds SINX input, the card will frequency-synchronize onto that SINX input.

When the network clock card is not locked onto an external T1 clock signal, it free-runs at a pre-fixed frequency. When a valid input is detected, the FB-20922-A card locks onto the external clock, cancels its free-running mode, and resynchronizes to the new input. When the network clock card is in its free-running mode, the system is synchronized to the free-running clock frequency. When the network clock card locks onto an SINX T1 input, the system is synchronized to the T1 input and functions in the slave mode.

Table 29.2 Analog T1 Nailed Connections Allowed

FROM	TO			
	GRD.START	**LOOP**	**E&M**	**LOOP DID**
GRD. START	No	No	No	No
LOOP	No	Yes	Yes	No
E&M	No	Yes	Yes	No
LOOP DID	No	No	No	No

1. The supervisory signals must be at least 200 msec in duration.
2. When unlike trunks are connected and any incoming signals are not translatable to the outgoing type of signal, the signal will be ignored.

REMOTE DATA PROCESSORS SYSTEM CONNECTIONS

30.0 Remote Processors (RPs) such as the APMs (Asynchronous Packet Managers), DFP/APM (Digital Featurephone with Asynchronous Packet Manager), and SPMs (Synchronous Packet Managers) are normally connected as shown in Figure 30.1.

The asynchronous and synchronous processor (APM, DFP/APM, SPM) can be located remotely to reduce traffic within the system. The remote processors eliminate the need for in-house processors.

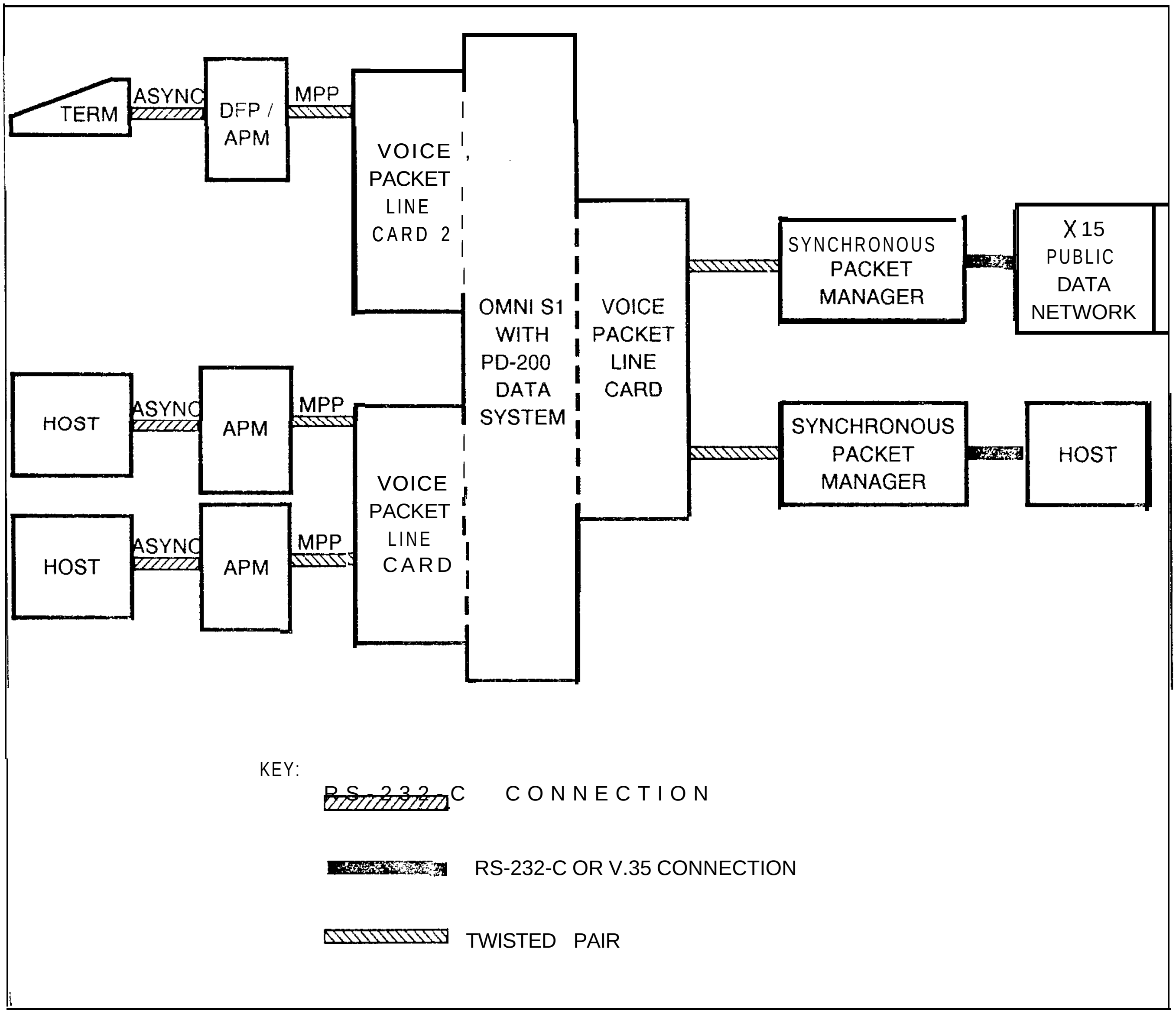


Figure 30.1 Connection of Data Devices

THIS PAGE IS INTENTIONALLY LEFT BLANK.

APM (ASYNCHRONOUS PACKET MANAGER) EC-221 79-A

31.0 The microprocessor controlled APM (Figure 31 .1) provides the interface between the asynchronous data equipment (terminals, personal computers, etc.) and the VPLC (FB-17226) card or VPLC2 (FB-17426) card. Each APM connects to one asynchronous data device.

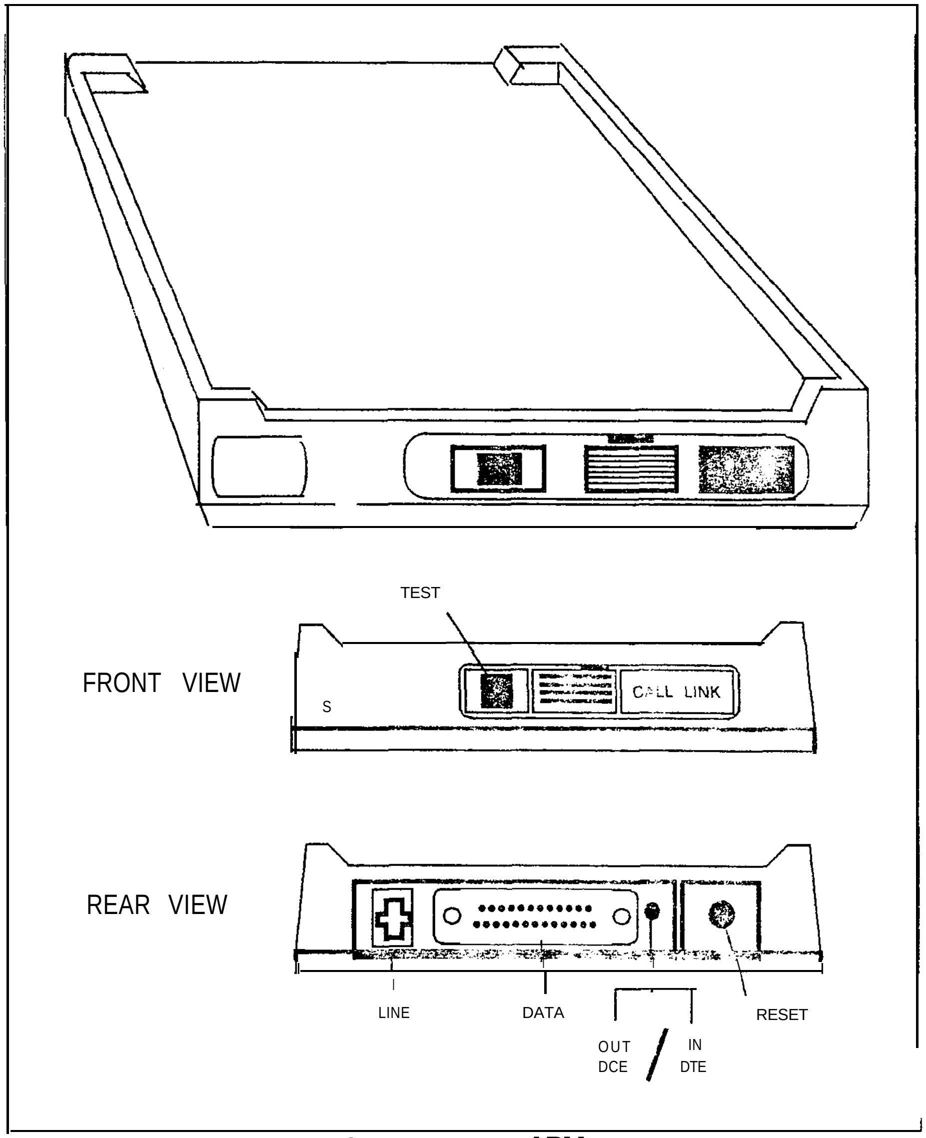


Figure 31.1 APM

The APM Assembly EC-22179-A

- Is used for devices that require an asynchronous W-232-C type interface
- Supports devices that operate at speeds up to 19.2 Kbps

The VPLC (FB-17226) that is used to support an APM is configured as a type VPLC 0 in data base, allowing all 8 ports of the card to be used (see Figure 31.2).

NOTE: The VPLC2 (FB-17246) card, when configured as a type VP20 card in data base, can support APMs, DFP/APMs, and DFPs.

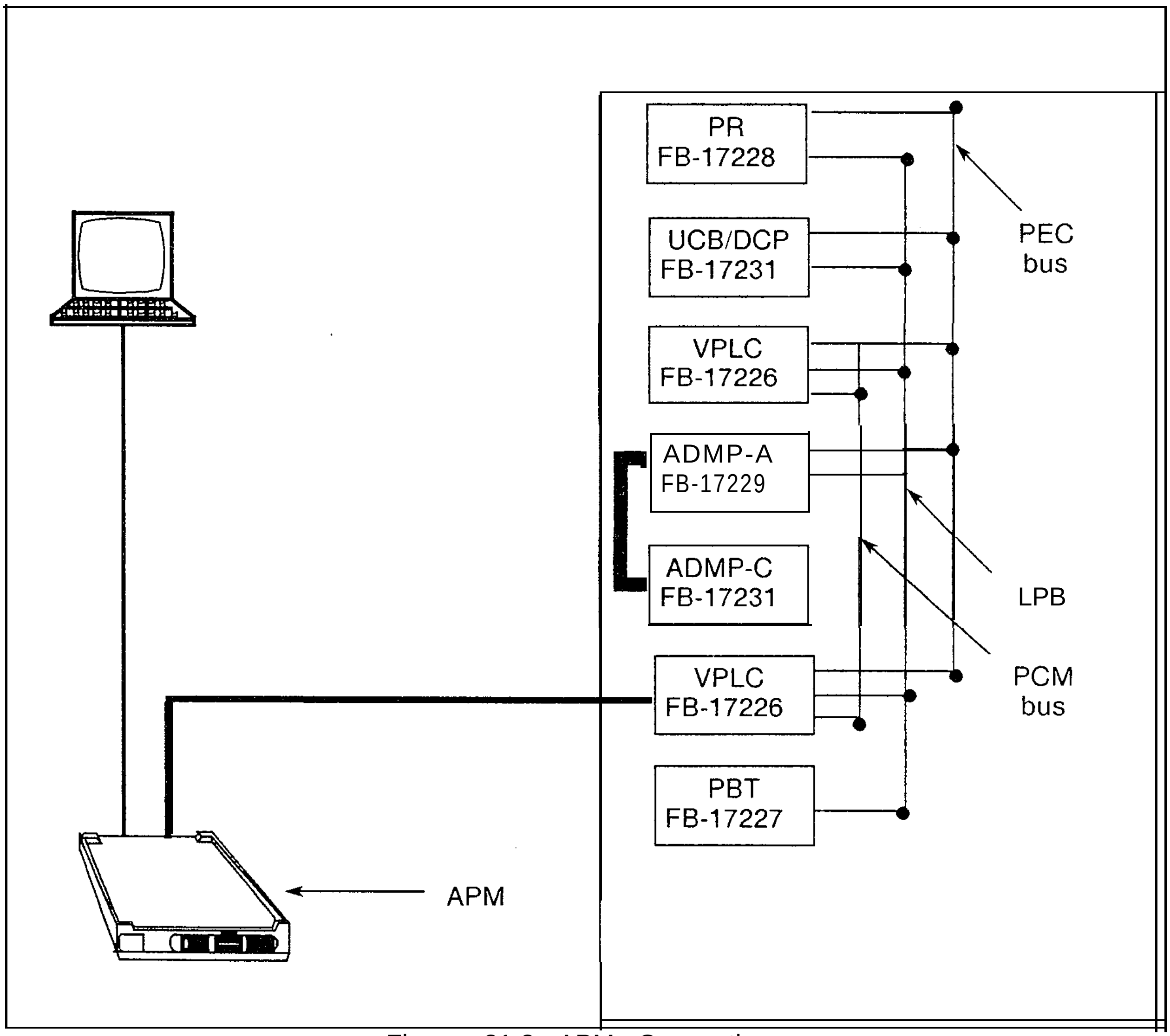


Figure 31.2 APM Connection

The APM, designed for desk-top installation, is housed in a small plastic case. A telephone can be put on top of the APM; cable connections are at the back of the APM. Each APM communicates with the system over a single twisted wire pair. This connection provides for continuous two-way transfer of data only (no voice). Using 26 gauge wire, the maximum distance from the APM to the system is 3000 feet. The maximum distance from the APM to the terminal equipment is 50 feet. The twisted pair (tip and ring) connection maximum distance between the APM and the system is 3,000 feet at 26 gauge. The maximum distance between the APM and its data equipment is 50 feet.

The APM operates from 115 VAC through a step down transformer (to 22 VAC).

The APM supports one single-speed asynchronous data device operating at speeds of 110, 300. 4 2(?), 2400. 4800, 9600 baud, or 19,200 BPS (bits per second). For RS-232-C compatible signals (ASCII) to be transmitted to the data system, it must first be converted into X.25 packets. The X.25 packet then must be broken down into 11 byte self-routing mini-packets for transportation to the system (Figure 31.3).

The APM contains an 8-bit microprocessor memory, 16K of parity protected RAM, 8K of ROM, and a UART (Universal Asynchronous Receiver/Transmitter). Seven RS-232-C/V.24 data set signals can be controlled. The AMP contains the mini-packet receiver/transmitter and line driver circuits used to communicate with the system. The ROM (Read-Only Memory) in the APM contains all programs necessary to boot the APM from the switch, and debugging and limited hardware self-tests. The RAM in the APM provides the operational code needed to communicate via the data system and is used for temporary storage, buffers, and operational parameters.

The APM performs PAD (Packet Assembler/Disassembler) functions as well as containing the CRC (Cyclic Redundant Checking) error detect logic (see Figure 31.4).

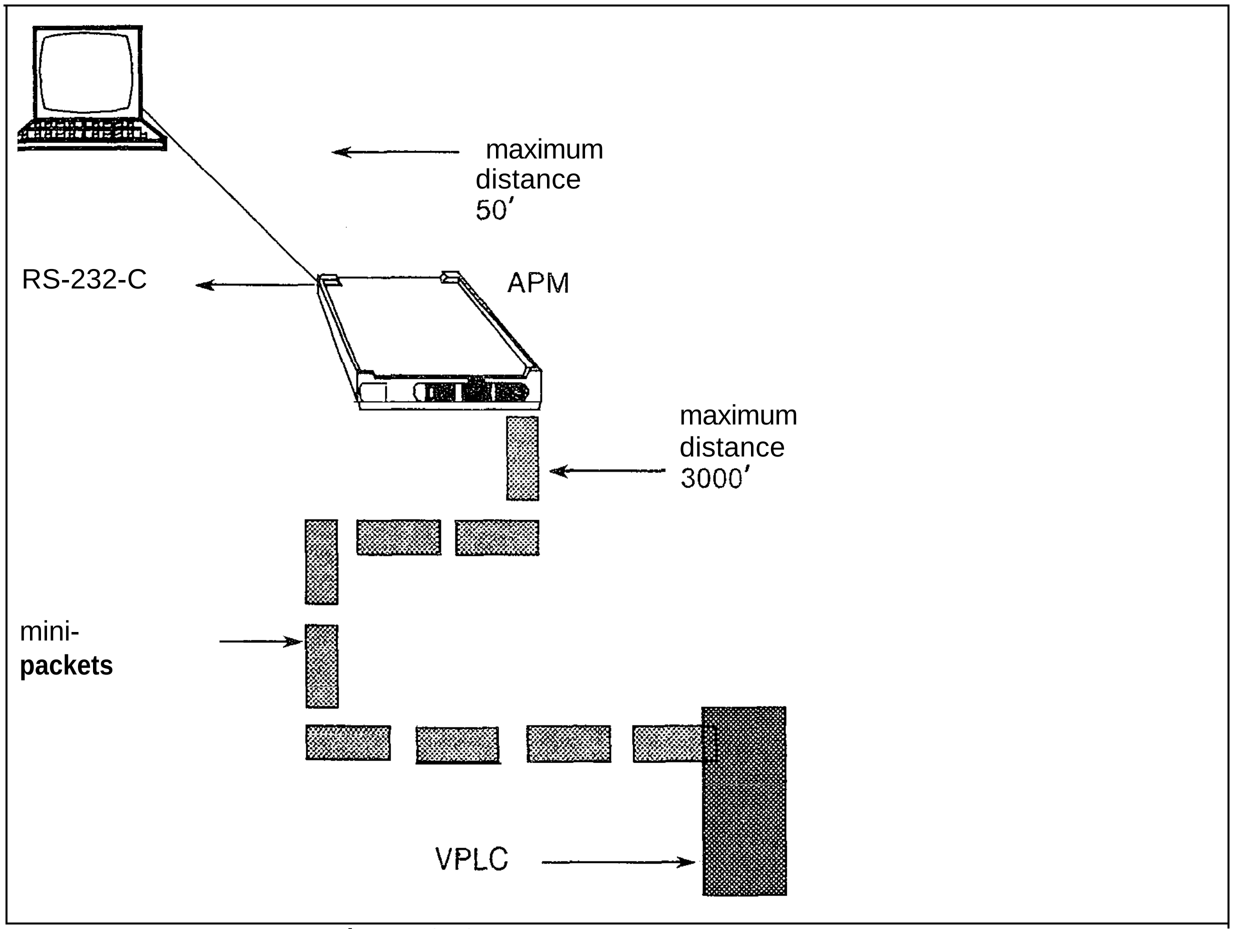


Figure 31.3 Transport of Mini-Packets

The protocol that governs the function of the PAD is X.3. The X.3 protocol provides a set of values or parameters that characterize the operations of the terminal that is paired with the APM.

X.3 Parameters

31.1 The X.3 parameters define the following basic functions of the data terminal:

- Packet Forwarding
- Break Handling
- Flow Control
- Device Specific Characteristics
- Local Editing
- Local Echo

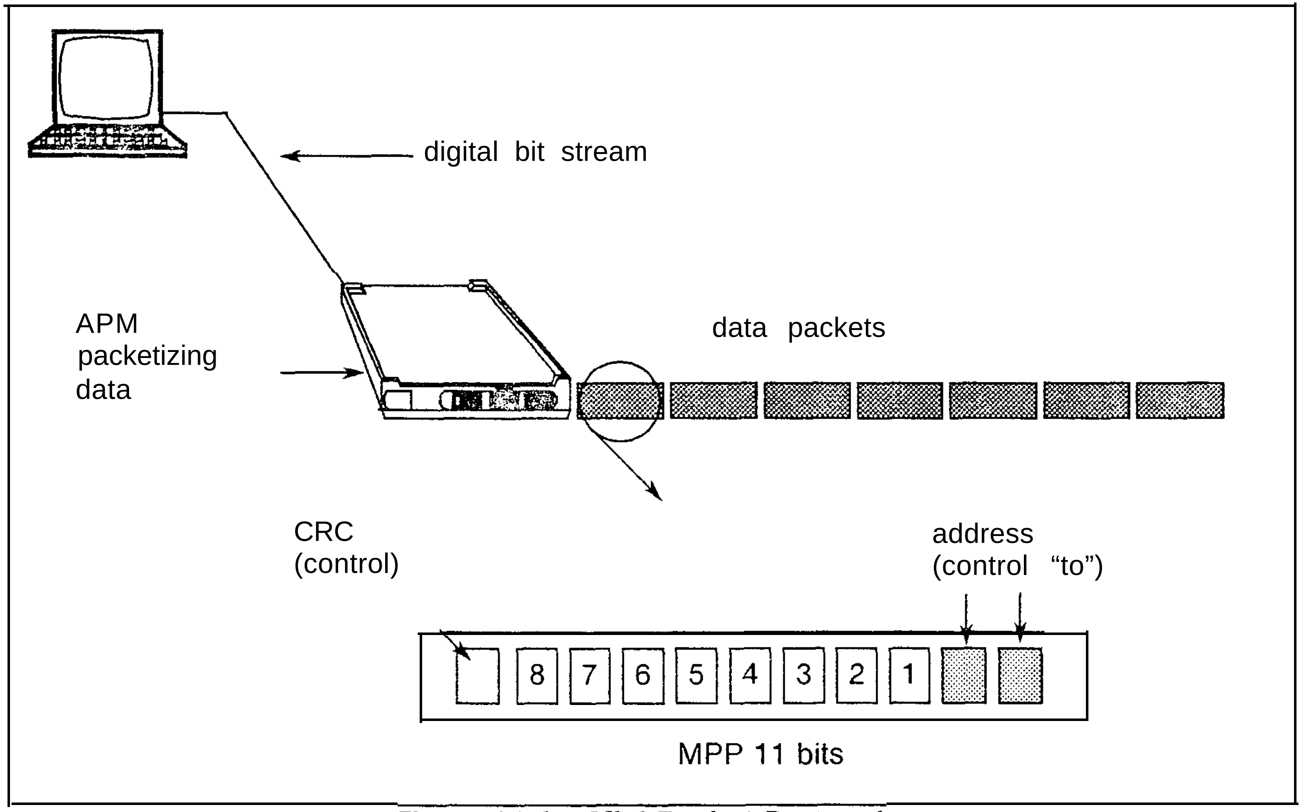


Figure 31.4 Mini-Packet Protocol

There are eighteen standard X.3 parameters. The OMNI requires an additional seven non-standard parameters, which are referred to as local parameters. The X.3 parameters are defined as follows:

Parameter 1 - Escape from Data Transfer

Parameter 2 - Echo

Parameter 3 - Selection of Data Forwarding Signal

Parameter 3 - Selection of Idle Timer Delay

Parameter 5 - Ancillary Device Control

Parameter 6 - Control of Pad Service Signals

Parameter 7 - Selection of Operation of the Pad on Receipt of the Break Signal

Parameter 8 - Discard Output

Parameter 9 - Padding After Carriage Return

Parameter 10 - Line Folding

Parameter 11 - Binary Speed

Parameter 12 - Flow Control of the Pad by the Start-Stop Mode DTE (Data Terminal Equipment)

Parameter 15 - Editing

Parameter 46 - Character Delete

Parameter 17 - Line Delete

Parameter 18 - Line Display

Local Parameters

31.2 In addition to the eighteen X.3 parameters the system operates under seven local parameters

- 0 - Escape to Local Parameters
- 1 - Profile ID (Terminal ID)
- 2 - Eight Bit Transparency
- 3 - Parity
- 4 - Full Duplex
- 5 - Interval Timer
- 6 - Echo Mask

An APM may be connected to a modem for access to the public switched network. In this case, it is necessary that the characteristics of the modem on each side of the connection match.

The APM contains the following switches and status indicators:

- LINK Indicator. The LINK indicator shows whether the APM is in a test condition or is receiving/transmitting mini-packets.

- CALL Indicator. The CALL indicator shows whether the APM is in a call setup or loopback condition.

- TEST Switch. The test switch is used to send an event to the ADMP to test the path to the APM.

- RESET Switch. The RESET switch resets the APM when activated.

- DTE/DCE Switch. The DTE/DCE switch selects the APM communication mode of operation. The APM communicates with terminal equipment in the DTE mode of operation. The APM communicates with modems in the DCE mode of operations.

SPM (Synchronous Packet Manager) (EC-221 78-A/B)

32.0 The microprocessor-controlled SPM (Figure 32.1) provides the interface between the synchronous X.25 data equipment (Public Data Network, Host Computer, etc.) and the VPLC FB-17226 card or the VPLC2 FB-17246 card. Each SPM connects to one synchronous X.25 data device.

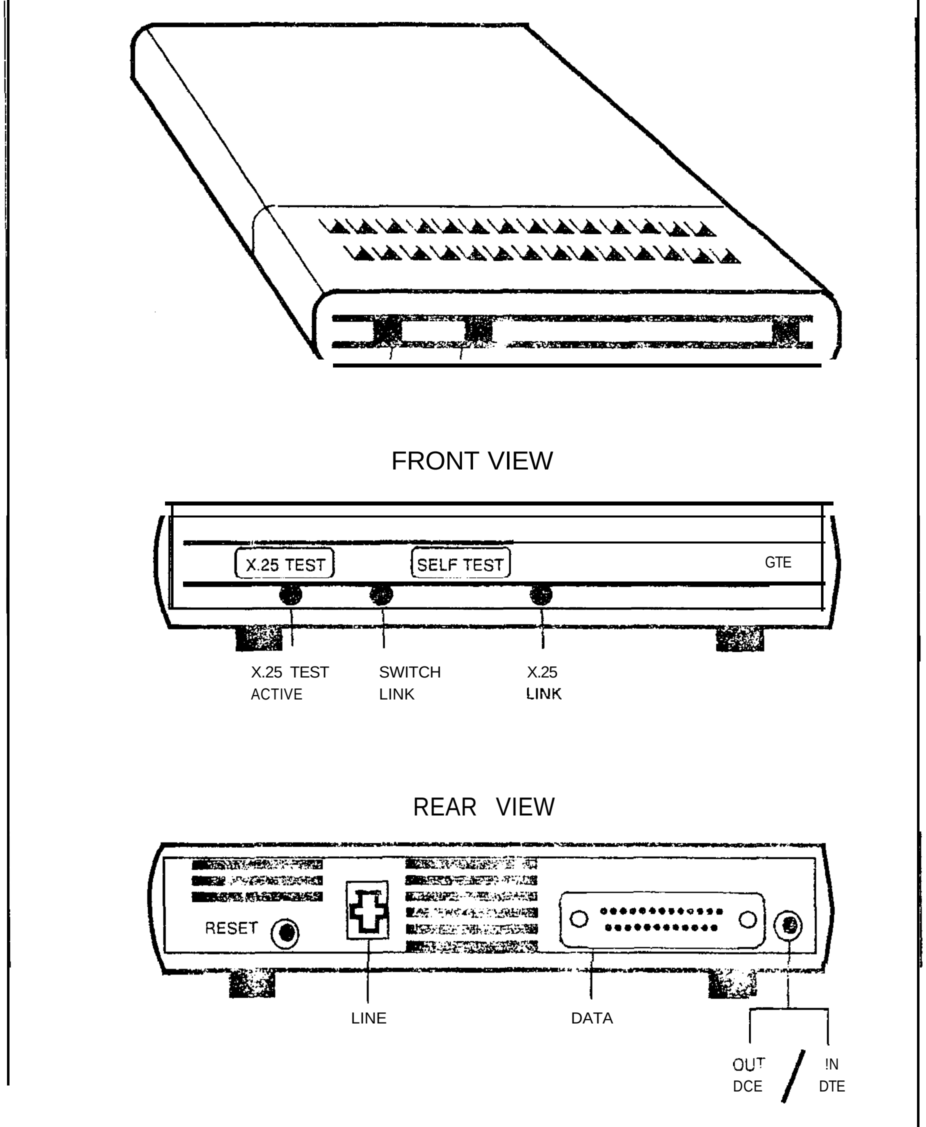


Figure 32.1 SPM (Synchronous Packet Manager)

Two versions of the SPM are available:

- Assembly EC-22178-A (Figure 32.1)
  - Required by devices that use an RS-232-C/V.24 type interface.
  - This assembly supports devices that operate at speeds up to 19.2 Kbps.
  - The VPLC (FB-17226) used to support the EC-22178-A, is configured as VPLO in data base allowing all 8 ports of the card to be used.

NOTE: The VPLC2 FB-17246 card, when configured as a VP20 card in data base, can support low-speed SPMs as well as DFP/APMs and APMs.

- Assembly EC-22178-B (Figure 32.1)
  - Required by devices that use an V.35type interface.
  - This assembly supports devices that operate at speeds up to 64 Kbps.
  - The VPLC (FB-17226) used to support the EC-22178-B, is configured as VPL1 in data base allowing it to use only two ports on the card. It is necessary to reduce ports available for high speed devices due to throughput requirements.

NOTE: The VPLC2 FB-17246 card, when configured as a VP21 card in data base, can support two high speed SPMs.

The SPM, designed for deskto-p installation, is housed in a small plastic case somewhat larger than the APM. Cable connections are made at the back of the SPM.

Each SPM communicates with the system over a single twisted-pair wire. This connections provides for continuous two-way transfer of data only (no voice) information. Using 26 gauge wire, the maximum distance from the SPM to the system is 3000 feet. The maximum distance from the SPM to the terminal equipment is 100 feet. An SPM is locally powered over a separate twisted pair.

The SPM operates from 115 VAC through a stepdown transformer to 24 VAC. The SPM supports up to 256 virtual circuits allowing for up to 256 multiple calls over the SPM's twisted pair line. Multiple virtual circuits are maintained by the logical channel numbers and device numbers that are assigned to each data call by the SPM.

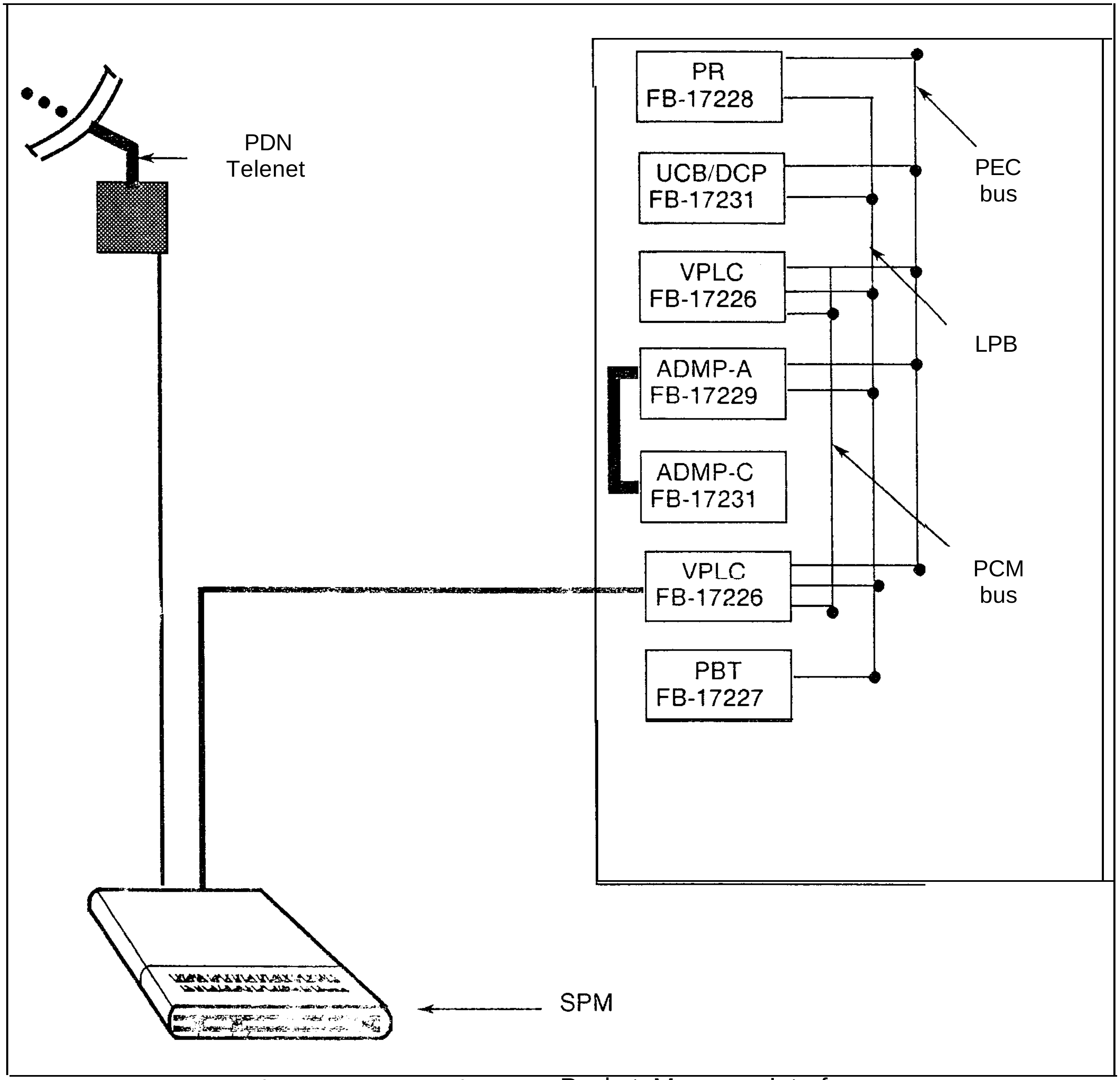


Figure 32.2 Synchronous Packet Manager Interface

For an X.25 packet to be transmitted to the data system, it must be broken down into 1 l-byte packets, called mini-packets, which adhere to the MPP (Mini-Packet Protocol) format. For transmission from the PABX to the data network, mini-packets are stored (buffered) until a full X.25 packet is formed and then transmitted (see Figure 32.2).

The SPM contains two 8-bit microprocessors, 80K of parity protected RAM, and 8K of ROM. The SPM contains the mini-packet receiver/transmitter and line driver circuits used to communicate with the PABX. The ROM in the SPM contains the programs necessary to boot the SPM from the PABX, a debugger, and limited hardware self-tests. The RAM in the SPM is provides the operational code needed to communicate with the data system, and is used for temporary storage, buffers, and operational parameters. The SPM performs packet assembler/disassembler functions and also contains the CRC (Cyclic Redundant Checking) error detect logic.

The SPM contains the following switches and indicators:

- SWITCH LINK Indicator. This indicator shows whether the SPM is in a test condition or is receiving/transmitting mini-packets.

- X.25 TEST ACTIVE Indicator. This indicator is off during normal operation. When the TEST pushbutton is depressed and diagnostic software is being downloaded from the system, this indicator will be on.

- W25 TEST Pushbutton. This pushbutton is not operational for this version.

- RESET Pushbutton. The RESET pushbutton causes a reset condition when depressed.

- DTE/DCE Switch. This two-position switch sets the SPM data communications mode of operation. When depressed, the SPM is in the DTE (Data Terminal Equipment) mode, allowing it to be connected to terminal equipment. When released, the SPM is in the DCE (Data Communication Equipment) mode, allowing it to be connected to transmission lines.

DFP/APM (DIGITAL FEATUREPHONE WITH APM)

33.0 The DFP/APM provides voice communication as well as the interface between the asynchronous data equipment (terminals, personal computers, etc.) and the VPLC2 FB-17246 card.

The DFP/APM interfaces with the voice system over the PEC and PCM buses and the data system over the LPB (Figure 33.1).

Connection to the VPLC2 (FB-17246-A) card is over a single twisted pair. This connection supports both voice and data communications. The DFP/APM is either locally or remotely powered over a separate twisted pair connection.

Connection to the data device is provided by a standard RS-232-CN.24 interface, located in the rear of the phone. Operating speeds of the data device can be 110. 300, 1200, 2400, 4800, 9600 baud, and 19,200 bps.

Contained within the DFP/APM are the 8-bit microprocessor, memory, packet assembler/disassembler, transmit/receive buffers, the CRC (Cyclic Redundant Check) error detect logic.

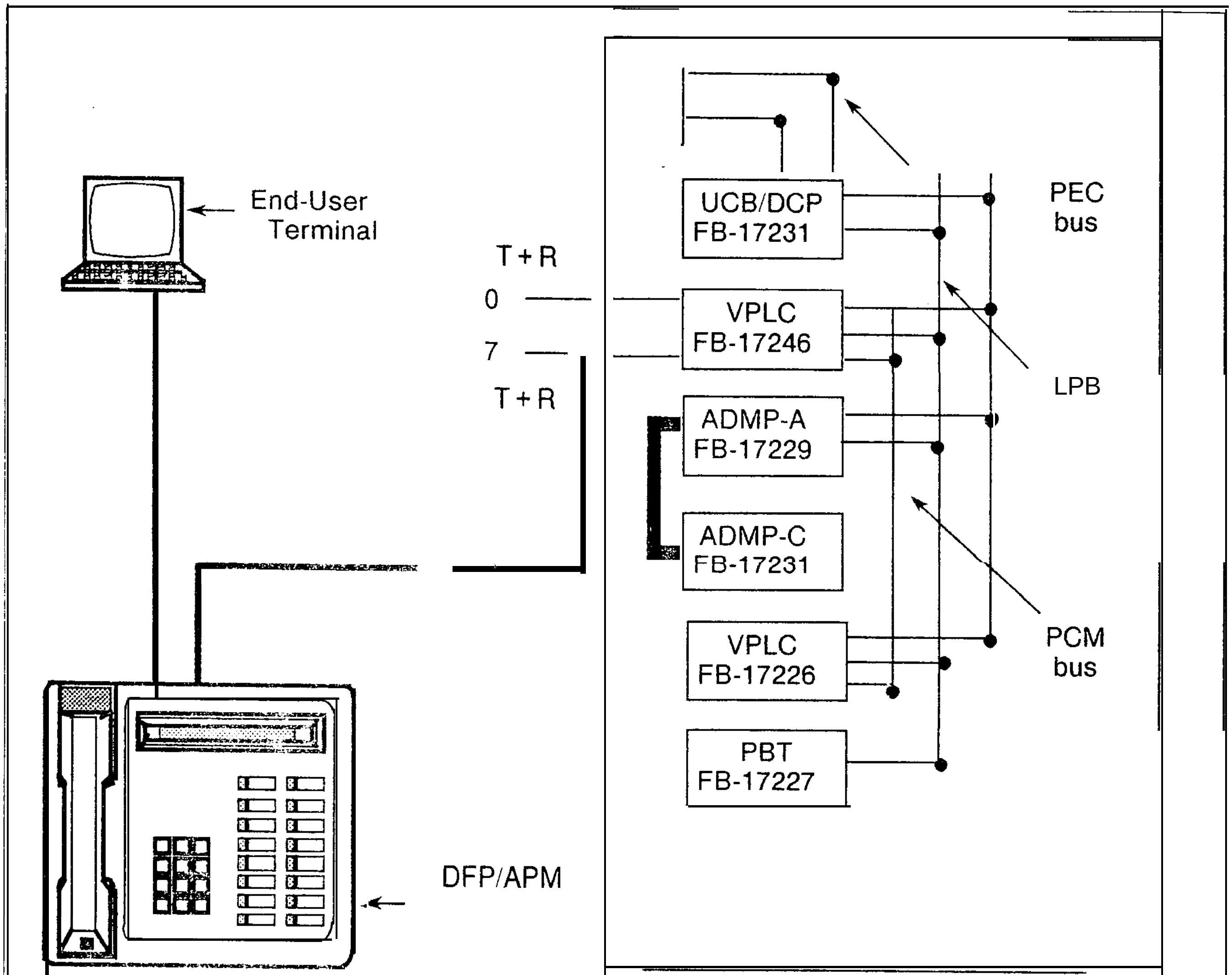


Figure 33.1 DFP/APM Voice and Data Bus Connection

DFP/APM Transmission

33.1 Transmission of 11 -byte voice, data, and control information is sent as mixed mini-packets. The 11 -byte mini-packets are transmitted at 128 Kbps in each direction. The effective throughput is 64 Kbps voice, 19.2 Kbps data and overhead = 128 Kbps in each direction.

See Table 33.1 for feature comparisons of the APM, SPM, and DFP/APM. Tables 33.2 and 33.3 provide CCITT recommendations and descriptions for the APM, SPM, and DFP/APM.

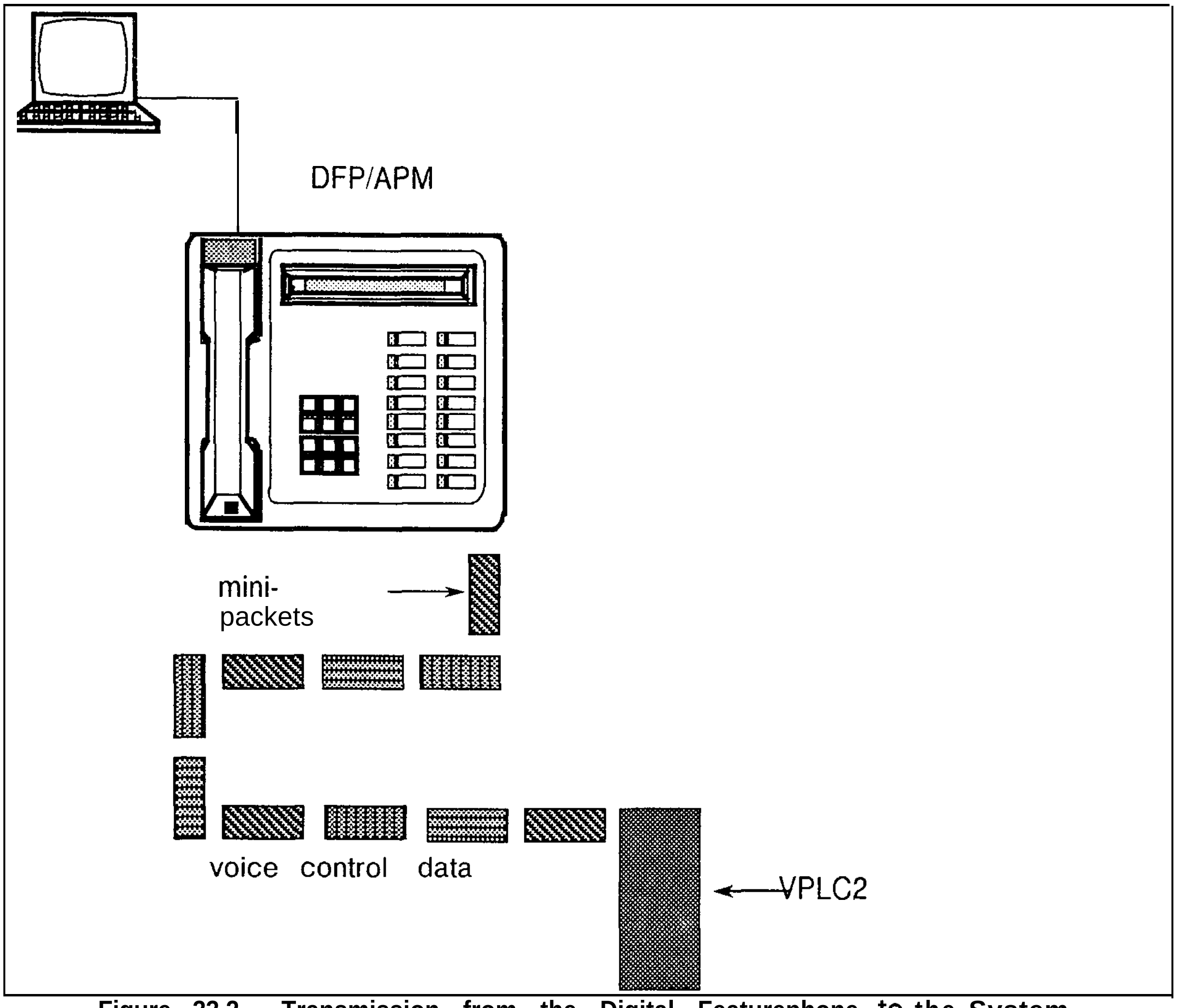


Figure 33.2 Transmission from the Digital Featurephone to the System

Simultaneous Voice Packetizing and Data

33.2 The simultaneous voice and data structure is:

- Data device log-on
- Data call distinctive or simultaneous with voice call
- Data characters briefly buffered at Digital Featurephone
- Data characters packetized in 8-byte groups at Digital Featurephone
- Preceded with two address bytes
- Followed with one CRC error-checking byte
- 11 bytes equal one mini-packet

When the mini-packet arrives at the VPLC2 (Voice Packet Line Card) (Figure 33.2), it is given two extra bytes providing “from” information.

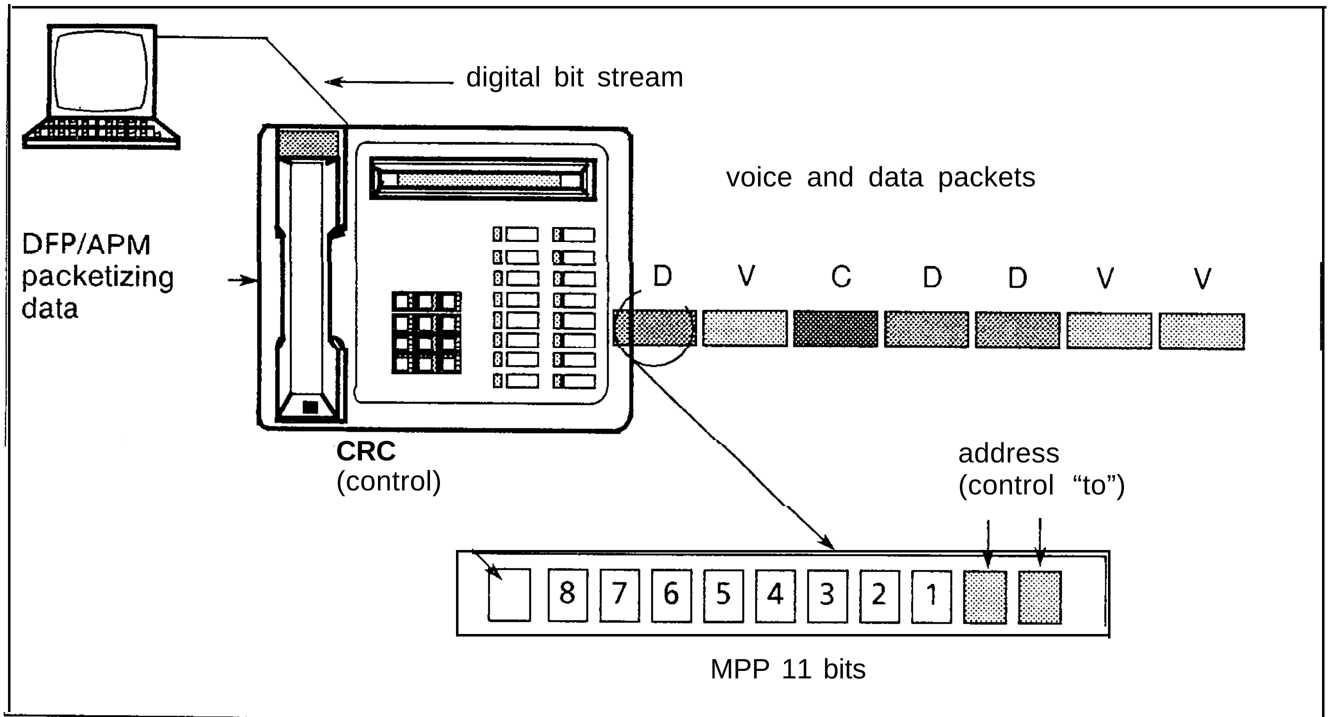


Figure 33.3 Simultaneous Voice and Data Packet Structure

Network Interface Card

33.3 The NIC (Network Interface Card) bridges the LPB (Local Packet Bus) and the PCM (Pulse Code Modulation) bus. This provides data networking capabilities between systems across a T1 network interface. HDLC (Higher Data Level Control) conveys the data to/from the PCM bus while MPP (Mini-Packet Protocol) conveys data to/from the LPB (Figure 33.3).

The NIC provides an X.25 communication link between a circuit-switched T1 trunk line and a packet-switched async/sync data line. The card indirectly interfaces to an outside T1 line via internal circuit-switched connections on the PCM bus. True intergrated voice and packet-switched data networking capabilities are provided by this card (see Figure **33.4).**

The NIC's responsibility is similiar to that of the SPM. However some major physical differences between the NIC and the SPM are:

- The NIC occupies a complete slot space allowing it an ABI (Arbitrated Bus Interface) and a PCM interface
- The NIC has 256K bytes of shared memory
- The NIC has DMA (Direct Memory Access) hardware support, an associated new MPCC (Multiple Protocol Communication Control) chip, and a new interrupt control mechanism.

Table 33.1 Feature Comparison

FEATURE	DFP/APM	APM	SPM
Connects to switch via (VPL0) *1		X	X low speed only
Connects to switch via VPLC (VPL1) *1			X
Connects to switch via VPLC2 *1	X	X	
Connects to switch over twisted wire pair	X	X	X
Powered locally or remotely over separate twisted pair at +24 VAC or -48 VDC	X	X	
Powered locally over separate twisted pair at +22 V		X	
Powered locally over separate twisted pair at +24 V			X
Maximum distance from switch-3000 feet at 26 gauge	X	X	X
Uses a standard RS-232-C/V.24 interface	X	X	X*2 VPL0/VP20
Uses a CCITT V.35 interface			X*4 VPL1/VP21
Can operate at speeds of 110, 300, 1200, 2400, 4800, 9600 baud & 19,200 bps	X	X *3	X
Can be set as DTE or DCE	X	X	X
Maximum distance from data device-50 feet	X	X	
Maximum distance from data device-100 feet			X
Supports one active data (virtual circuits) call at half or full duplex	X	X	
Supports 255 active data calls (virtual circuits)			X

NOTES:

* 1. For information on VPLC cards, see section 7.0.
* 2. SPMs can interface with an RS-232-C if speeds are below 19.2 Kbps.
* 3. Throughput will vary according to the PAD (Packet Assembler/Disassembler) parameter settings and data format.
* 4. For speeds > 9.6 Kbps up to 64 Kbps.

Table 33.1 Feature Comparison (Continued)

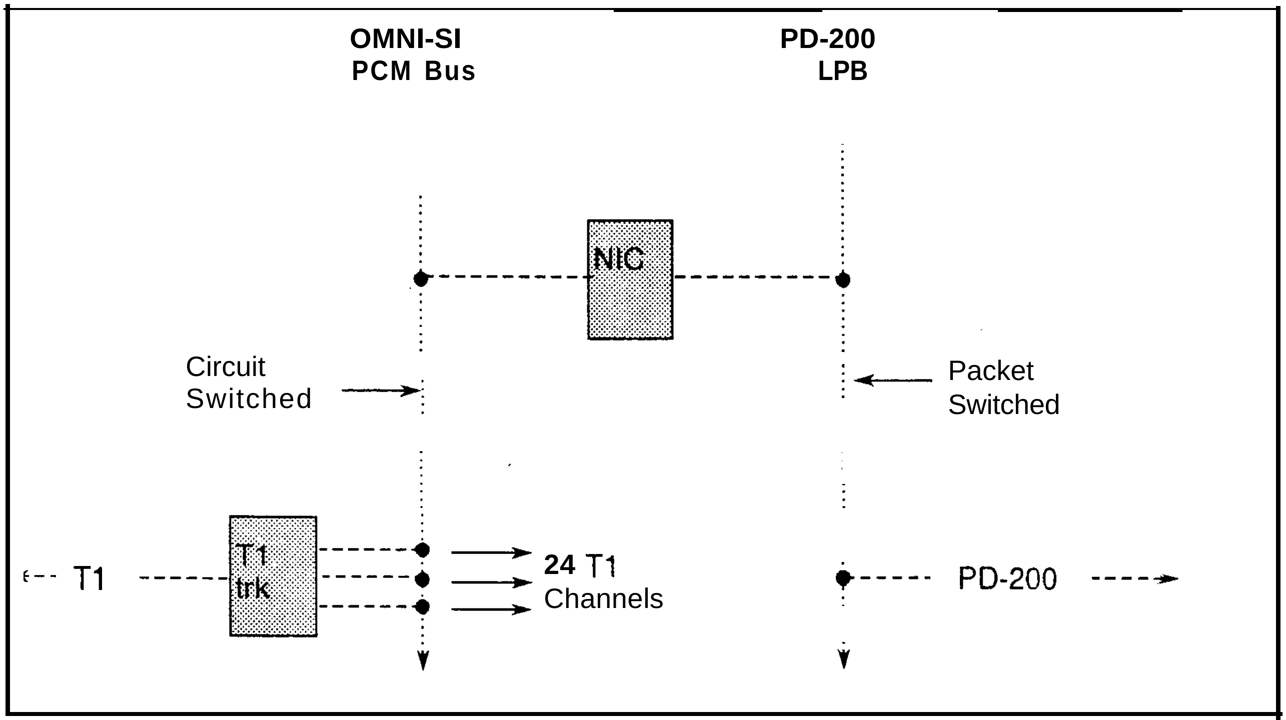
FEATURE	DFP/APM	APM	SPM
Provides all processing and transmission of data, other than call-control related events (call setup, disconnect, etc.)		X	
Contains 8 bit microprocessor	X	X	X
Contains Memory Packet Assembler/ Disassembler (PAD)	X	X	
Contains transmit/receive buffers	X	X	X
Contains CRC error detect logic	X	X	X
Contains CODEC and other associated circuitry	X	X	X

Table 33.2 CCITT Recommendations

FEATURE	DFP/APM	APM	SPM
Addressing by X.121 numbering scheme	X	X	X
Assignment of more than 1 X.121 addresses if routing pattern calls for it			X
X.3 packet assembler/disassembler function	X	X	
X.28 type command language (not fully X.28 compatible)	X	X	
Supports X.25 level 3 (packet level) and level 2 (link level, LAP-B protocol)			X
NOTE: X.28 is the language by which an X.3 command is entered, converting async. data into X.25 packets, thereby allowing the user to enter information through the keyboard of the data device attached to the APM			
Interfaces externally to X.25 host or X.25 PDNS (Telenet)			X

Table 33.3 DFP/AMP, APM, SPM Description

DEVICE	FEATURE
DFP/APM	• Provides interface between asynchronous data equipment and the system • Receives and transmits W-232-C compatible signals, converting them into mini-packet protocol • Communicates with the system over a single, twisted-wire pair • Supports speeds up to 19.2 Kbps
EC-221 79-A .APM	• Provides interface between asynchronous data equipment and the system • Receives and transmit RS-232-C compatible signals, converting them into mini-packet protocol • Communicates with the system over a single, twisted-wire pair • Supports speeds up to 19.2 Kbps
EC-221 78-A SPM Type A	• Used for devices needing RS-232-C interface • Supports devices operating at speeds up to 19.2 Kbps • Communicates with the system over a single, twisted-wire pair
EC-221 78-B SPM Type B	• Used for devices needing V.35 interface • Supports devices operating at speeds up to 64 Kbps • Communicates with the system over a single, twisted-wire pair



Fiaure 33.4 NIC Card within the OMNI SI System's Architecture

SYSTEM SOFTWARE

34.0 System software is classified as either generic software (operating system) or data base software (site dependent). All major call-processing functions are performed by the system generic software, which receives direction from the operating parameters incorporated into the system data base software.

The software is loaded into the system through a disk drive. First, a floppy disk containing the generic program is put in the disk drive and loaded onto hard disk. Next the data base disk is loaded onto the hard disk. The system memory can then be loaded from the contents of the hard disk. Details of disk loading are contained in TL-130300-1001. For information on updating disks to accommodate changes, see TL-130400-1001.

Generic Software and Subprograms

34.1 All major call-processing functions are performed by the system generic software, which receives direction from the operating parameters incorporated into the system data base software.

Generic software residing in the system CPU memory is primarily divided into the following subprograms:

- Executive
- Inter-process Communications
- Call Control
- Attendant Console/Agent Instrument Control
- Digit Analysis
- Administration/Maintenance
- Line Control
- Station Features
- Trunk Control
- System Features
- CAS Branch
- Message Detail Recording
- Hotel/Health Care
- Featurephone Directive-Command Handling
- Featurephone Response-Event Handling

Executive Subprogram

34.1.1 The executive subprogram maintains real-time control of the system environment and is responsible for performing the following operations:

- Job scheduling
- Monitoring and controlling system operations
- Allocating common system resources

The operating subprogram provides routines within the supervisory program that accomplish process-report and real-time services for the application programs. The executive subprogram is divided into the foreground scheduler and the background scheduler.

- The foreground scheduler consists of a dispatcher that periodically schedules the following routines:
  - Peripheral processor foreground scheduler activation
  - System real-time clock update
  - Outpulsing DTMF and MF tone
  - Sending CAS audible tone identification

- The background scheduler consists of a dispatcher that schedules routines that perform the following:
  - Check for 100-ms, 500-ms, l-second, and l-minute scheduled jobs.
  - Process events including line- and trunk-status changes, such as on-hook and off-hook conditions and digit presentation, Attendant Console actions, and system maintenance test results.
  - Process information obtained by the Foreground Scheduler.

Inter-Process Communication Subprogram

34.1.2 The inter-process communication subprogram performs the following operations:

- Passes the action directive to the proper processing application program.
- Reports the events from each peripheral device and passes them to the proper processing application program.

Call Control Subprogram

34.1.3 The call control subprogram processes the events reported by the digit-outpulsing and trunk-scan routines, the console-handler and digit-analysis subprograms, and allocates the following system resources when required:

- Call stores maintain the current status of all calls in progress in allocated blocks of memory. A call store is allocated at the time a line or trunk seizure is processed and retains this information throughout the call duration.
- Register-senders (digit stores)
  - Register-sender (digit store) memory blocks are dynamically allocated upon processing of a line seizure and are linked to the associated call store during digit accumulation and outpulsing.
  - The register-sender (digit store) memory contains the control words required to accumulate, store, and outpulse a maximum of 16 digits.

Attendant Console Control Subprogram

34.1.4 The Attendant Console/Agent Instrument control subprogram performs the following operations as required by the background scheduler:

- Analyzes the status of the Attendant Console and Agent Instrument pushbuttons as reported by the peripheral processors' status-handler routine.
- Causes the appropriate lamps on the Attendant Console and Agent Instrument to light.
- Reports the status of the Attendant Console and Agent Instrument pushbuttons to the call processing subprogram for further disposition.

For example, depressing one of the LOOP pushbuttons and the START/STOP pushbutton on the Attendant Console causes the remote console handler subprogram to light the LOOP, POS RLSE, and START/STOP pushbuttons/LEDs and to report the event to the call processing subprogram. The call processing subprogram then allocates a call store, a register-sender, and a DTMF receiver to that loop.

Digit Analysis Subprogram

34.1.5 The digit analysis subprogram is activated when the system receives a digit and performs the following operations:

- Analyzes each received digit.
- Checks system features against the codes dialed.
- Checks to ensure that a connection can be made to a terminator.
- Reports the type of equipment to which the call has been terminated.

On station-to-trunk calls, the digit analysis subprogram monitors the outpulsed digits and performs the appropriate digit-absorption and code-screening functions.

Administration Subprogram

34.1.6 The administration subprogram performs the following:

- Memory-reload routine

  - Reloads information into the system's generic and data base memories when the system is initialized or on a return from a power failure.

  - The routine is resident in the ROM on the system CPU card and, when activated, reloads all of the software data base.

  - The memory reload function in the peripheral processor is controlled by the on-line central processor.

- Disk routine

  When activated, outputs the data base from system CPU memory onto the hard disk in data base files.

- Traffic-data-verification routine. When activated, performs the following functions:

  - Collects event and usage data.
  - Automatically outputs the collected data if desired.
  - Provides an interface for remote data-collection systems.

- Input/output routine. When activated, performs the following functions:

  - Provides an interface for the digital system test set.
  - Provides an interface for the TTY.
  - Provides local/remote memory read/write capability.

Maintenance Subprogram

34.1.7 The maintenance subprogram consists of the following routines:

- The fault-detecting test routines are run continuously to detect the following faults:

  - Memory faults
  - Network faults
  - Interprocessor communication faults (central processor-to-peripheral processor and central processor-to-central processor)

- The fault-analysis and fault-report routines analyze detected faults and perform the following functions:

  Rerun the fault-detecting test or a different test.

  - Report the fault to the Attendant Console where it is indicated by an alarm display and hard copy via maintenance TTY.

  - Store fault data in an error log and initiate a reconfiguration.

- The system-reconfiguration-and-call-recovery routine performs the following functions:

  a. Reloads the peripheral processor after a peripheral processor fault.

  b. Saves all established two-way and three-way calls.

- Installation initialization, cold start for the data system (equipped).

Line Control Subprogram

34.1.8 The line control subprogram monitors the lines for on-hook, off-hook, and flash conditions, and reports any activity to the call control subprogram. The line control subprogram assigns the digit stores and DTMF receivers to lines when needed.

Station Features Subprogram

34.1.9 The station features subprogram controls and executes the station features with the digit analysis and call control subprograms.

Trunk Control Subprogram

34.1.10 The trunk control subprogram processes incoming and outgoing trunk seizure, incoming and outgoing trunk signaling, and trunk release information with the call control subprogram. The trunk control subprogram also allocates digit stores and DTMF receivers to trunks when needed.

System Features Subprogram

34.1.11 The system features subprogram performs the system features under the control of the digit analysis subprogram, and reports to the call control subprogram. The following features are executed by this subprogram:

- Intercept
- Class-of-service checks
- AIOD (Automatic Identification of Outward Dialing)
- Station hunting
- ACD (Automatic Call Distribution)

Centralized Attendant Service Branch Subprogram

34.1.12 The CAS branch subprogram monitors seizure, signaling, and release of the PRLTs (PCM Release Line Trunk) with the call control subprogram. The CAS branch subprogram also allocates digit stores and DTMF receivers for the PRLTs. The CAS branch subprogram also performs the following CAS related features:

- CAS attendant camp-on and recall

- CAS attendant recall on no-answer

- CAS attendant silent hold

- PRLT night answer control and direct and group access

Message Detail Recording Subprogram

34.1.13 MDR (Message Detail Recording) provides the means for recording call information pertaining to incoming and outgoing trunk **calls.** As the trunk calls proceed through the system, MDR obtains various pieces of information about the calls. MDR analyzes this information and stores the results in a call record block. The call record block can then be sent to one of several terminal devices. Call record blocks can be output via one port of the FB-20992-A NSDC (Narrow Serial Device Control) card.

The MDR output may be in ASCII or GTE-modified EBCDIC. The information may be delivered in real time (i.e., each call reported as it is finished), or the call records may be saved for a period of time (elected in the data base) and delivered in a batch.

The MDR output may be delivered to a TTY, a minicomputer, a recording/mass storage unit, or a modem (which will connect to a remote device).

Hotel/Health Care Subprogram

34.1.14 The Hotel/Health Care subprogram handles the following device and feature operations:

- Wake-up (timed reminder) processing
- Message waiting (message center) processing
- Room-to-room blocking processing
- Room restriction processing
- Maid service processing
- H/HC feature activation/cancellation via telephone
- H/HC feature activation/cancellation via Attendant Console
- KEDU (Key Entry Display Unit) processing
- H/HC printer processing
- Message metering processing
- DND (Do Not Disturb) processing
- CLR (Combined Line and Recording) trunk operation
- H/HC dynamic data back-up and reload

Integrated Featurephone Directive Command Handling Subprogram

34.1.15 This Integrated Featurephone subprogram handles all Integrated Featurephone related functions. The CPU responds to the program, performing the necessary processing and translations associated with all Integrated Featurephones connected to the system, controlling the CIP (Control Interface Processor) data ports, and sending the appropriate commands to the Integrated Featurephone. The following related Integrated Featurephone activities are perforrned in this subprogram:

- New call command queuing
- Status command distribution (idle, ringing, busy, hold, etc.)
- Reset command distribution (line or DSS appearances)
- Directory number verification (line and intercom)
- Status keeping (line, hold, privacy, and waiting call count)

Integrated Featurephone Response-Event Handling Subprogram

34.1 .16 This Integrated Featurephone subprogram processes responses received from the connected Integrated Featurephones. Cross-correlation between Integrated Featurephones, their associated line complements, and the system line-directory numbers is performed, and the event is sent to the CPU. This handling subprogram also initiates control of associated Integrated Featurephone functions, such as commands returned or sent to other appearances, as well as the updating of privacy, status and new call queue tables. The typical response handling performed by this subprogram are the following types of Integrated Featurephone-initiated activities:

- Accept/reject
- Account code
- Answer
- Call hold
- Call retrieve
- Forward
- Off-hook/on-hook
- Privacy

System Data Base

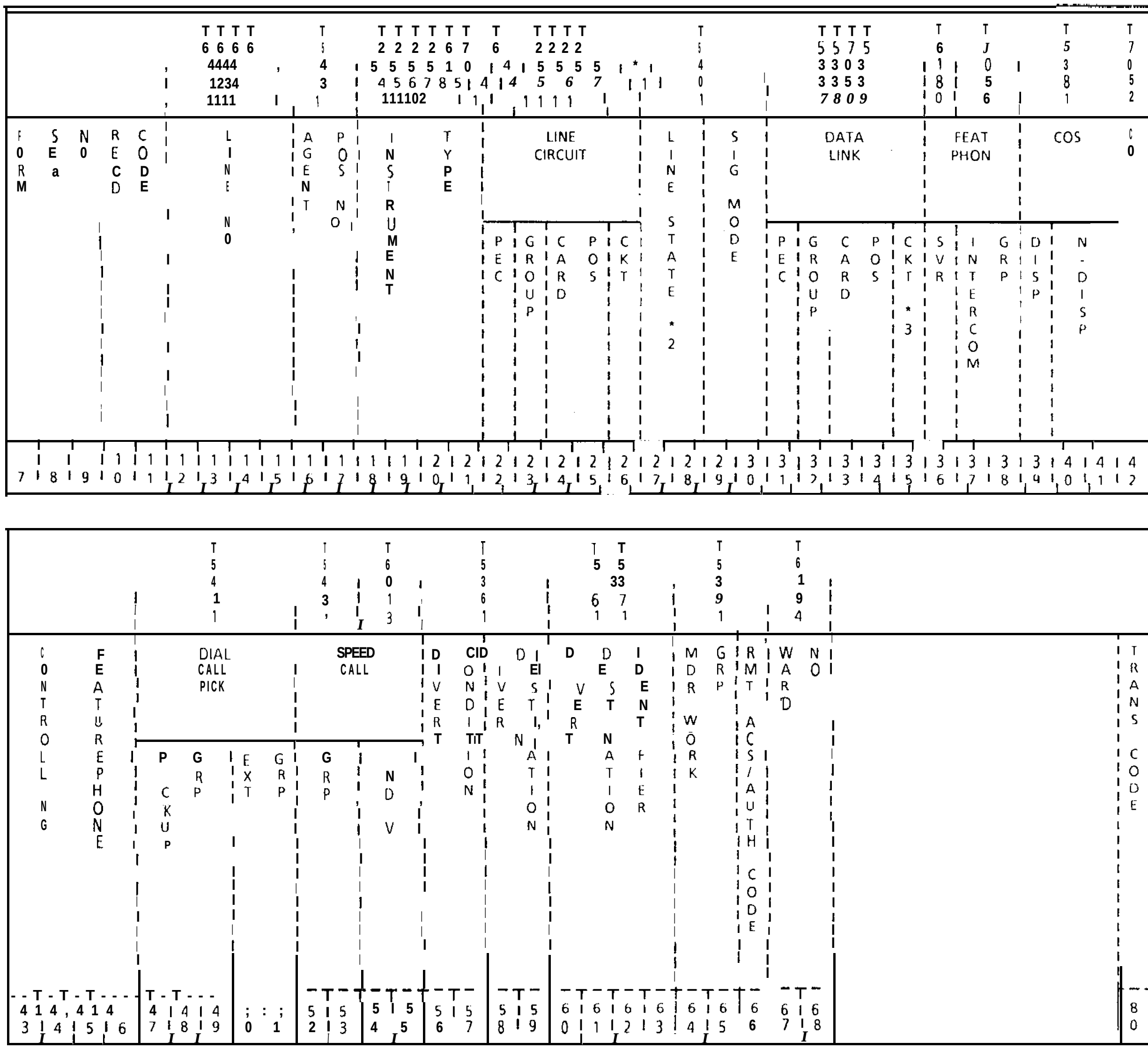
34.2 The system data base contains all the customer-unique information (such as the amounts and types of lines, trunks, station features, and hardware items) that are a part of the system. The data base is arranged in system memory as a series of tables. The tables are designated with the letter T and identified by a four-digit number (T-XXXX).

A hard copy printout of all the T tables (data base) is supplied with each system. The hard copy provides the system memory address of each table and is a printed record of the data contained for that system. Information in the tables can be changed as needed when features are added or deleted or when the system hardware is modified (telephones added, etc.). Changes to the system data base must be routinely noted in the listing so that references to the data base are accurate. TL-130400-l 001, describes how information is initially prepared for the tables. TL-130200-1001 describes how the contents of the tables are altered (Recent Change) after they have been written into system memory.

Data Base Memory Listings

34.2.1 System configuration and features are set up when data base programming is performed. Data base programming sheets (Record Codes) contain information that must first be organized into various groupings and then loaded onto a floppy disk. These groups of data are identified as tables, which are designated with a T and identified by a 4-digit number. These groupings are then loaded into specific addresses in system memory.

For example, refer to the Line Data (LD) Record Code Assignment Data Sheet (Figure 34.1) and note that above the file and card position entry header (columns 18, 19, and 20) are the table references T2541, T2551, T2561, and T2571.



NOTES:

* 1 = USED IN DETERMINING SOFTWARE ID WHICH IS PLACED IN T6441, T7054, T5330.
* 2 = USED IN FILLING TABLE TJ055.
* 3 = USED IN DETERMINING FEATUREPHONE DATA LINK NUMBER.
MAXIMUM NUMBER OF RECORDS = 256.

Figure 34.1 Line Data Record Code Assignment Data Sheet

Now refer to the hard copy of the customer Memory Tables listing. The Table of Contents (Table 34.1) in the listing identifies these tables as the Line Card Address Tables for PEC 0 and also shows that they are contained on Data Page 0.

Next, look up the table in the actual listing (Table 34.2), for example T2541 (PEC 0). Note that each table in the listing consists of two parts, a page with the descriptive details of the table (header), followed by the page (or pages) of the actual address structure and the data contained in each address (resulting from the data base programming).

The descriptive information indicates that "This table provides the hardware position of each equipped line card, determines the order of line scanning, and establishes the SOFTWARE ID for every line in Peripheral Equipment Complex 0." Each bit position representation in each word or byte is also described, which makes it possible to interpret or read the ones and zeros of the digital data. For instance, bits 3 and 2 are used to specify in which file a line card is installed, where file A is represented by 00 and file B is represented by 01.

The listing also shows that the line software ID numbers begin with 0, 8, 16, etc., through 248. A note in the descriptive information describes how to determine a line software ID. First, find that line's card position and file in the table; then, take that location's partial software ID (under entry number) and add that line's circuit number to it to calculate its software ID. Thus, with 8 circuits on each line card, these 32 entries provide the system with the software ID (O-255) for 256 lines.

For any line software ID, the memory contents (in binary) can be found and interpreted using the descriptive information sheet. The hexadecimal address, which is used in the microprocessor program to find the location in memory of this information, is also provided. In normal operation a request for a "read" from the maintenance TTY would reference this address. The hex entry equivalent of the binary contents is also provided since the maintenance TTY outputs the memory contents in hexadecimal. For convenience purposes, each page of the listing also provides a conversion table. When a change to the binary contents is to be made, the hexadecimal equivalent can be determined for each entry using the maintenance TTY.

Table T2581 is used for the TC (Trunk Circuit) Record Code, where the file, card position, and circuit columns refer to trunks located in PECO. When referring to the descriptive information sheet for this table and comparing it to the line data, it becomes apparent that the memory contents layout is the same, except that now bits 1 and 0 indicate the actual trunk circuit on the card.

The use of T tables by the system CPU now becomes more apparent. As described earlier, the line and trunk circuits are scanned by the MPB85 cards in the same file; therefore, the card position and file must be known in order to perform the scan routines for lines and trunks. The CPU processes all calls using software IDs. When a request for service is detected during a scan, the system receives sensing data, which is then compared with data in the T tables as a means of searching for a match. When a match occurs, the software ID is established. If the request for service was a station, the CPU can take the software ID, go to T5381 (LD (Line Data) Table) to determine the N-displayable and displayable COS (Class-of-Service) group that this software ID (station) is in. T5381 provides the index number (O-15) for T6091 (N-displayable COS, Record Code NC) and the index number (O-9) for T6101 (Displayable COS, Record Code DC) where the CPU can find all the specific COS assignable features permitted/denied for the line requesting services.

To summarize, the Data Base Programming/Assignment Data Sheets information is used to generate on ODDB (Office Dependent Data Base) floppy disk and listing (either a specific customer's data base or the FGBS standard data base). The system's generic program and the ODDB are loaded into the system, which places all this data in specific memory addresses where groupings of data are designated as T tables. See Tables 34.3 and 34.4.

Table 34.1 **Office-Dependent Data Base Listing Table of Contents**

TABLE OF CONTENTS

CEC DATA PAGE 0

TABLE NUMBER	TABLE NAME	START ADDRESS
T2541	PEC-0 LINE CARD ADDRESS TABLE	1010
T2581	PEC-0 TRUNK CARD ADDRESS TABLE	1031
T2621	PEC-0 CONSOLE CARD ADDRESS TABLE	1073
T2661	PEC-0 DTMF RECEIVER CARD ADDRESS TABLE	1075
T2701	PEC-0 AIOD CARD ADDRESS TABLE	107D
T2741	PEC-0 CONFERENCE CARD 1 ADDRESS TABLE	107E
T2742	PEC-0 CONFERENCE CARD 2 ADDRESS TABLE	107F
T3111	PEC-0 RLT CARD ADDRESS TABLE	1080
T3151	PEC-0 RLT TRUNK TYPE TALE	1090
T3201	PEC-0 KEDU ADDRESS TABLE	1 0A0
T3202	PEC-0 PRINTER ADDRESS TABLE	1 0A2
T3203	PEC-0 PRINTER BAUD RATE AND PARITY TABLE	1 0A4

Table 34.2 Table T2541

ENTRY TYPE	ENTRY NO. DEC	(HEX)	WORD NO.	CONTENTS BINARY	ADDRESS HEX	HEX ENTRY	
				6 7 5 4 3 2 1 0		1	2
LINE SOFTWARE ID NO.	0	(0)	0	0 0 0 1 0 0 0 0	1010	1	0
LINE SOFTWARE ID NO.	8	(8)	0	0 0 0 1 1 1 0 0	1011	1	C
LINE SOFTWARE ID NO.	16	(10)	0	0 0 0 1 1 0 0 0	1012	1	B
LINE SOFTWARE ID NO.	24	(18)	0	0 0 1 0 1 1 0 0	1013	2	C
LINE SOFTWARE ID NO.	32	(20)	0	0 0 1 0 1 0 0 0	1014	2	B
LINE SOFTWARE ID NO.	40	(28)	0	0 0 1 0 0 1 0 0	1015	2	4
LINE SOFTWARE ID NO.	48	(30)	0	0 0 1 0 0 0 0 0	1016	2	0
LINE SOFTWARE ID NO.	56	(38)	0	0 0 1 1 1 1 0 0	1017	3	C
LINE SOFTWARE ID NO.	64	(40)	0	0 0 1 1 1 0 0 0	1018	3	B
LINE SOFTWARE ID NO.	72	(48)	0	0 0 1 1 0 1 0 0	1019	3	4
LINE SOFTWARE ID NO.	80	(50)	0	0 0 1 1 0 0 0 0	101A	3	0
LINE SOFTWARE ID NO.	88	(58)	0	0 1 0 0 1 1 0 0	101B	4	C
LINE SOFTWARE ID NO.	96	(60)	0	0 1 0 0 1 0 0 0	101C	4	B
LINE SOFTWARE ID NO.	104	(68)	0	0 1 0 0 0 1 0 0	101D	4	4
LINE SOFTWARE ID NO.	112	(70)	0	0 1 0 0 0 0 0 0	101E	4	0
LINE SOFTWARE ID NO.	120	(78)	0	0 1 0 1 1 1 0 0	101F	5	C
LINE SOFTWARE ID NO.	128	(80)	0	0 1 0 1 1 0 0 0	1020	5	B
LINE SOFTWARE ID NO.	136	(88)	0	0 1 0 1 0 1 0 0	1021	5	4
LINE SOFTWARE ID NO.	144	(90)	0	0 1 0 1 0 0 0 0	1022	5	0
LINE SOFTWARE ID NO.	152	(98)	0	0 1 1 0 1 1 0 0	1023	6	C
LINE SOFTWARE ID NO.	160	(A0)	0	0 1 1 0 1 0 0 0	1024	6	B
LINE SOFTWARE ID NO.	168	(A8)	0	0 1 1 0 0 1 0 0	1025	6	4
LINE SOFTWARE ID NO.	176	(B0)	0	0 1 1 0 0 0 0 0	1026	6	0
LINE SOFTWARE ID NO.	184	(B8)	0	0 1 1 1 1 0 0 0	1027	7	B
LINE SOFTWARE ID NO.	192	(C0)	0	0 1 1 1 0 0 0 0	1028	7	0
LINE SOFTWARE ID NO.	200	(C8)	0	1 1 1 1 0 0 0 0	1029	F	0
LINE SOFTWARE ID NO.	208	(D0)	0	1 1 1 1 0 0 0 0	102A	F	0
LINE SOFTWARE ID NO.	216	(D8)	0	1 1 1 1 0 0 0 0	1028	F	0
LINE SOFTWARE ID NO.	224	(E0)	0	1 1 1 1 0 0 0 0	102C	F	0
LINE SOFTWARE ID NO.	232	(E8)	0	1 1 1 1 0 0 0 0	102D	F	0
LINE SOFTWARE ID NO.	240	(F0)	0	1 1 1 1 0 0 0 0	102E	F	0
LINE SOFTWARE ID NO.	248	(F8)	0	1 1 1 1 0 0 0 0	102F	F	0
SPECIAL ENTRIES			0	1 1 1 1 1 1 1 1	1030	F	F

CONVERSION TABLE A

DEC	BIN	HEX
0	0000	0
1	0001	1
2	0010	2
3	0011	3
4	0100	4
5	0101	5
6	0110	6
7	0111	7
8	1000	8
9	1001	9
10	1010	A
11	1011	B
12	1100	C
13	1011	D
14	1110	E
15	1111	F

Table 34.3 T2541 Description

```
* * * * * * * * * * * * * * * * * * * * * * * * * * * * * *
* T2541 - PEC-0 LINE CARD ADDRESSTABLE                       .
* * * * * * * * * * * * * * * * * * * * * * * * * * * * * *
THIS TABLE PROVIDES THE HARDWARE POSITION OF EACH EQUIPPED LINE CARD, DETERMINES THE ORDER OF LINE
SCANNING, AND ESTABLISHES THE SOFTWARE ID FOR EVERY LINE IN PERIPHERAL EQUIPMENT COMPLEX 0
LOCATION:          CEC DATA PAGE 0 FOR THE SI SYSTEM.
                   COMMON MEMORY FOR PEC-0 FOR THE S3 SYSTEM.
TABLE SIZE:        33 BYTES
TABLE LAYOUT:      32 IDENTICAL 1 BYTE BLOCKS WITH A HEX FF END OF TABLE MARK IN THE LAST BYTE.
BLOCK INDEXING.    THE LINE CARD ADDRESSES ARE ACCESSED BY AN INDEX INTO THE LINE CARD ADDRESSES
                   TABLE   THIS INDEX IS THE LINE CARD ADDRESS NUMBER.
HEX FILL VALUES FOR UNUSED BYTES. BYTE 0-F0
* * * * * * * * * * * * * * * * * * * * * * * * * * * * * *
LINE CARD ADDRESS ND 0                        *
       | 7 | 6   5 | 4 | 3 | 2 | 1 | 0 |      * CARD SLOT       :  SLOT #--PHYSICAL LOCATION
       +---+---+---+---+---+---+---+---       * VALID RANGES FOR *   OOOO=O 0100=4 1000=8
       |               |               |      *  SI  o-11        *   0001=1 0101=5  1001=9
BYTE   |   CARD SLOT   | GROUP  LINE TYPE     *  S3  O-10        *   0010=2 0110=6  1010-10
0      |               |      I        |      *                  *   0011=3 0111=7  1011-11
       +---+---+---+---+---+---+---+---+      *
                       *                      *  GROUP                GROUP       #--PHYSICAL
                       *                      *                       LOCATION
                       *                      .                       00 = A
LINE CARD ADDRESS NO 31                       .                       01 =B
                       .                                              10 = C
                                                                      11 = D
       +---+---+---+---+---+---+---+---       *
BYTE   |   CARD SLOT     GROUP     L I N E |  *  LINE TYPE            00 = POTS   POTS PHONE
0      |               I         I  TYPE   |  .                       01 = FEAT  FEATUREPHONE
       ---+--t--,--+--+--i--+--            *                       10 = LOGL   LOGICAL LINE
SPECIA |         END OF TABLE              |
ENTRY  | 1   1   1   1   1   1   1   1     |  *  END OF TABLE MARK-THIS SPECIAL ENTRY IS ALWAYS
                                              *  SET TO
                                                               FF
* * * * * * * * * * * * * * * * * * * * * * * * * * * * * *
```

THE LINE SOFTWARE ID IS DETERMINED BY THIS TABLE. TO DETERMINE A LINE SOFTWARE ID FIND THAT LINE'S CARD SLOT AND GROUP IN THE TABLE. TAKE THAT LOCATION'S PARTIAL SOFTWARE ID (UNDER ENTRY NUMBER) AND ADD THAT LINE'S CIRCUIT NUMBER TO IT TO CALCULATE ITS SOFTWARE ID.

THE MUSIC ON HOLD CIRCUIT IF PRESENT IS TREATED IN AN IDENTICAL MANNER AS A LINES PHYSICAL LOCATION.

THE PHYSICAL LOCATION OF LOGICAL LINES MUST BE AS FOLLOWS: GROUP A, CARD SLOT 00.

Table 34.4 Party Identifer/Circuit Number

PCMUS 4, FILE A	HARDWARE IDENTIFICATION NUMBER	EQUIPMENT INTERFACE CIRCUIT NUMBER
Calling Party	24	4
Called Party	26	6

Processing a Call 342.2 This paragraph describes the various tasks performed by the system when a call is processed. The example presumes that a station-to-station call is being processed and that particular pieces of equipment are used. (Table 34.4). Specific hardware items have been chosen so that descriptions of the system activities can also be illustrated in terms of the time-switch network memories and their data contents. Illustrations are presented to show the changes in memories that occur to effect each task in call processing.

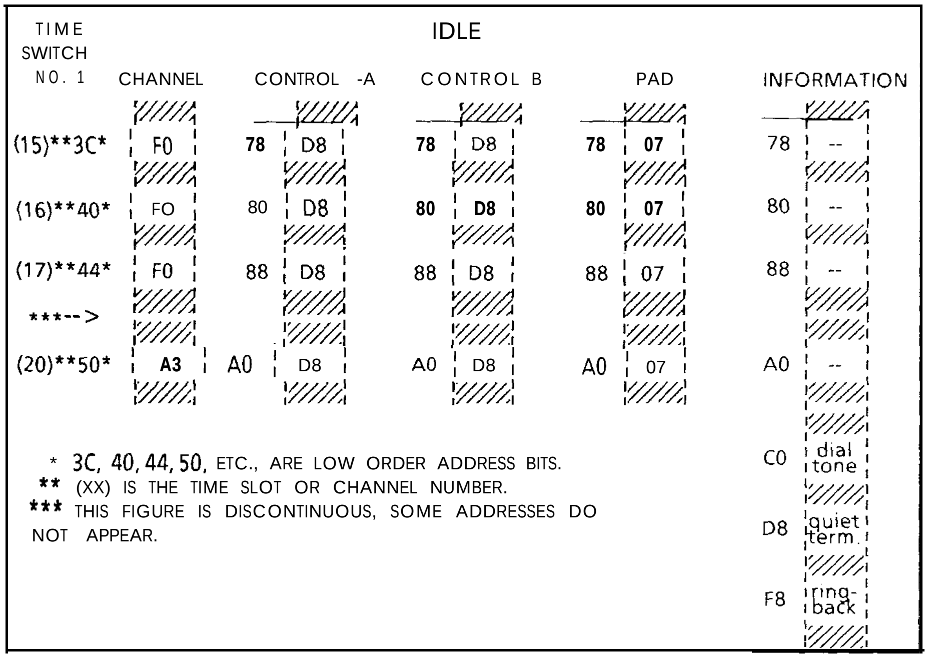


Figure 34.2 Time-Switcn Memories During Idle

Figure 34.2 shows the status of the memories during an idle condition, no calls are in progress and no calls are attempting to get started. In channel memory, the FO indicates idle status. The A3 indicates a time slot that is dedicated to a PCM DTMF receiver. In control memory, D8 is a special information memory address that contains quiet tone.

In pad memory, the 07 indicates idle, no attenuation factor is selected and interconnect memory is not in use (see word format in TL-130200-1001). In information memory, there is no data because no calls are in progress. The contents of control memory B will always be the same as control memory A because three-way calls are not included in this description.

An arrow head (>) is used in the memory illustrations to indicate the changes in content. For example, FO > 24 means that channel memory address has changed from idle status (FO) and now contains the hardware identify number (24) assigned to a piece of equipment.

The CPU routinely executes a 100-millisecond slow scan of sense points on all the line, trunk, and feature equipment interface cards. All circuits on equipment interface cards have request-for-service sense points. Some equipment interface cards such as trunk, DTMF receiver, Attendant Console, and some feature cards have more than one sense point.

Call Request for Service

34.2.3 The following sequence of events occurs when a subscriber goes off-hook, or requests service from the system by some other means (e.g., depresses LOOP pushbutton switch on Attendant Console).

- Subscriber goes off-hook. This signals his line, trunk, or equipment interface card that service is requested. This request causes the sense point associated with that line to be conditioned to the request-for-service state.

- The slow-scan routine consists of the CPU sending address inputs to the FB-17215-A MPB85 (Multiprocessor Buffer) card. In response, the MPB85 card produces equipment interface select (card select) outputs that systematically enable (scan) the equipment interface cards, one at a time. The MPB85 card also produces circuit select outputs that systematically enable the sense point circuit(s) on the enabled equipment interface card.

- Each time an equipment interface card is scanned, the level on the sense point output(s) from the selected equipment interface circuit(s) on the equipment interface card is applied as input(s) to the MPB85 card. The line interface card is enabled by one equipment interface select (card select) from the MPB85 and does not require the use of circuit address inputs. Up to eight sense points representing a digital word are read from equipment interface cards. Each bit in a digital word may indicate a particular type of request, e.g., seizure request, or feature initiation request.

- The CPU routinely addresses the MPB85 card and reads up to eight sense point status inputs through the MPB85 card, onto the CPU data bus, and into the CPU.

- The CPU then examines the eight sense point status bit inputs to determine if the status changed on any of the sense points from the last time they were read. As an example, when a telephone connected to a line card is taken off-hook, the tip and ring input to the line card will change from an open condition to a closed condition. This causes the related seizure request input to the CPU from the MPB85 card to change from a high to a low level. This change is detected as a line seizure request for service by the CPU and is loaded into an event data message.

NOTE: The CPU actually checks the seizure request input twice (two 100-millisecond scans) before a request is verified.

- The CPU routinely loads event data messages into common memory on the MPB85 card. The CPU then reads these messages from each MPB85 card. This event data message contains the hardware identification for the equipment interface card circuit or circuits (ports) that produced the request for service. It also contains the type or condition that produced the request (i.e., identifies that the request was produced by an on-hook to off-hook condition from line interface card X; where X = hardware ID assigned to that line card within the data base); also, it identifies whether or not the request was produced by an attendant depressing a LOOP or feature pushbutton on the Attendant Console.

- The CPU checks the event data message to determine the type of request that was generated (e.g., on-hook to off-hook, feature initiation request, etc.).

- The CPU enters the Line-Record-Code Check routine.

Line Record Code Check

34.2.4 After receiving a request for service, the system CPU checks the line (or trunk) record code tables assigned to the call originator's line (or trunk). Data base line record data tables (resident in system CPU RAM) define to the system CPU exactly what features are assigned to the originator's line including the type of subscriber's equipment (i.e., DTMF telephone, DP telephone, or MIXED (where mixed = DTMF or DP telephone) or NONE (where none = Not DTMF, and not DP, and not MIXED).

After checking the line record code table, the CPU enters the time slot assignment routine.

Time Slot Assignment

34.2.5 When the system CPU detects a request for service in an event data message, it looks for an available time slot to assign to the requesting interface card.

The following sequence of.events occurs when the CPU receives a request for service from an equipment interface card:

- Based on the equipment interface card hardware identification number in the event data message, the CPU examines tables (data base) in CPU memory that give the software identity for the equipment file in which the equipment card is located.

- The CPU then checks "time slot busy/idle status" tables in the CPU memory. These tables indicate which information memory locations are currently available within the block of 24 information memory locations allocated to each associated PCMUS group. Information memory for the system is located on the EPCMN (Expanded Pulse Code Modulation Network) card. By continually updating the time slot busy/idle status tables, the CPU keeps track of the number of time slots that are available or busy within each block of 24 information memory locations (groups).

- If an information memory location (time slot) is not available, the originating party's line remains silent until a time slot becomes available.

- If an information memory location (time slot) is available, the CPU loads the hardware identification number (of the caller) into the channel memory address associated with (one-to-one correspondence) the available information memory (time slot). This action assigns that time slot to that particular hardware. The CPU then performs the following functions:

  - The CPU writes into the control memory address of that time slot. The data written into control memory address is an information memory address that is listened to by the originating party; first for dial tone and, as will be explained later, then another information memory address for voice or data from the called party. The CPU reserves this time slot for the duration of the call or feature implementation.

- The CPU loads an attenuation factor address into the time slot associated address in PAD memory on the EPCMN card. The contents of this PAD memory address are used by the PROM PAD attenuator chip on the EPCMN card, which provides the proper attenuation as the system transmits (PCM) tones or voice to the originating party or called party. The attenuation factor address bits that are loaded into PAD memory are based on the CPU's determination of the type of line or trunk to which the originating party is connected. These addressing bits determine the dB level of the signal out of the PROM PAD to the originating party or called party. The dB levels are different for different types of lines or trunks. The type of line or trunk was determined during the line record code check.

- The CPU routinely reads the common memory on the MPB85 card and locates a directive to load the equipment interface ID number into the time slot associated address (one-to-one correspondence) in channel memory. The contents of this address will be used to enable the equipment interface card when the system begins to transmit dial tone to the equipment interface card or begins to sample voice, digital, or DTMF information from the equipment interface card (explained later in this call-processing description).

- The CPU also loads that equipment interface hardware identification number into a fast-scan table.

- The CPU now proceeds to scan the requesting interface card at a fast scan rate (1 O-millisecond cycle time). Any other interface card that was previously assigned a time slot is also being scanned at the fast scan rate. The fast-scan sequence is sandwiched with the slow-scan sequence, which is still being performed on those interface cards that have not requested service.

- If the line record code check determined that the request was for line or trunk seizure and the originator's line or trunk is assigned to the the DTMF or MIXED signal mode, the CPU enters the DTMF receiver assignment routine. If the originator is not assigned to DTMF or MIXED service, the CPU enters the transmit dial tone routine.

DTMF Receiver Assignment Routine (Except Featurephone)

34.2.6 This routine is only executed if the line record code check performed by the CPU determines that a line or trunk requesting service is assigned to a DTMF or MIXED signal mode.
If the subscriber has a DTMF telephone, the DTMF tones produced when the subscriber keys in a number must be converted to digital bits that the system computer can interpret. This is done by a DTMF receiver. Therefore, if the subscriber has a DTMF telephone, a DTMF receiver must be assigned to receive the DTMF tones.

The DTMF receivers (just as any other peripheral equipment) are assigned time slots and have sense points that are scanned by the CPU. A dedicated time slot is assigned to a DTMF receiver, so that the receiver's hardware identification number is always present in a time slot's channel memory. Thus, the information memory address of the subscriber's time slot is put into the receiver's control memory so that the receiver can listen to the subscriber tones when dialing. As memory sampling occurs (time-switch network control), the DTMF receiver sees PCM samples (coming through the time-switch network from the subscriber) and puts the samples on the digital data link (sense/read) to the CPU through the MPB85 card.

A fully configured system can support eight DTMF receivers. Trunks or lines supporting DTMF telephones share the DTMF receivers. A DTMF receiver that has been assigned to a DTMF input is busy during the time that it has been assigned to receive DTMF tone(s) from a particular subscriber. Thus, when a request-for-service is received from a DTMF telephone, the system CPU must check for the availability of a DTMF receiver before transmitting dial tone (permission to dial) to the calling party.

Dial Tone to Subscriber

34.2.7 After the system has determined that a time slot is available and has assigned the time slot to the subscriber, it must make a path from the dial tone source to the requesting interface card, and then to the calling party. Special extra addresses in information memory are used to store digital data samples (PCM) of all the various tones used in the system. These extra addresses are updated regularly with samples from the tone source card (FB-20974-A PCMTS). Dial tone will notify the calling party that the system is working, that the calling party's request-for-service has been recognized, and that permission has been granted to dial a number.

The following sequence of events occurs for dial tone to be transferred to the subscriber's equipment after a request-for-service has been recognized, and a time slot has been assigned to the equipment interface card (it shall be assumed that the dial tone is to be transferred to a line interface card connected to a telephone).

- The Channel Memory card produces binary-coded outputs that are decoded into a discrete equipment interface select output by the FB-17189-A PCMFS (Pulse Code Modulation Frame Sync) card. The PCMFS outputs systematically enable the requesting equipment interface card circuits, one at a time, in the file for a period of approximately one microsecond.

- The CPU accesses the control memory of the subscriber's time slot and prepares to write an information memory address into control memory. The address now in control memory is one of the special, extra information memory addresses that contains the quiet code. For this particular case (dial tone), the control memory must contain the address of information memory that has dial tone (another special, extra information memory address).

- The CPU, therefore, writes into the subscriber's control memory, the information memory address that contains dial tone. This action connects the subscriber to dial tone (Figures 34.3 and 34.4).

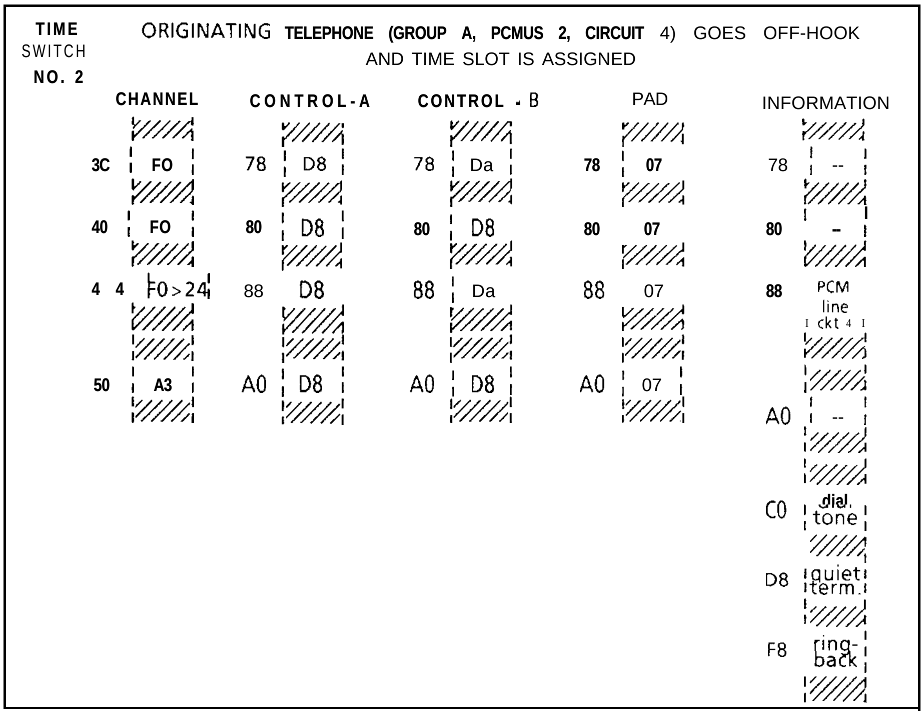


Figure 34.3 Off-Hook Time Slot Assignment

- The INCKS (Synchronizable Intermediate Network Clock) card generates control signals which produce a dial tone select input to the PCMTS (Pulse Code Modulation Tone Source) card. This action reads digital dial tone data (PCM) out of PROM storage on the PCMTS card. Under control of a write signal from the INCKS card, dial tone PCM data is written into the special information memory location assigned to dial tone. Similarly, digital samples (PCM) of other tones and signals that are used by the system are taken from the PCMTS card and written into special information memory locations. The samples are updated every 125 microseconds and can be connected to any subscriber simply by writing the appropriate information memory location into the control memory of the subscriber's time slot.

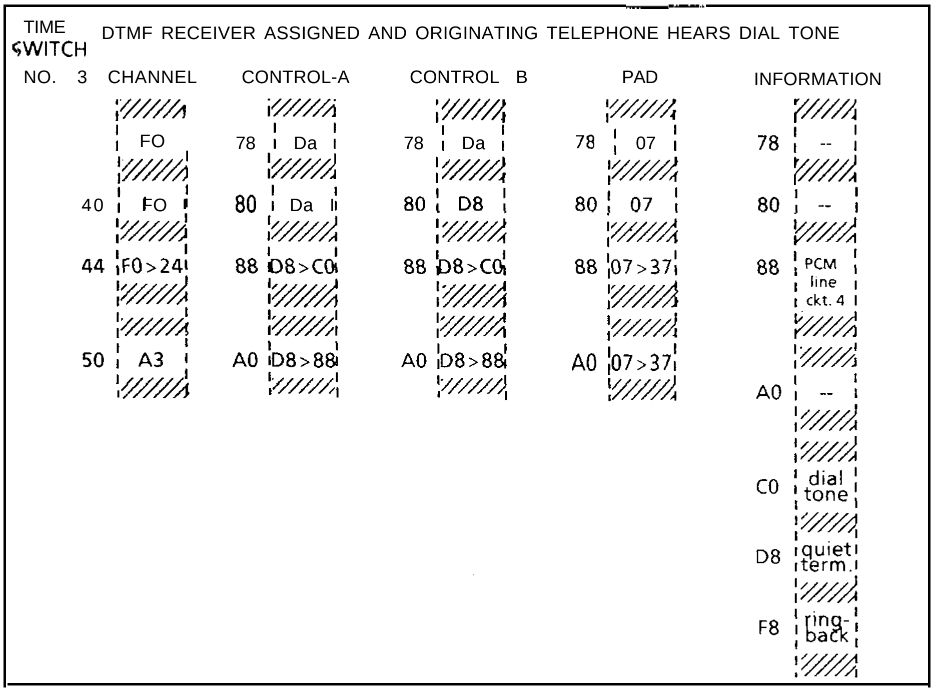


Figure 34.4 DTMF Receiver Assignment and Dial Tone

- After dial tone is connected to the subscriber, a read control signal is activated from the INCKS card. This causes dial tone PCM to be read out of information memory and to be transferred through the PROM PAD to interface circuits on the CHM85 (Channel Memory) card.

- The dial tone PCM data is transferred through the interfacing circuits on the FB-17218-A (Channel Memory) card and onto a common bus that feeds the various conversion devices used in the system (e.g., PCM T1 Buffer and Span Interface). The conversion device that is enabled at this point depends on the channel addressing input to the Channel Memory card.

- The dial tone PCM data is converted to analog by the line interface card associated with the telephone that was taken off-hook. The line interface card is receiving an enable from the PCMFS card under control of the Channel Memory card.

- The analog dial tone is transferred through the line interface card onto the tip and ring leads that are connected to the subscriber's telephone. The subscriber hears dial tone. Dial tone notifies the calling party to start dialing the destination number.

- Dial tone is transmitted to the subscriber until the first digit is dialed or timeout occurs.

In addition to transmitting dial tone to the originating subscriber, the system checks the subscriber's class of service. Data base class-of-service tables (resident in system CPU RAM) define to the system exactly what features are assigned to the originator's line

Reception of DTMFDigits

34.2.8 The subscriber recognizes dial tone as a signal to begin dialing the desired number. The CPU prepares to accumulate and interpret the dialed digits. After the first digit is dialed, the CPU removes dial tone from the subscriber. The CPU then writes the address of the special information memory location that contains quiet code into the subscriber's control memory, thereby disconnecting the subscriber from dial tone (Figure 34.5). The sequence of receiving digits is as follows:

- The subscriber commences to key in the number to be called on the touch pad of the DTMF telephone. Each number keyed generates a dual-tone frequency output on the telephone's tip and ring input connection into the associated line interface card.

- The dual-tone frequency from the telephone tip and ring is converted to digital and is applied to a sample gate output on the line interface card.

- The Channel Memory card produces binary-coded outputs which are decoded by the PCMFS card into a discrete equipment interface select output. This discrete output from the PCMFS card enables this line interface card sample gate for approximately 1 microsecond, during which time the dual-tone frequency appears on the PCM bus. The output of the line interface card becomes a series of l-microsecond-wide digital pulses whose amplitudes vary in step with the dual-tone frequency inputs from the telephone.

- This output is routed through the time-switch Nntwork, and then is applied to the DTMF receiver.

- The fast-scan routine (executed every 10 milliseconds) sequentially samples for digit collection on all DTMF receiver channels (time slot) that have been made busy by a request-for-service.

- The DTMF receiver output is then applied to the MPB85 card and the MPB85 card transfers up to eight sense-point status inputs onto the CPU bus where is it finally applied to the CPU. During this digit collection routine, the CPU accumulates the dialed digits via the MPB85 card and executes a digit store routine (Figure 34.6).

- During a digit store routine, the CPU stores each digit in a scratch pad area of the CPU memory. The digits are stored in the digit store scratch pad only long enough for the CPU to identify the station being called (dialed). Then, this information is transferred to call store for the duration of the call. (The word format for digit stores and call stores is described in TL-130200-l 001.)

Reception Of DP Digits

34.2.9 The subscriber recognizes dial tone as a signal to begin dialing the desired number. The CPU prepares to accumulate and interpret the dialed digits and, after the first digit is dialed, the CPU removes dial tone from the subscriber. The CPU then writes the address of the special information memory location that contains quiet code into the subscriber's control memory thus, disconnecting the subscriber from dial tone. The sequence for receiving digits is as follows:

- The subscriber commences to rotary dial the number being called. As the rotary dial returns to the home position, it makes and breaks the tip and ring connection into the associated line interface card. Each make and break of the tip and ring connection creates one DP (Dial Pulse) into the line interface card. Dialing a one creates one pulse; dialing a nine creates nine pulses.

- Each tip and ring pulse received by the associated line interface card enables and disables a sense point output from a circuit on the line interface card. The dialed pulses cause the sense point output to vary at an 8 to 12 pulses per second rate.

  Note: The line interface card sense point used for DP digits is the same sense point used to input request-for-service from a DP telephone (i.e., current flow detected).

- Each time a line interface card is enabled, the level on the sense point output from the line interface card is applied to the FB-17215-A MPB85 (Multi-Processor Buffer) card. Each sense point is enabled (read) for a period of 1 microsecond.

- As well as executing the fast-scan routine every 10 milliseconds, which samples the DP digits, the CPU also routinely executes the DP digit collection routine for those DP telephone channels that have requested service and received dial tone. During the DP digit collection routine, the CPU counts the pulses and interdigital pauses read from the MPB85 card and routinely transmits the accumulated DP digits to the system CPU via the MPB85 card.

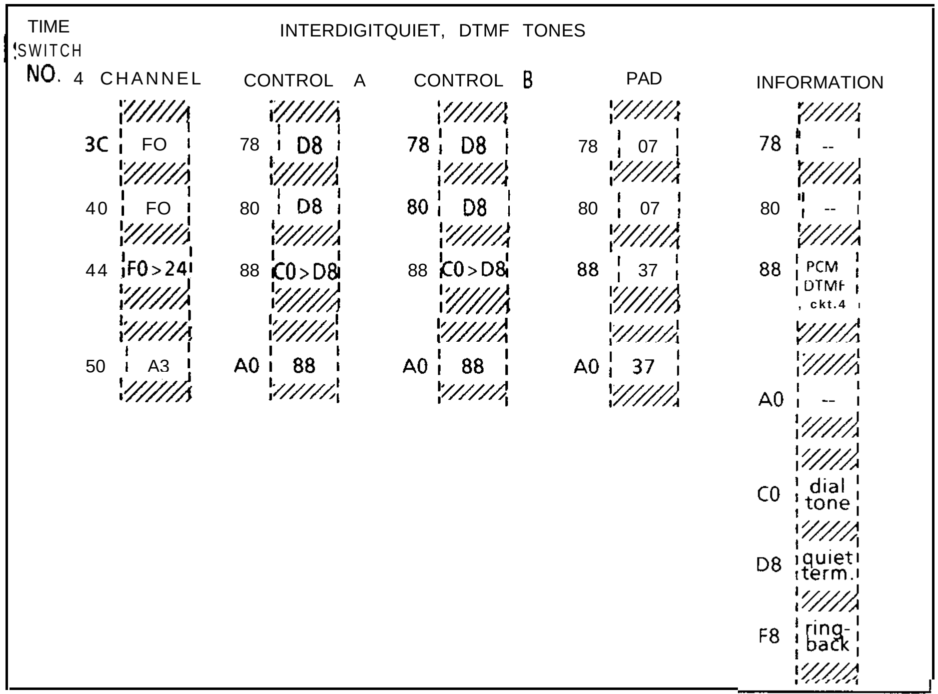


Figure 34.5 Interdigit and DTMF Tones

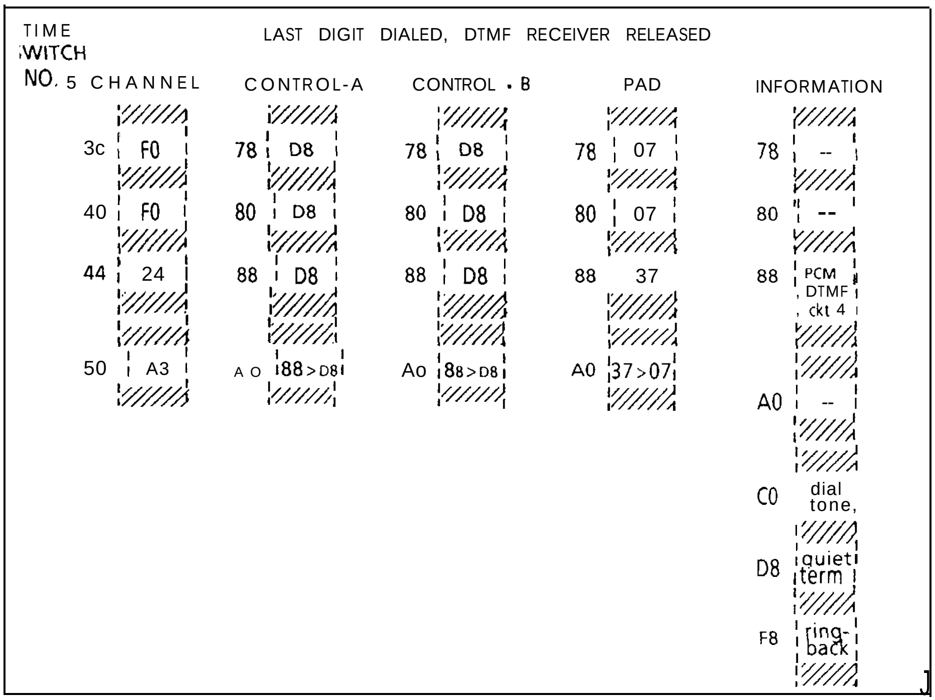
- By routinely executing a call store and the associated digit store routines, the system CPU accumulates the dialed DP digits by storing the digits in a scratch pad area of system CPU memory. The digits are stored only long enough for the CPU to identify the station being called (dialed). (The word format for digit stores and call stores is described in TL-130200-1001.)

Ringing and Ringback

34.2.10 After the CPU has received all the dialed digits, it will determine who is being called or how the call is to be routed. It also checks the subscriber's class of service (line record code data base tables) to determine if the subscriber is allowed to make the kind of call dialed. If the call is allowed, for instance station-to-station, the CPU determines whether or not the called telephone is busy, sends a ringback (or busy) signal tone to the originating subscriber, and then rings (if not busy) the telephone being called (Figure 34.7). The follows explaines the call processs:

- As the CPU routinely addresses the MPB85 card, it reads the sense points of equipment interface cards through the MPB85 card. The sense-point bits indicate the status of the equipment connected to the interface card being scanned.

- If the telephone being called is busy (off-hook), the CPU writes into the control memory of the originating subscriber's time slot the special information memory address that contains



Figure'34.6 Last Digit Dialed, DTMF Receiver Released

the busy signal tone. This action connects the originating subscriber to the busy signal tone.

- If the telephone being called is not busy, the CPU enters the time slot assignment routine, and assigns a time slot to the telephone being called by writing the hardware identification number of that telephone into the channel memory of the new time slot (Figure 34.8).

- The CPU now sends a control bit to the equipment interface card of the telephone being called, causing the telephone to ring.

- The CPU now writes into the control memory of the originating subscriber's time slot the special information memory address that contains the ringback signal tone.

Answering a Call

34.211 When the subscriber being called answers the telephone, the system CPU detects the off-hook action. The system then stops ringing the telephone.

The system CPU also writes into the control memory of the originating subscriber's time slot; the information memory address of the subscriber (called the subscriber's time slot). Then the CPU writes into the control memory of the called subscriber's time slot; the information memory address of the

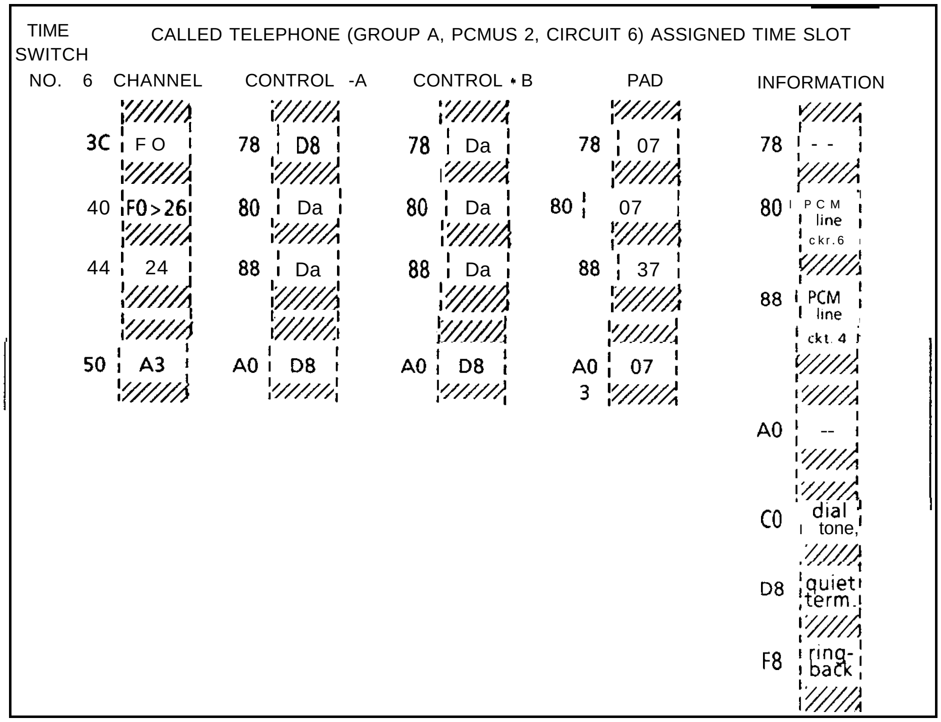


Figure 34.7 Called Telephone Assigned Time Slot

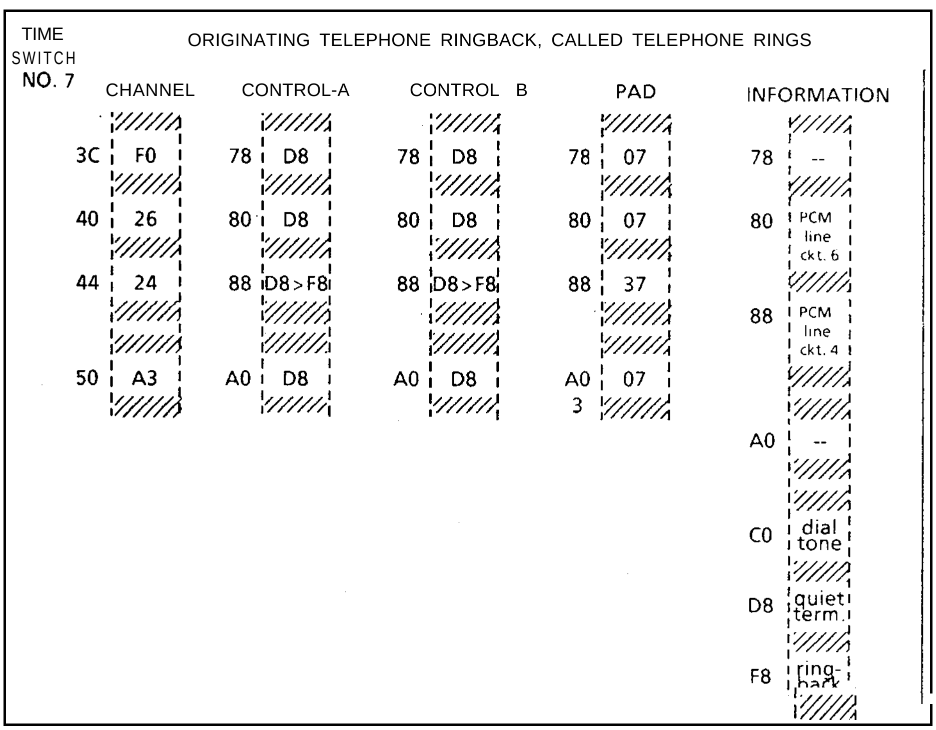


Figure 34.8 Ringback Sent to Originating Caller

originating subscriber's time slot. This action cross-connects the two time slots so that each is listening to the other.

Conversation

34.2.12 When the time switch is arranged as shown in Figure 34.9, the two parties can talk to each other, and the two time slots are in a stable call condition. The time switch (under control of the INCKS card) cycles through a regular scanning routine and looks at all of the system time slots, one at a time. During each look. four tasks are executed:

- Channel memory is read to obtain the hardware identification number (26) of the originating equipment and assigned to this time slot. A data sample is then taken from the equipment and placed into the time slot information memory (PCM) voice circuit.

- Control memory of the originating equipment time slot is read to obtain the address in memory (88) that the time switch reads next. It is the information memory address (of some other time slot) to which the call originator wants to listen.

- Information memory of the distant end time slot is read to obtain a data sample (PCM voice circuit 4). The data sample from the distant end time slot is sent to the originating equipment (26) time slot. Based on the exchange of data samples, the two parties are able to conduct conversation. The call remains "stable" until a change in call status occurs (e.g., one party hangs up or a hookswitch flashes).

- The fourth task involves writing data into channel memory or control memory. The time switch allocates the last part of its look at each time slot so that the CPU can write data into channel memory or control memory. This is only done when the status of a call is to be changed. (For example, when a time slot is assigned, the CPU must write a hardware identity number into channel memory.)

Termination of Call

34.2.13 The telephones are placed on-hook when the conversation has ended, and the following occurs:

- As part of the slow-scan routine, the CPU detects a change in status of the sense-point status bits related to the station equipment.

- The CPU then writes idle data (FO) into channel memory of each time slot (when that time slot is scanned) and then writes the special information memory address of quiet tone (D8) into the control memory of each time slot (Figure 34.10).

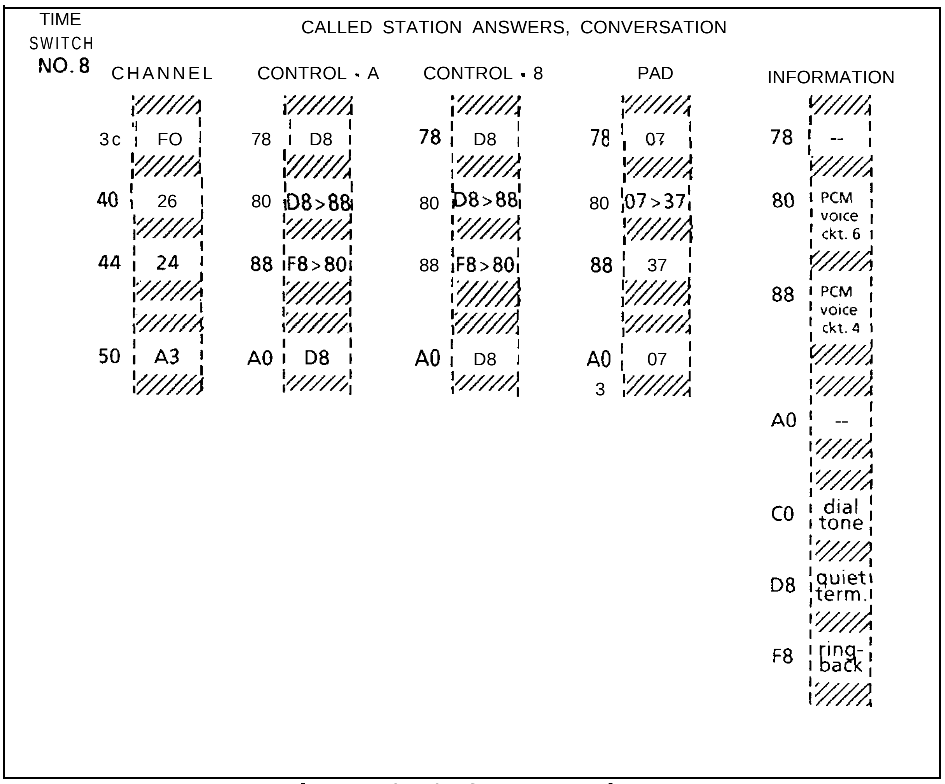


Figure 34.9 Conversation

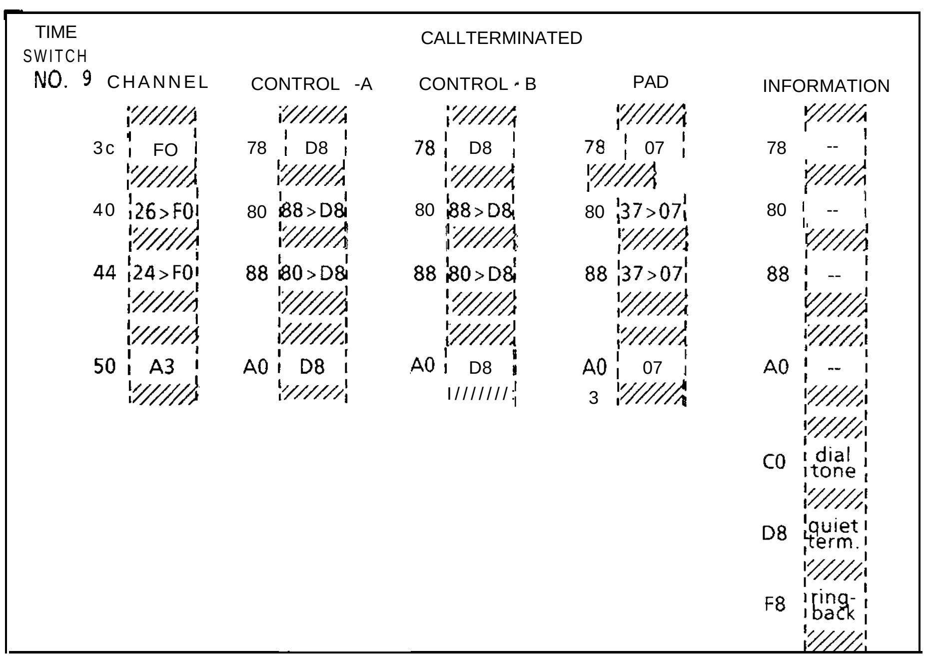


Figure 34.10 Call Terminated

THIS PAGE IS INTENTIONALLY LEFT BLANK.

Software Description

35.0 The software components used to provide the data system call and interconnect features are divided into the following categories:

- Executive class software
- Packet device handler software
- Data call-processing software
- Administration, control, and maintenance software

Software categories are implemented by using the techniques of compile time switching designed into the program code. All of the code is memory resident. The CHASM high-level language is used on all cards, except the ADMP-C card processor. The ADMP-C card processor is programmed in the language "C".

Executive Class Software

35.1 Executive class software is the core software that operates the processor based cards used for providing data features for all APMs, SPMs, NICs, UCBs/DCPs, and ADMPs. This software class contains the OS (Operating System), NETLINK, ROMs debugger, and diagnostics software. The executive class software performs the following tasks:

- Initiating and terminating data processes.
- Queuing and de-queuing messages.
- Sending and receiving messages to/from other cards.
- Allocating and deallocating buffer space.
- Providing for software debugging.
- Carrying out the low-level MPP (Mini-Packet Protocol) link protocol.

Operating System

35.2 The operating system is a compile time configurable software component that services multiple timed and/or event-driven tasks on two priority levels. It operates on single and/or dual 6502 compatible processor systems. The operating system also services the real-time clock and maintains an eight-bit multiple scale timer for each task running under its control.

ROM Debugger

35.3 The ROM debugger executes commands that provide the following:

- Read/modify memory
- Halt processing
- Single step
- Breakpoint software
- Dump error logs

Boot ROM

35.4 Boot ROMs reside on all RAM-resident cards for the purpose of downloading their code and tables. This download is performed at system initialization, system reconfiguration, and device installation. A common boot ROM is used for the APM, SPM, and UCB/DCP cards. It resides on the A processor. A boot PROM is used on the ADMP-C processor.

Netlink 35.5 The NETLINK is the software component that provides intercomponent communications across the packet transport system and central packet link. It resides, in various forms, on the APM, SPM, UCB/DCP, NIC and ADMP cards. NETLINK consists of three layers:

- Layer 1 (Physical Layer). This layer provides the software interface with the packet line interfaces and the packet transport system hardware.

- Layer 2 (Mini-Packet Protocol Layer). This layer implements the link layer protocol. This guarantees sequenced, error-free delivery of nontransparent control mini-packet traffic on each logical link. It also guarantees sequenced, error-free delivery on nontransparent user mini-packet data traffic.

- Layer 3 (Message Layer). This layer controls the mini-packet protocol layer and provides an interface to all higher level software components for sending and receiving X.25 packets and data control messages (APM and SPM only).

Superlink/ADMP 35.6 SUPERLINK resides on the ADMP and serves as an interface to transfer messages from NETLINK between the ADMP application tasks on the ADMP-C processor and NETLINK on the ADMP-A processor. Messages are transferred via inbound and outbound message transfer lists in the common data area. The inbound message transfer list is for messages from the C processor to the A processor, and the outbound message lists is for messages going in the other direction.

ADMPPECLINK 35.7 The ADMP-based PECLINK component receives ADMP-bound messages sent over the PEC bus, validates message ntegrity, processes assurance responses, and adds all other valid messages to the PEC-to-ADMP message queue. These messages are then retrieved and processed by the ADMP application tasks. Unknown commands are ignored by the ADMP application task. The PEG-bound messages from the ADMP application tasks are retrieved from the ADMP-to-PEC message queue and sent to the PEC again via the PEC bus. PECLINK will act as a black box for all PEC-bound messages, with the exception of he print event and device event messages,

File Input/Output

35.8 The ADMP file system interface subsystem, known as FIOS, provides an interface for the application tasks running on the ADMP-C processor to perform input and output operations on the disk subsystem. It receives the input/output requests from the application task, transforms them into file control transaction blocks, and relays them to the file management system processor across the SASI bus. A file control transaction block contains all information necessary for the file management system processor to execute the input/output operation. As each request is completed, the result of the operation is returned to the application task, again via FIOS. Data passing between the application tasks and the file management system is also transmitted via FIOS.

Packet Device Handler Software

35.9 Packet device handler software interfaces with the customer device and the PD-200 Data System and provides the protocol service necessary to support the devices.

Asynchronous Device Handler/APM

35.10 The asynchronous device handier software provides an interface with asynchronous data devices. Both DTE and DCE interfaces are supported. The asynchronous device handler resides in an APM card and provides an X.3 PAD function. This includes asynchronous devices to/from X.25 protocol translation and flow control.

X.25 Device Handler/SPM

35.11 The X.25 device handler software provides an X.25 interface. It resides on an SPM card and supports interfaces to X.25 DTEs and X.25 PDNs (Public Data Networks). LAPB (Link Access Procedure Balance) is supported with HDLC framing.

Pseudo Packet Device Handler/ADMP

35.12 The pseudo packet device handler is the ADMP component which provides an X.25 packet level interface between the administrative software and the external user to effect a data call. The pseudo packet device handler is compile time configurable to handle N logical channels as either outbound only or two-way. The pseudo packet device handler acts as a slave to the administrative software in the ADMP, allowing it to change pseudo packet device handler internal call states via pseudo packet device handler-supplied utilities.

Data Call-Processing Software

35.13 Call processing and switching refers to the setting up and tearing down of packet-switched data calls. This does not include voice or other circuit-switched data call setups. These calls may be any combination of X.25 end-point, asynchronous packet assembler/disassembler calls, and/or connections to or from the administrative ports built into the ADMP card. (These administrative ports are known as pseudo packet device handlers.) Call-processing software consists of data call processing, which resides on a UCB/DCP card, and packet device handling, which resides in the APM or SPM. The switch and command handler are part of the data call-processing software. The asynchronous packet assembler/disassembler (residing in the APM) or the X.25 device handler (residing in the SPM) are examples of pseudo packet device handlers. The switch, command handler, pseudo packet device handler, and other administrative software communicate with each other by using a set of well defined messages.

Administrative, Control, and Maintenance Software

35.14 Administration, control, and maintenance software includes the following utility software components:

- DMD (Device Message Distributor).
- ERMA (Event Report Message Administrator).
- UI (User Interface).
- TED (Table Editor).
- MAID (Maintenance and Interactive Diagnostics).
- FUTIL (File Utility).
- REAPER (Record Account Processing Event Records).
- DEAR (Dump Event and Accounting Records).
- Loader.
- CHANDLER (CEC Handler).
- ADMPINIT (ADMP Initialization).

DMD (Device Message Distributor)

35.15 The DMD (Device Message Distributor) knows the origin of a message and to which process to send it. Messages originating from the CEC (via PECLINK) or from data devices (via NETLINK, SUPERLINK) are distributed to the correct receiving process on the ADMP.

ERMA (Event Repot-t Message Administrator)

35.16 ERMA is the event report message administrator of the ADMP. This ADMP function receives and processes events from the data system devices. Event messages are generated from a device for many reasons, such as device malfunctioning, certain error condition reached, or garbled communications. ERMA is responsible for acting on events received by the ADMP. ERMA acts on the event by executing the appropriate primitives. These primitives may record the event as a printable message or as an ADMP statistic. The primitive may initiate another event due to threshold overflow or other internal conditions on the ADMP.

UI (User Interface)

35.17 The UI (User Interface) function of the ADMP is designed to perform menu presentation conventions for the other applications. It is the job of the UI to present menus to the user, screen and check data entered by users, perform security checking, and transmit finished data structures to applications.

TED (Table Editor) Data Recent Change

35.18 The TED (Table Editor) function of the ADMP allows users to dynamically view and optionally change the contents of many of the ADMP and device tables. TED deals in record structures, with menus designed to present the table data in user-readable form. TED will access a copy of a table, interact through the UI with the user to update the table, and then return the updated copy to disk, to an application, to a data system device, or to ADMP memory. The FIOS interface of TED is for updating tables to the disk.

MAID (Maintenance And Interactive Diagnostics)

35.19 The MAID (Maintenance and Interactive Diagnostics) function of the ADMP performs interactive maintenance functions beyond the existing CEC maintenance capabilities. The MAID is designed to interactively perform status read operations upon user demand. Some of these functions include loopback testing, taking down a call or an X.25 line, changing data set signals, viewing data system configuration and/or device status, etc. Since a debug code will exist in the ROMs of the data system devices, a special function of the MAID is to provide interactive debugging to users. A network debugging tool allows users to directly communicate with the ROM of a data device.

FUTIL (File Utility)

35.20 The FUTIL (File Utility) function of the ADMP allows users to copy disk files, interactively change the names of files, dump. delete, or perform disk maintenance functions.

REAPER (Record Account Processing Event Records)

35.21 The REAPER (Record Account Processing Event Records) accepts call accounting records and stores them on disk. This is performed in a similar manner to the event handler (ERMA), where records are stored until they are forwarded.

DEAR {Dump Event and Accounting Records)

35.22 The DEAR (Dump Event and Accounting Records) is the event and accounting record dumping mechanism. The DEAR is designed to accept an incoming X.25 call to dump records, or make an outgoing X.25 call to dump records to any terminal connected to the data system or to the RS 232-terminal. The DEAR operates by sequentially reading the event tracking file written by ERMA, and/or by reading the accounting tracking file written by the REAPER. The DEAR is the mechanism by which events and/or accounting records are forwarded to various devices.

Loader 35.23 The Loader is responsible for initializing and loading any data system device which requires code and/or tables. The loader accesses the disk through FIOS to read code or data records and transmits them to the device being loaded. Once a device is loaded, the success or failure of the load is reported to the ERMA event handler.

CHANDLER (CEC Handler) 35.24 The CHANDLER (CEC Handler) is an ADMP function which is solely responsible for dealing with requests from the CEC. The CEC may instruct the ADMP to make dynamic changes to the data system configuration or status, add or delete cards or lines, put a card or line out of service, load a device, or any other recent change or maintenance function. The CHANDLER is responsible for processing these requests and sending appropriate messages to other functions of the ADMP (such as the LOADER or ERMA).

ADMPINIT (ADMP Initialization) 35.25 The ADMP Initialization is a complex function which allows the ADMP to boot from the disk and begin processing. The ADMP ROM (Read-Only Memory) is responsible for loading the operational code. The operational code initialization step is responsible for starting all processes, and for initializing static data areas and files.

Transmission And Loss Plans

36.0 This section provides information on trunk transmission and loss plans as they apply to the system. The VNLP (Via Net Loss Plan) and the FLP (Fixed Loss Plan) are described and trunk loss objectives and digital pad switching rules are given. All trunks on a system should be designed for either VNLP or FLP and must conform to certain trunk loss constraints.

The OMNI SI is a time-divided pulse-code-modulated system that can be operated as main, tandem, or satellite PABX within a TTTN (Tandem Tie Trunk Network), or as a main, satellite, or Class SS3 PABX in a SSN (Switched Services Network).

Switching PADs

36.1 To provide the correct transmission interface parameters, the ONMI SI PABXs are arranged to provide pad switching. Pad switching is digitally provided under software control. This allows a four-wire Tie trunk to be designated a non-tandem, tandem, or intertandem by changing the data base.

The digital pad values are 0, 2, 3, or 5 dB, depending on the application. The values of the pads that are digitally stored are subtracted from the signal in a pulse code modulated format to arrive at the padded value.

Via Net Loss Plan

36.2 The VNLP was developed in the 1950s to provide a method of controlling return loss by determining circuit loss based on the type of facility and circuit length. VNL (Via Net Loss) is more precisely expressed in its component parts: (VNL factor) x (facility length in miles) + 0.4 dB. The VNL factor is a constant associated with the type of facility assigned to the circuit. The 0.4 dB is a factor added to account for variation in circuit loss due to changes in temperature and other factors. Refer to Section 875-000-071 for further information on determining VNL for various circuits.

When a digital PABX is intended to replace an existing analog PABX, due to growth or feature enhancements, or a new PABX is being installed to operate under the VNLP, no special treatment of trunks or facilities is required, assuming the trunks or facilities are properly engineered for any other type of PABX. The VNLP should not be used if the system is installed with a direct T1-type interface or if a direct Tl-type interface will be used in the future.

Tandem and intertandem Tie trunks should be assigned to four-wire facilities from PABX to PABX. If a satellite Tie trunk is to be switched to a tandem or inter-tandem Tie trunk, the satellite Tie trunk should also use four-wire facilities. It is also recommended that class 4 WATS circuits use four-wire facilities if possible.

When a Tie trunk is switched to a CO trunk that has less than 2 dB of loss, the PABX can be programmed to switch in a 2 dB pad to ensure good balance. A pad should not be provided when the CO trunk ICL (Inserted Connection Loss) exceeds 2 dB.

Because pad control is part of the data base, the fact that the PABX/CO facility is short (less than 2 dB) should be noted when ordering a PABX.

The ICL objectives for the VNL plan are given in Table 36.1. These are established network objectives and are independent of the type of PABX or facilities used.

In PABX applications, regardless of whether the VNLP or FLP is implemented, a 5 dB digital pad is inserted on line-to-line calls. Figure 36.1 shows the connection. A typical line-to-CO trunk connection under the VNLP is shown in Figure 36.2. No digital pad is inserted and the connection loss is the facility loss plus 0.5 dB across the PABX, Figure 36.3 shows a line-to-intertandem Tie trunk using the VNLP and Figure 36.4 shows a line-to-nontandem Tie trunk connection. In Figures 36.1 through 36.4, the pads are labeled A and B to allow correlation between the figures and Table 36.2. Table 36.2 lists the pad switching rules for a PABX operating under the VNLP.

Fixed Loss Plan

36.3 When a direct Tl-type interface is used to connect circuits in a system, it is necessary to use the FLP to provide correct circuit interface levels. The FLP can also be used when all facilities are analog.

To properly integrate into the North American telephone network, a transmission plan such as the FLP must conform to the following constraints:

1. EML (Expected Measured Loss) and ICL must be symmetrical; that is, they must be equal regardless of transmission direction.

2. The 2-dB TLP (Test Level Point) at the outgoing side of analog toll switches and the 0-dB TLP at end offices must be retained.

3. The input and output TLPs of carrier facilities (-16 TLP and + 7 TLP) must be retained.

4. Existing lineup and test procedures for D-type channel banks must be retained.

5. Loss, noise, and echo grades of service should be brought as close to optimum as possible to maintain or improve transmission performance perceived by the network user.

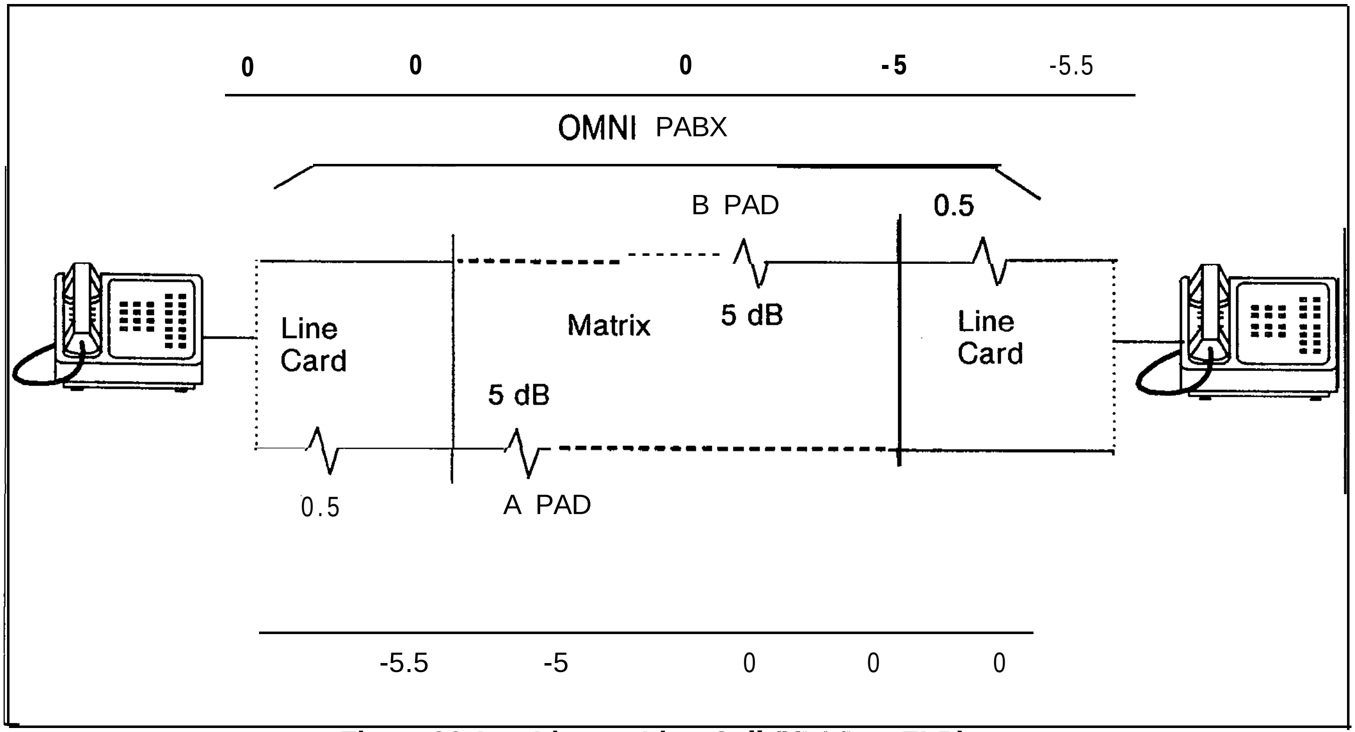


Figure 36.1 Line-to-Line Call (VNLP or FLP)

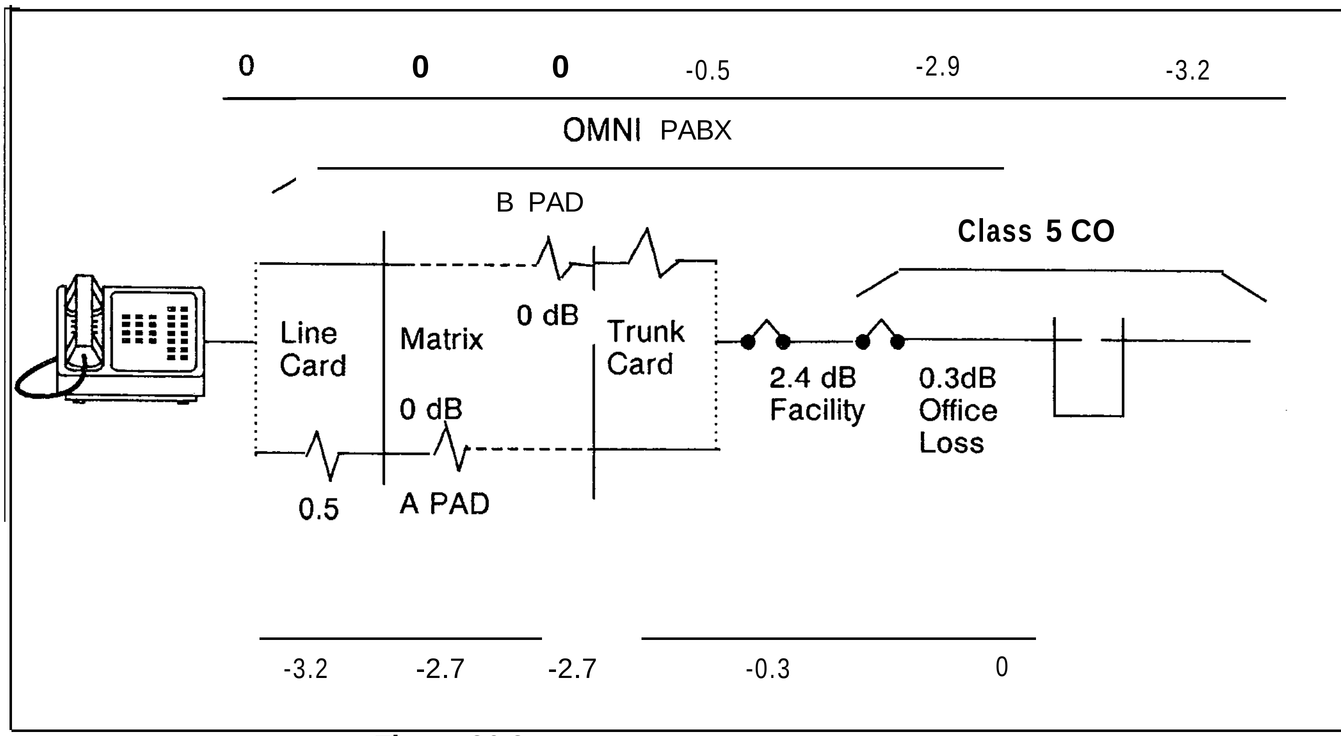


Figure 36.2 Line to CO Trunk Call (VNLP)

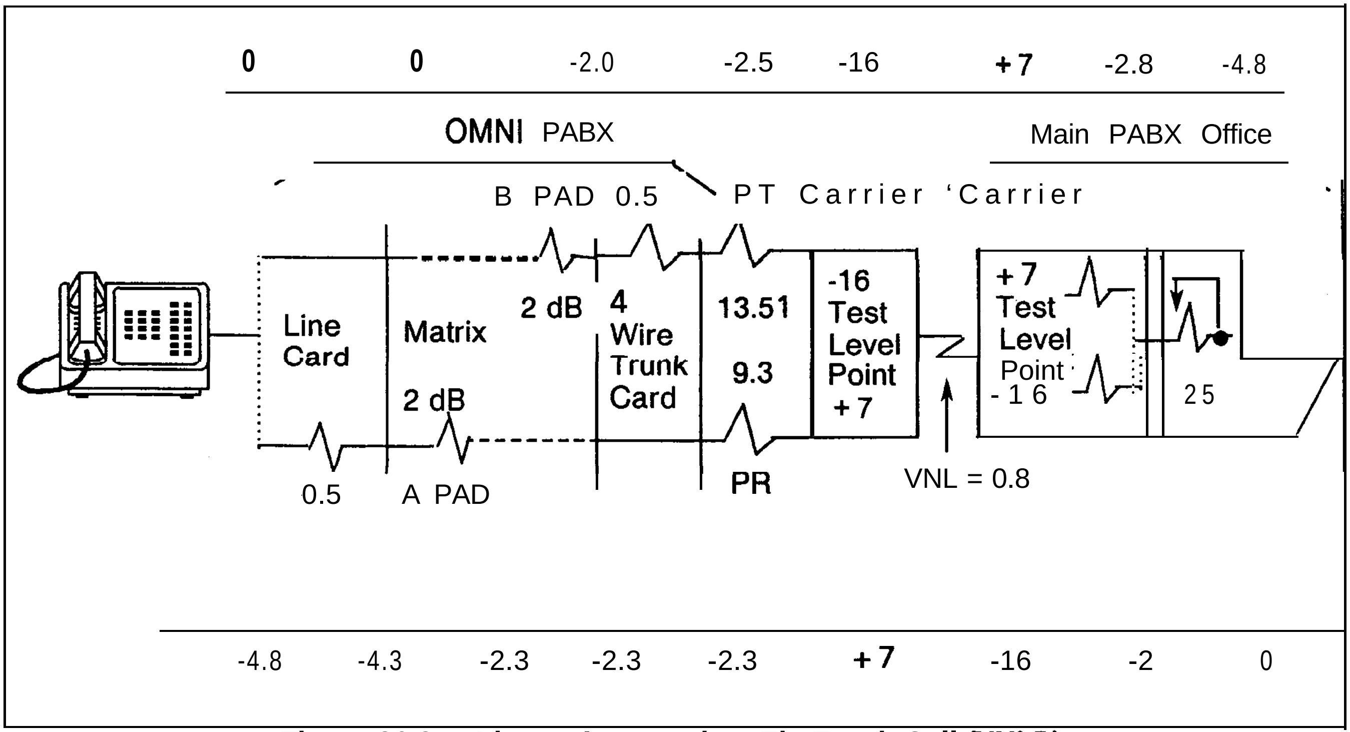


Figure 36.3 Line to Intertandem Tie Trunk Call (VNLP)

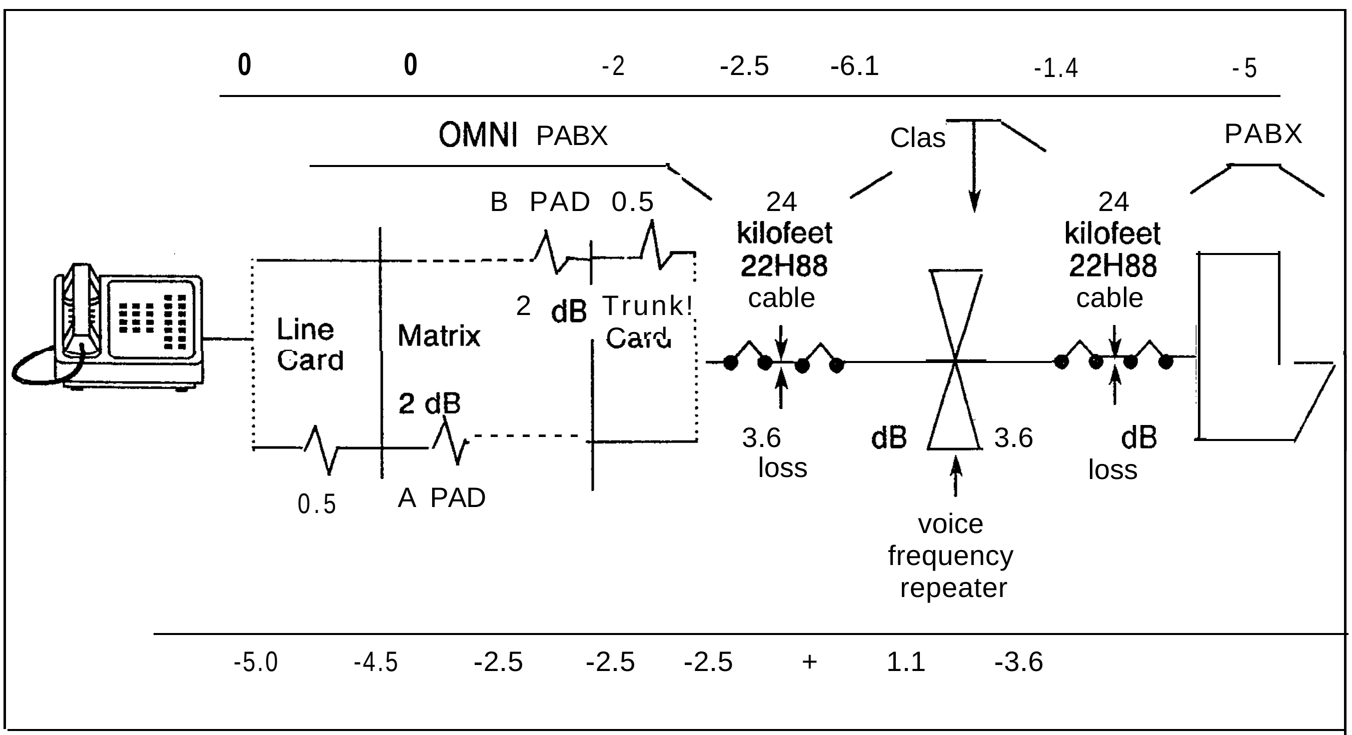


Figure 36.4 Line to Nontandem Tie Trunk Call (VNLP)

The FLP was developed specifically for systems where a direct digital interface to a digital trunk facility is provided. A direct digital trunk interface eliminates the toll terminating equipment (pads, four-wire terminating sets, and voice frequency repeaters) that is normally adjusted to provide the proper ICL on the receive transmission path of a PABX. A method of providing the required loss to the PABX was required and switchable pads provided the most reasonable solution. With different ranges of ICLs for the various trunk groups (Tie trunks and CO trunks) and the variation of VNL based on type of facility and circuit length, the number of pads required and the software control to insert the proper pad value became prohibitive. The FLP was chosen to eliminate the ICL variations caused by the differences in VNLs. The values chosen for CO and Tie trunks were a compromise that would not degrade typical connections beyond allowable limits nor require additional line treatment equipment, such as voice frequency repeaters on analog facilities that were acceptable for analog systems. It should be noted that a VNL trunk cannot be switched to an FLP trunk because improper levels and lack of symmetry would result.

Under the FLP, the center of the system is treated as a zero reference level point in both directions of transmission. This results in a symmetrical system with ideal characteristics for conference calls. Refer to Figures 36.5 through 36.11, which indicate the symmetry of levels at the mid-switch point.

It is virtually impossible to follow the VNLP when using digital or combination trunks. Trunk losses can be changed, but this usually results in violations of constraints listed in paragraph 36.3.

An FLP, by necessity, increases the loss of some trunks that would be assigned to low-loss facilities. An example would be a PABX located a few thousand feet from a CO that had a Tie trunk group assigned to a direct digital interface. The loss from the PABX to the CO would normally be approximately 0.5 dB. Under the FLP, it would be 3.5 dB. On the other hand, long trunks such as cross-country Tie trunks are limited to the fixed loss ICL, 5.5 dB, rather than the ICL determined by VNL + 2S + 2S (2S denotes a 2 dB switchable pad), which could be greater than 6 dB. Table 36.3 lists the fixed loss objectives and pad switching rules for a system.

Table 36.1 ICL Objectives for PABX Circuits (VNL Design)

CIRCUITS	COMMON LANGUAGE CODE	SHORT HUAL ICL (dB)	LONG HAUL ICL (GREATER THAN 6 MS ROUND TRIP DELAY) (dB)
Line on premises	PX		
Line off premises	OS	Up to 4.0	VNL + 4.0
PABX-CO trunk	TK	up to 3.5	
Foreign exchange trunk	FT	3.5	VNL + 4.0
Satellite Tie trunk	SA	VNL + 2S + 2S	VNL + 2S + 2S
Nontandem PABX Tie trunk	TL	VNL + 2S + 2S	VNL + 2S + 2S
Tandem PABX Tie trunk	TA	VNL + 2S + 2S	VNL + 2S + 2S
inter-tandem PABX Tie trunk	IT	VNL + 2S + 2S	VNL + 2S + 2S
Incoming class 5 WATS trunk	WI(5)	up to 3.5	VNL + 4.0
Outgoing class 5 WATS trunk	WS(5)	up to 3.5	VNL + 4.0
Outgoing class 4 WATS trunk	WS(4)	4.5	VNL + 4.0

NOTE: 2S denotes a 2-dB switchable pad.

The FLP trunk cards provide for adjustments to be made on any remaining analog trunks terminating on the system. These adjustments, essentially similar to voice frequency attenuation selection in channel units, compensate for the analog facility loss.

On two-wire trunks, the level change is made by changing strapping or strapping resistors in the trunk units. This change is indicated by the variable attenuators and amplifiers in Figures 36.5, 36.6, and 36.7. The FLP two-wire trunk cards may be used for both the VNLP and the FLP. Certain trunk cards are designed for the VNLP only and do not provide the adjustments required for the FLP. If there is a possibility that the system will change in the future from the VNLP to the FLP because a direct Tl-type interface will be used, the FLP trunk cards (applicable to both plans) should be installed initially.

On four-wire trunks, the level change is performed by using standard-level coordination pads. The values of the transmit and receive analog pads (PT and PR, respectively) required to adjust levels differ, depending on the loss plan implemented. Also, under the VNLP, PR varies with the VNL of the trunk. Under the FLP, PT is always 15.5 dB and PR is 7.0 dB (Figure 36.9). Under the VNLP, PT is always 13.5 dB and PR is (VNL + 6.5) dB (Figure 36.3). In those cases where carrier terminals are not collocated with the switching system, the pad values are reduced by the four-wire facility loss and the loss of any impedance matching equipment required.

Figures 36.5 through 36.11 provide typical circuit layouts and losses for systems operated under the FLP.

Table 36.2 FCL and VNLP Digital Pad Switching Rules

CONNECTION	END TO END ICL (dB)	DIGITAL PAD (A) RECEIVE (dB)	DIGITAL PAD (B) RECEIVE (dB)	REMARKS
Line to line	5.5	5	5	
Line to trunk group 1	Less than 3.6	0	0	See Note 1
Line to trunk group 2	VNL + 2S + 2S	2	2	See Note 2
Trunk group 1 to trunk group 1	Less than 7.1	0	0	
Trunk group 2 to trunk group 2	VNL + VNL + 4	0	0	
Trunk group 2 to trunk group 1	VNL + 6.6	0 or 2	0 or 2	See Note 3

NOTES:

1. Trunk group 1 consists of PABX-CO, foreign exchange, and class 5 WATS trunks.
2. Trunk group 2 consists of Tie and class 4 WATS trunk.
3. Under the VNLP, when the PABX-CO trunk has an ICL of less than 2 dB, a 2 dB pad is inserted when the PABX-CO trunk is switched to a Tie trunk. If the PABX-CO trunk ICL exceeds 2 dB, no pad is inserted.
4. 2S denotes a 2-dB switched pad.

Table 36.3 ICL and FLP Digital Pad Switching Rules

CONNECTION	END TO END ICL (dB)	DIGITAL PAD (A) RECEIVE (dB)	DIGITAL PAD (B) RECEIVE (dB)	REMARKS
Line to line	5.5	5	5	
Line to trunk group 1	3.5	3	0	See Note 1
Line to trunk group 2	5.5	5	0	See Note 2
Trunk group 1 to trunk group 1	3.5	0	0	
Trunk group 2 to trunk group 2	5.5	0	0	
Trunk group 2 to trunk group 1	5.5	0	2	

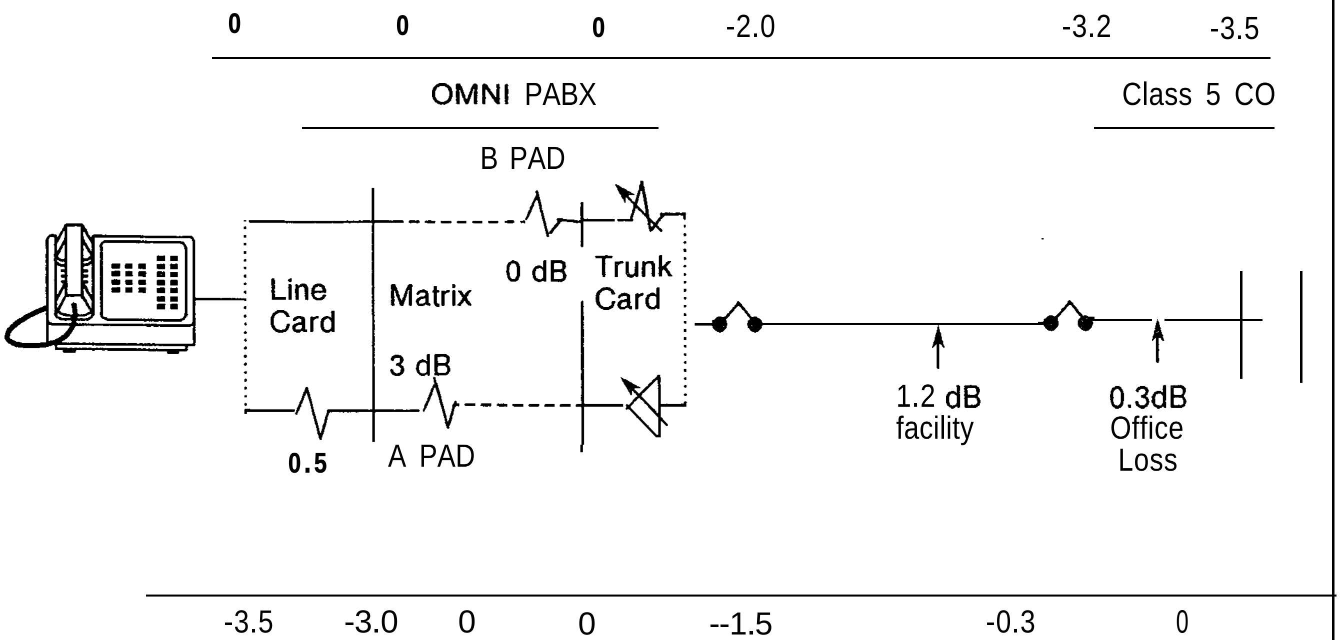
NOTES:

1. Trunk group 1 consists of PABX-CO, foreign exchange, and class 5 WATS trunks.
2. Trunk group 2 consists of Tie and class 4 WATS trunk.
3. Gain and loss for two-wire trunks must be done in trunks by changing strapping resistors.

Transmission Performance and Testing Procedures

36.4 This paragraph provides information on transmission performance and testing procedures as they apply to the systems.

The systems provide a means of switching two-wire and four-wire voice frequency facilities via a four-wire time-divided pulse-code-modulated transmission medium. The systems are designed to switch line circuits that have a nominal characteristic impedance of 600 ohms, and trunk circuits that have either a 600- or 900- ohm characteristic impedance. All systems are capable of direct digital interface to a DS-1 level signal (D2 or D3 format) from a Tl-type repeater line.

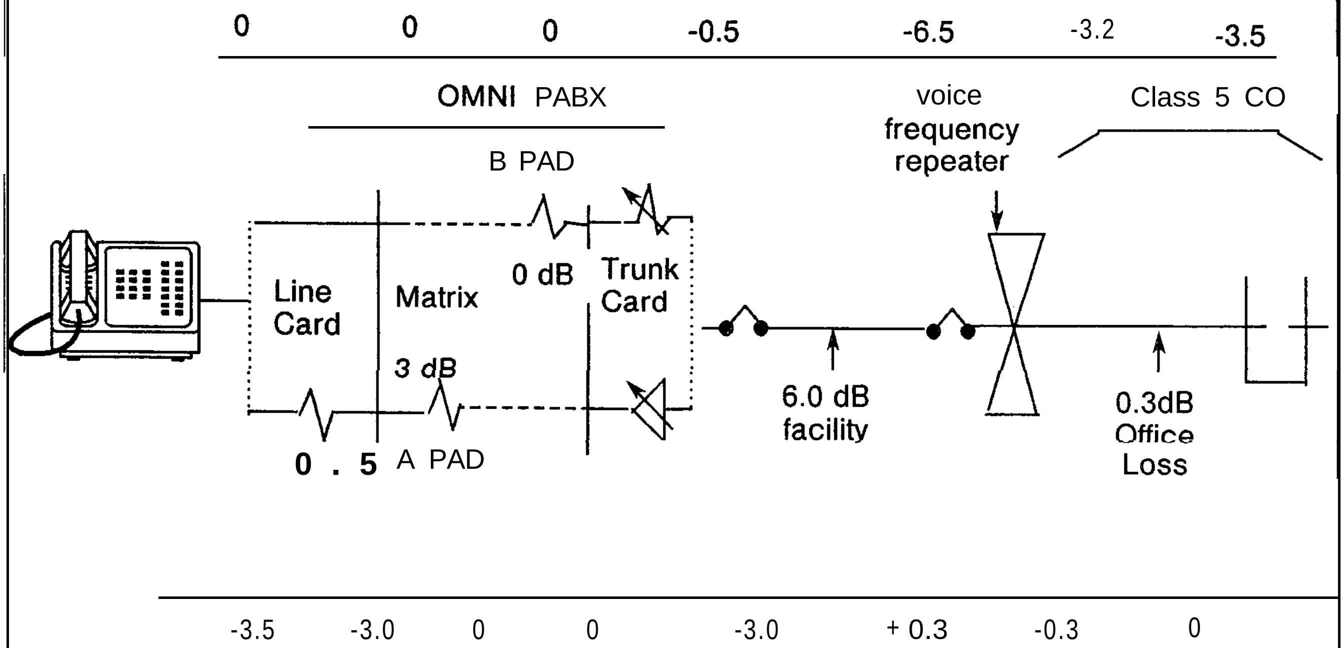


Notes:

1. Strapping resistors are changed to compensate for the facility to obtain a 3.5 dB circuit.
2. Digital pads are labeled A and B to Table 36.3

Figure **36.5** **Line to CO Trunk Call (FLP)**

The systems are designed to be installed as new PABXs or to replace existing installations. They may operate as main, tandem, or satellite PABXs within a TTTN (Tandem Tie Trunk Network), or as a main, satellite, or class SS3 PABX in a SSN (Switched Services Network). Pad switching is digitally provided under software control, enabling a four-wire Tie trunk to be designated a nontandem, tandem, or intertandem trunk by changing the data base. No hardware change is required to upgrade or downgrade a four-wire Tie trunk. All tandem and intertandem Tie trunks should be assigned to four-wire facilities. Non-tandem and satellite Tie trunks are strongly recommended for four-wire design because transmission on universal service connections is improved and because Tie trunks are often upgraded as customer needs expand and the tandem Tie trunk network grows.

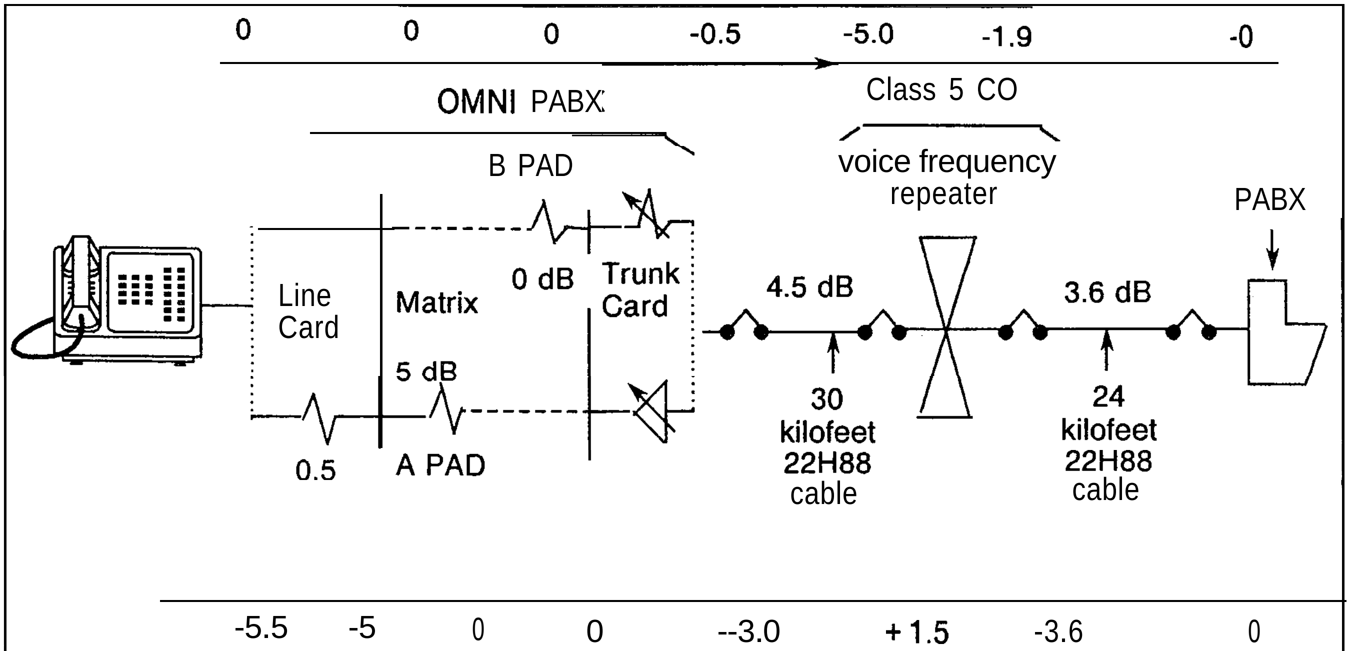


Note:
Strapping resistors are changed to compensate for the facility to obtain a 3.5 dB circuit.

Figure 36.6 Line to CO Trunk Call (FLP) with Voice Frequency Repeater

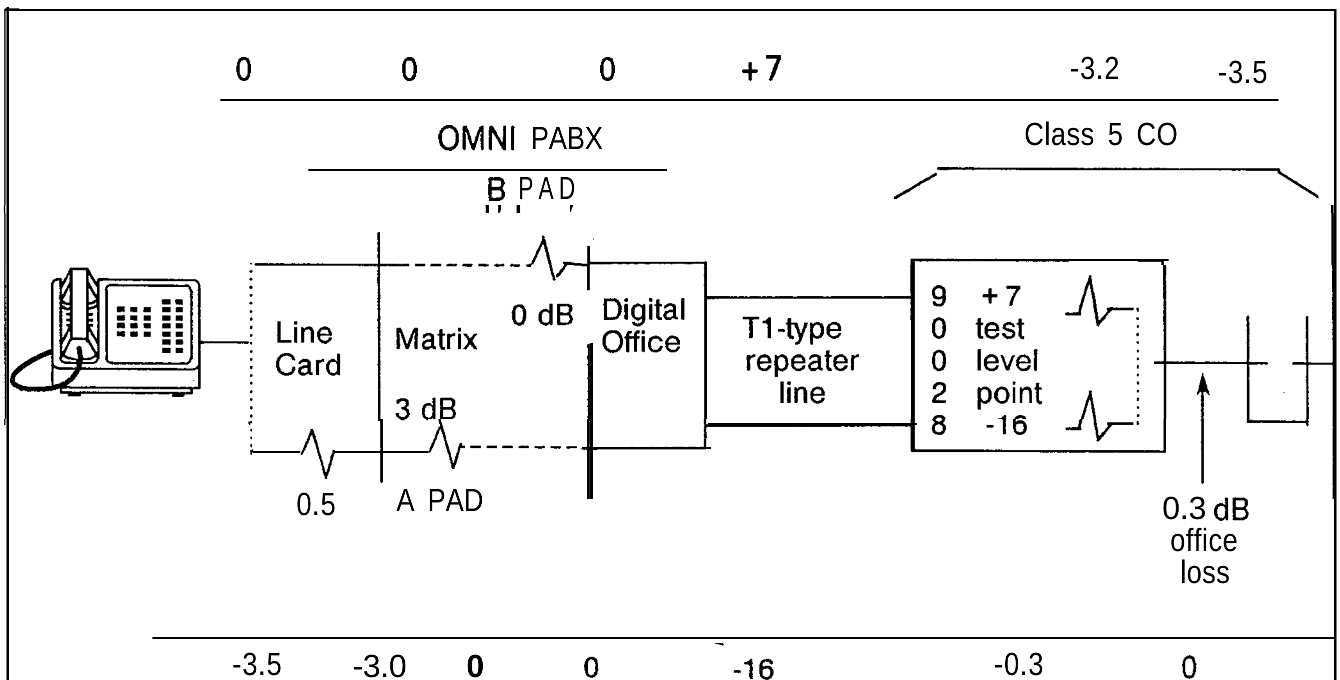
If a system is intended to replace an existing analog PABX, due to growth or feature enhancements, no special treatment of trunks or facilities is required, assuming the trunks and facilities were properly engineered for the analog PABX. The PABXs provide pad switching, a station-to-station insertion loss of 5.5 dB, and 400-ohm to 400-ohm battery feed devices for lines in compliance with latest industry recommendations. Extended long loop line cards that have 900-ohm line impedance and 200-ohm to 200-ohm battery feed devices are available. The PABXs have been designed to give a high grade of service with excellent return loss and singing stability.

While the PABXs can operate under either the VNLP (Via Net Loss Plan) or the FLP (Fixed Loss Plan) with no direct digital interface, the FLP is required when a direct digital interface is used. If a system is anticipated to grow and require direct digital interface in the future, installation under the FLP is recommended to avoid later adjustments and rearrangement. It should also be noted that not all trunks are applicable to both the VNLP and FLP. If there is a possibility that the system will change in the future from the VNLP to the FLP, the trunks cards applicable to both plans should be installed initially.



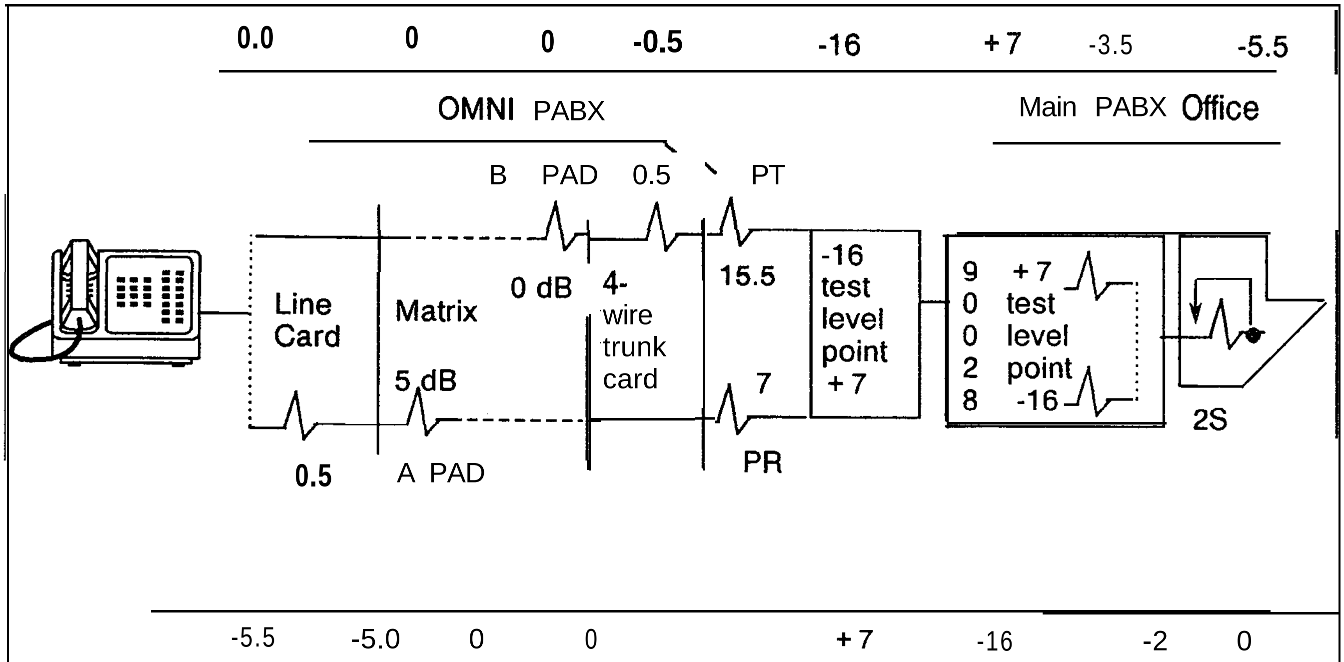
Notes:
1. Strapping resistors are changed to compensate for the facility to obtain a 5.5 dB circuit.
2. Digital pads are labeled A and B to allow correlation to Table 36.3.

Figure 36.7 Line to Nontandem Tie Trunk Call (FLP)



Note: Digital pads are labeled A and B to to allow correlation to Table 36.3.

Figure 36.8 Line to CO Trunk Call (FLP) with T1-Type Repeater Line



Notes:
1. Pads PT and PR are selected to obtain an FLP loss of 5.6 dB.
2. Digital pads are labeled A and B to allow correlation to Table 36.3

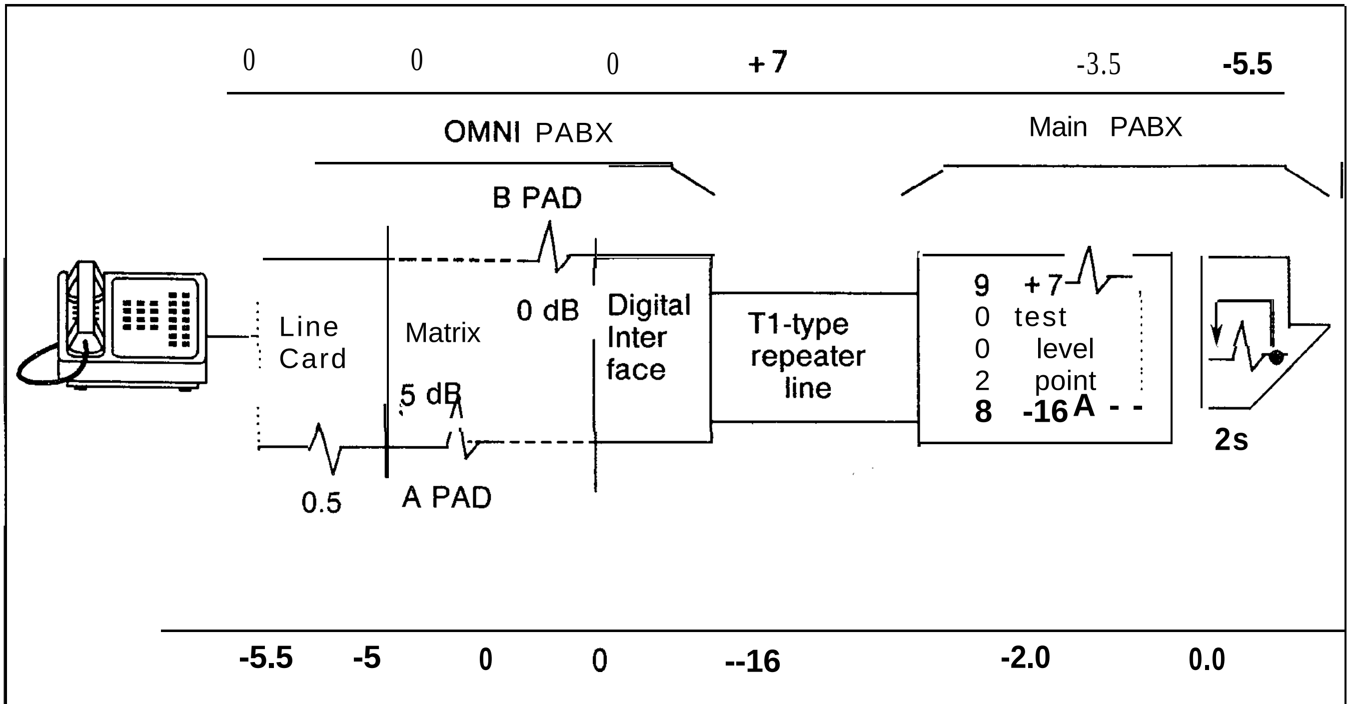
Figure 36.9 Line to Intertandem Tie Trunk Call (FLP)

When a trunk group is provided on different facilities (i.e., direct digital interface and physical facilities), the physical facilities should be reviewed to ensure that proper transmission levels will be achieved. Loading and/or line treatment should be provided as required to avoid contrast between trunks and subsequent customer complaints.

Test Equipment

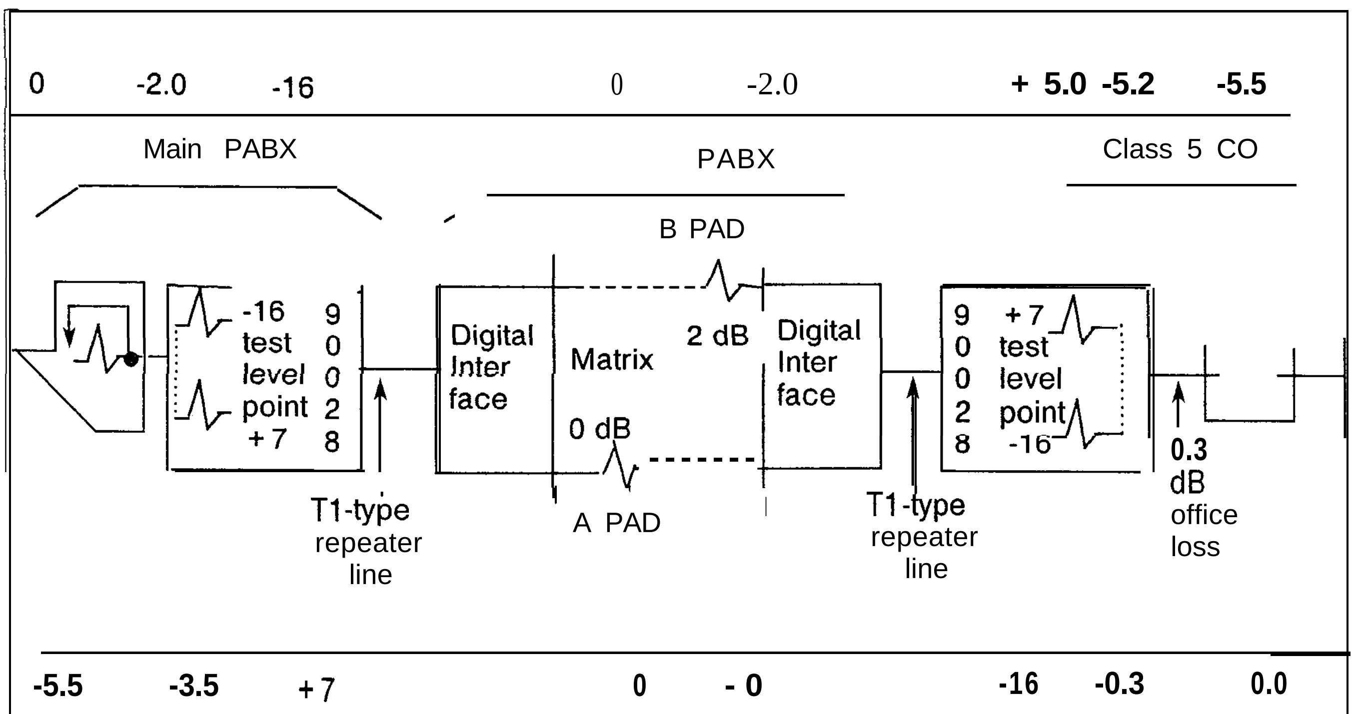
36.5 No attempt has been made to specify a particular manufacturer's equipment type. However, this paragraph does provide a list of recommended test equipment. Certain tests require very precise measurements and the use of test procedures or test equipment not equivalent to that specified (wrong impedances, improper test method, etc.) will result in errors. The following are examples of test equipment necessary to perform the procedures given in this section:

- Models 4937A and 4938A transmission measuring set (Hewlett Packard)
- Transmission measuring set (Hekimian Laboratories)
- Model VH795G 600 ohm attenuator (Daven & Company)
- Model 3581C frequency selective meter (Hewlett Packard)
- Model 4945A transmission impairment set (Hewlett Packard)
- Model 9031 return loss measuring set (Wiltron)
- Model TTS-58A impulse noise test set (N. E. Electronics Corp.)
- Model T-207 longitudina! balance test set (Wilcom Products. Inc.)
- Model 3779D Primary Multiplex Analizer (Hewlett Packard)



Note: Digital pads are labeled A and B to Table 36.3

Figure 36.10 Line to Intertandem Tie Trunk Call (FLP) with Tl-Type Repeater Line



Note: Digital pads are labeled A and B to Table 36.3

Figure 36.11 Intertandem Tie Trunk to CO Trunk Call (FLP)

Transmission Performance Specifications

36.6 This paragraph lists the transmission specifications for the OMNI SI PABXs. The specifications are for the PABX alone and do not include any outside plant effects. Short lengths of cabling (50 feet or less) from the PABX to an MDF (Main Distributing Frame), or equivalent, are accounted for.

Impedance and Resistance

36.6.1 Impedance and resistance are defined as follows:

- Line circuit impedance: 600 ohms
- Line circuit battery feed resistance: 800 ohms
- Long loop line circuit impedance: 900 ohms
- Long loop circuit battery feed resistance: 400 ohms
- Two-wire trunk circuit impedance: 900 ohms
- Four-wire trunk circuit impedance: 600 ohms
- Four-wire trunk idle circuit termination: 600 ohms

Insertion Loss

36.6.2 The nominal insertion loss on any transmission path through the OMNI SI PABX shall be 0.5 dB, referenced to 0 dBm at 1 KHz. This loss does not include the loss of digital pads that are included on various path configurations. On station-to-station calls, an additional 5-dB loss will be switched in, giving a loss of .5 dB + - 0.5 dB. On calls involving trunks, digital pads values of 0 dB, 2 dB, or 3 dB may or may not be switched in as the site engineering parameters dictate.

Frequency Response

36.6.3 Frequency response limits of any voice frequency path (station-to-station, trunk-to-trunk, or trunk-to-station) relative to the 1 KHz insertion loss are as follows:

Frequency	Response
204 Hz	0 to + 5.0 dB
304 to 3,004 Hz	-0.8 to + 1 .O dB
3,204 Hz	+ 1.5 dB maximum
3,404 Hz	0 to + 3.5 dB

NOTE: + dB means more loss and - dB means less loss.

Level Tracking Error

36.6.4 The maximum level error caused by the system from a + 3 to -50 dBm amplitude (1,004 Hz) input range is as follows:

Input Level	Required Relative to I-KHz Insertion Loss
+ 3 dBm to -37 dBm	+ -0.5 dB
-37 dBm to -50 dBm	+-1.0 dB

Signal to Distortion (C-Notch)

36.6.5 The C-notch signal to distortion ratio is as follows:

Input Level (100 Hz)	Signal to Distortion Ratio
0 dBm to -30 dBm	33 dB or greater
-40 dBm	27 dB or greater
-50 dBm	22 dB or greater

Intermodulation Distortion (Four-Tone Method)

36.6.6 This test method uses two sets of tones (one set at 856 and 863 Hz with a 7-Hz spacing, and the second set at 1,374 and 1,385 Hz with an 11 -Hz spacing) that are applied simultaneously to the path at a combined level of -10 dBm with the second and third order distortion products being measured. The requirements are as follows:

- The second order distortion shall measure less than -45 dB.
- The third order distortion shall measure less than -45 dB.

Single Frequency Distortion

36.6.7 Single frequency distortion is as follows:

Frequency of 0 dBm0 Input Signal	Maximum Distortion Power
0 to 12 kHz	-28 dBm0 (Note 1)
1.020 kHz	-40 dBm0 (Note 2)

Notes:

- For the given range of input frequencies, the output power in any other single frequency shall not exceed the given power.

- The output at any other frequency from 0 to 4 KHz shall be less than this power.

Envelope Delay Distortion

36.6.8 Envelope delay distortion measured on any path through the system when referenced to 1,804 Hz is as follows:

- At 604 to 3,004 Hz, less than 300 µs.
- At 404 to 3,204 Hz, less than 550 µs.

The PAR (Peak-to-Average Ratio) reading will be 94. These values are the corrected readings that have test holding and battery feed configuration delays subtracted.

It must be kept in mind that the digital matrix has absolute delay but does not have envelope delay distortion. The values listed above indicate the envelope delay distortion of the line card, the battery feed circuitry, the two-wire to four-wire conversion, and the filtering applied prior to analog-to-digital PCM (Pulse Code Modulation) conversion. In those cases where a direct digital interface is used for trunking, the benefits of direct digital interfaces become apparent in that the specifications listed above apply from the station to the far end of the directly interconnected PCM system. The far-end system must, however, be compatible and capable of meeting published D2/D3 delay specifications.

Path Message Noise

36.6.9 Path message noise on any path through the system should not exceed the following values:

- Message noise maximum: 23 dBrnC, 35 dBrn 3KHz.
- Average: 18 dBrnC.

The typical noise on any station-to-station path is in the 15 to 18 dBrnC range.

Path Impulse Noise

36.6.10 Impulse noise on any path through the system should be less than or equal to 5 counts in 5 minutes at the 52-dBrnC level.

Crosstalk Attenuation

36.6.11 The worst case crosstalk attenuation measured between any two paths should be 70 dB at 1,004 Hz relative to the sending path level. Typical paths will exceed 75 dB.

Longitudinal Balance

36.6.12 Longitudinal balance attenuation of any line circuit or trunk circuit when measured with the IEEE method, single port, at Z_M equal to 750 ohms, should be greater than or equal to the following values:

- At 204 Hz, 55 dB
- At 1,004 Hz, 60 dB
- At 3,004 Hz, 60 dB

Return Loss

36.6.13 The ERL (Echo Return Loss) and SRL (Singing Return Loss) performance of stations and trunks is as follows:

- Station-to-station: 26 dB ERL minimum and 18 dB SRL minimum.

- Station-to-trunk: 26 dB ERL minimum and 18 dB SRL minimum.

- Trunk-to-trunk: 27 dB ERL minimum and 18 dB SRL minimum.

In the idle condition, all stations and trunks (except ground start/ loop CO trunk circuits) will exhibit an SRL of better than 12 dB.

Lonaitudinal Susceptibility

36.6.14 With a 60-Hz longitudinal input current of up to 10 mA or 50 VAC, the metallic noise measured on the T&R should be as follows:

- 24 dBrnC or less for stations.
- 18 dBrnC or less for trunks.

It may be difficult to achieve 10 ma of longitudinal current when measuring trunk circuits that have a high longitudinal impedance. These circuits should be tested with a longitudinal test voltage from tip to ground or ring to ground of up to 50 VAC.

NOTE: Exercise caution when extending four-wire trunks over metallic facilities that may convert longitudinal voltages to metallic voltage prior to reaching the PABX.

36.6.15 The systems are capable of supplying digitally generated tones (the tone levels indicated refer to levels of tones at their point of departure from the PABX towards a station), which are used in normal operation and with various features. See Table 4.2 for a description of these tones.

36.6.16 The minimum facility leakage tolerance for trunks is 20 kilohms. On station lines, the minimum facility leakage tolerance is dependent upon the ringer equivalency total connected to the line in question as follows:

- For one equivalent ringer, 15 kilohms
- For two equivalent ringers, 17.5 kilohms
- For three equivalent ringers, 20 kilohms
- For four equivalent ringers, 30 kilohms
- For five equivalent ringers, 50 kilohms

36.6.17 To compensate for cable distances between the OTS (Office Terminating Shelf) and the systems, optional pads are available as listed in Table 36.4.

36.7 The system matrix is a zero TLP (Test Level Point) and the digital milliwatt described in the D2- and D3-type compatibility specification is used to align the systems in the factory. A similar digital milliwatt is provided in the GTD-4600 system and is an option in the OMNI SI. When the digital milliwatt is dial accessed, the pad switching rules for a line are implemented. The systems were designed to be compatible with D2- and D3-type carrier systems and their alignment procedures.

The system must operate as the master unit when a D2- or D3-type channel bank is associated with it via a direct digital interface. The channel bank is aligned on a looped basis as it would be if the distant terminal was another channel bank. The + 7 and -16 TLPs are maintained, and drop levels per the VNLP or FLP (as applicable) are set by adjusting the pads associated with the four-wire terminating set. Once the channel bank is adjusted properly, levels from the system's digital tone source or from an oscillator connected to an analog port on the system can be verified.

As noted in paragraph 36.6.1, the line impedance of the system is 600 ohms. Two-wire trunks are 900 ohms and four-wire trunks are 600 ohms. When making terminating measurements on the system (MDF to MDF), hold coils and possibly battery feed devices are required. If the test equipment used does not provide these functions, a hold coil can be built locally. Figure 36.12 shows one such device that can provide holding and battery feed functions with minimum transmission impairment.

Table 36.4 Optional Pads

OPTION	CABLE	FEET
A	Type 4162	0 to 45
	GTS-8510 (GT-16)	0 to 50
B	Type 4162	46 to 320
	GTS-8510 (GT-16)	51 to 350
C	Type 4162	321 to 550
	GTS-8510 (GT-16)	351 to 650

Transmission Test Procedures

36.8 All systems are factory tested for transmission performance. This paragraph provides a description of the test procedures that are to be used if there is a need to make transmission tests on the systems.

The path to be tested may be set up by using the available test panels or by dialing up the connection in a normal fashion. Refer to the appropriate section of FGBS Technical Practices for system and test panel operation.

Battery Feed and Hold Coil Arrangements

36.8.1 Depending on the test equipment used, a hold coil or battery feed coil may be necessary to maintain the connection across the system. Other test equipment may require DC blocking and a large capacitor will be required. The hold coil and battery feed arrangements shown in Figure 36.12 are recommended. For precise measurements, the loss, distortion, etc. of these devices should be taken into account.

Insertion Loss Aad Frequency Response

36.8.2 To measure insertion loss and frequency response, proceed as follows (test objectives are listed in paragraphs 36.6.2 and 36.6.3).

1. Establish a station-to-station, station-to-trunk or trunk-to-trunk path through the system. (Provide DC holding for stations or incoming loop trunk circuits and provide a battery feed arrangement for CO-type loop trunk circuits as shown in Figure 36.12.)

2. Connect the transmission measuring set on the established path as shown in Figure 36.13, and measure the end-to-end loss in the frequency band from 204 to 3,404 Hz with the oscillator set to a sending level of 0 dBm.

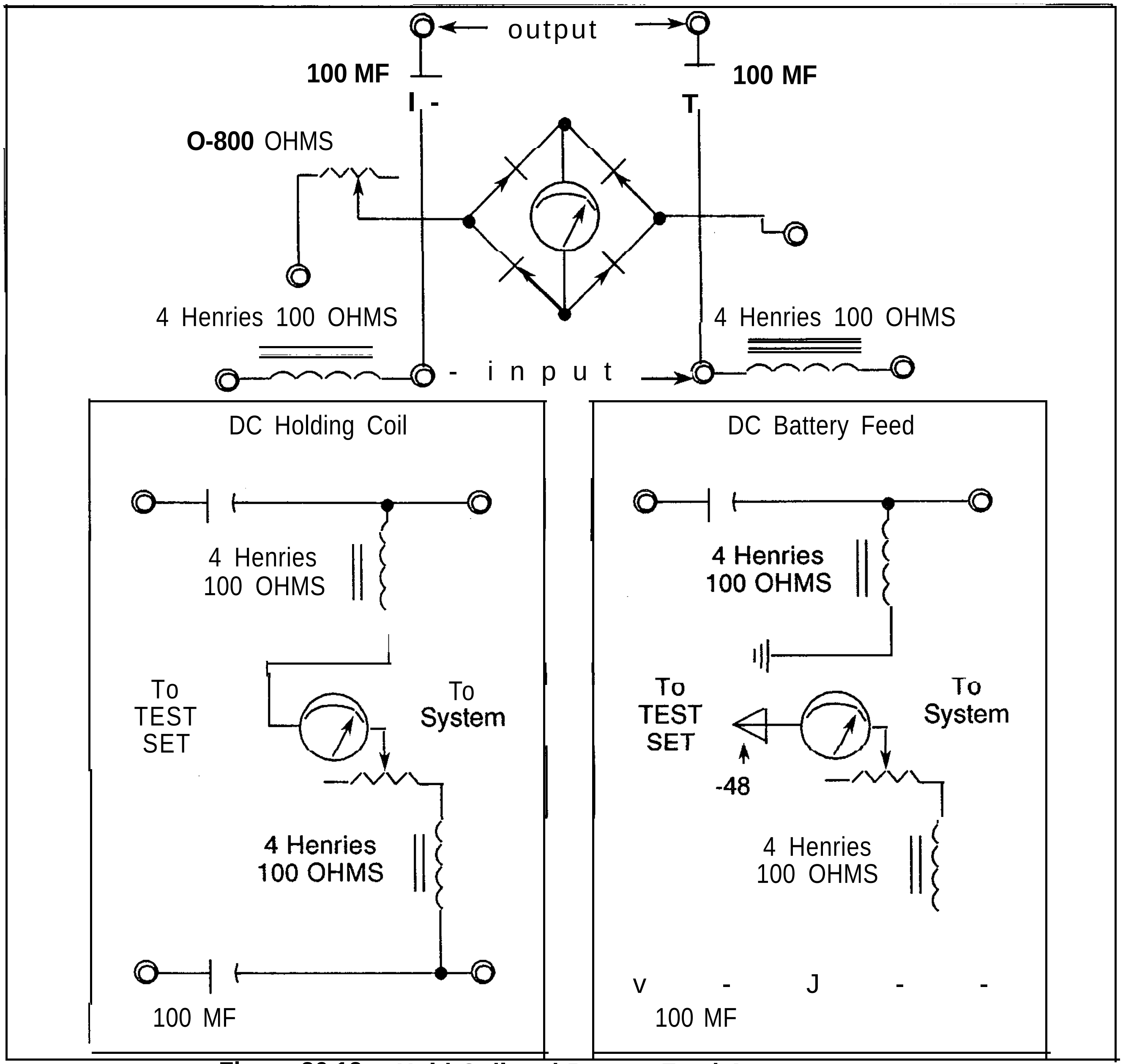


Figure 36.12 Hold Coil and Battery Feed Arrangements

Level Tracking Error and C-Notch Distortion

36.8.3 To measure the level tracking error, proceed as follows (test objectives are listed in paragraph 36.6.4).

1. Establish a station-to-station, station-to-trunk, or trunk-to-trunk path through the system. (Provide dc holding for stations or incoming loop trunk circuits and provide a battery feed arrangement for CO-type loop trunk circuits as shown in Figure 36.12.)

2. Connect both the 600-ohm oscillator and frequency selective meter through a 0- through 100-dB, 600-ohm decade attenuator as shown in Figure 36.13.

3. Adjust the oscillator for 1.004 KHz at 0 dBm into 600 ohms and the send attenuator for 0-dB loss.

4. At the terminating end, adjust the frequency selective meter for 1.004-kHz and -50 dBm input sensitivity and the receive attenuator for 50-dB loss.

5. Vary the level into the PABX from 0 to -50 dBm by increasing the send attenuator loss, while at the same time decreasing the receive attenuator loss by equal steps.

NOTE: Two attenuators are needed to provide accurate level adjustments in 1-dB steps and to use the same meter scale for all measurements to obtain an accuracy of + - 0.1 dB.

To measure the C-notch distortion, proceed as follows (test objectives are listed in paragraph 36.65).

1. Substitute a noise measuring set equipped with a C-notch filter for the receive attenuator and frequency selective meter (Figure 36.15).

2. Adjust the oscillator for the C-notch frequency at levels from 0 to -50 dBm.

3. Measure and record the C-notch distortion. The signal-to-distortion ratio is the difference (in dB) of the measurement without the C-notch filter and the measurement with the C-notch filter for each input level selected.

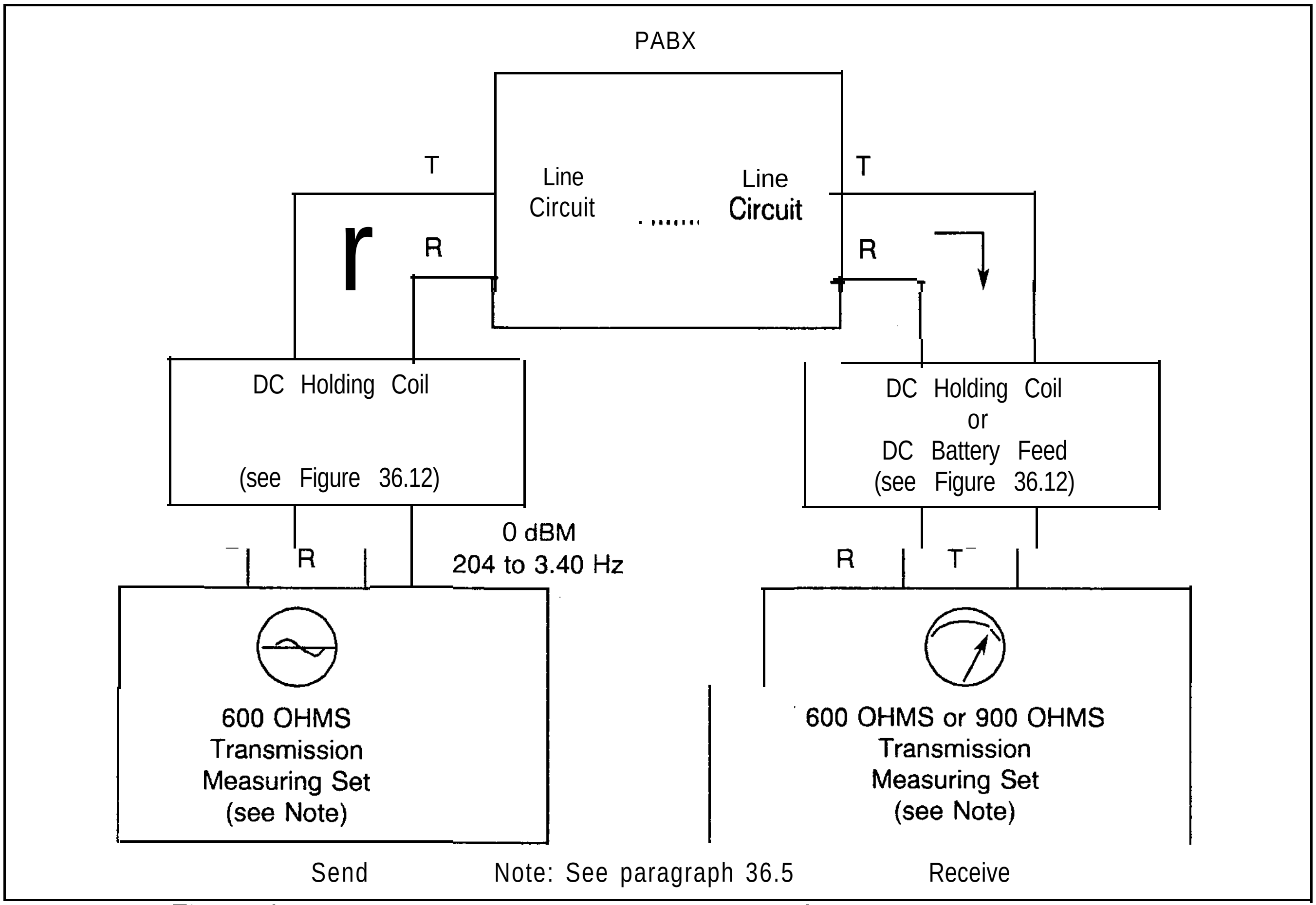


Figure 36.13 Insertion LOSS and Frequency Response Measurement

Intermodulation Distortion, Envelope Delay Distortion, and Noise

36.8.4 To measure intermodulation distortion, proceed as follows (test objectives are listed in paragraph 36.6.6).

1. Establish a station-to-station, station-to-trunk, or trunk-to-trunk path through the system. (Provide DC holding for stations or incoming loop trunk circuits and provide a battery feed arrangement for CO-type loop trunk circuits as shown in Figure 36.12).

2. Connect the transmission measuring set on the established path as shown in Figure 36.18.

3. Using the intermodulation distortion test unit, adjust the send level for -10 dBm and measure the second and third order distortion products.

To measure the envelope delay distortion, proceed as follows (test objectives are listed in paragraph 36.6.8).

1. Use the same test configuration established in paragraph 37.4.3, 1 and 2 along with an envelope delay test unit, which is set at a modulation frequency of 83-1/3 Hz.

2. Adjust the send level for -10 dBm. Adjust the test unit for 1,840 Hz equal to 0µs.

3. Vary frequency and record delta from 1840 Hz reading.

To measure path message and impulse noise, proceed as follows (test objectives are listed in paragraphs 36.6.9 and 36.6.10).

1. Use the same test configuration established above along with a noise measuring test set.

2. Set the sending portion of the transmission measuring set to quiet termination.

3. Measure the C-message, 3-KHz flat, and impulse noise.

Crosstalk Attenuation

36.8.5 To measure the crosstalk attenuation, proceed as follows (test objectives are listed in paragraph 36.6.11).

1. Establish two paths through the system and provide the proper holding or battery feed arrangement shown in Figure 36.42.

2. On one path, connect a 900-ohm/600-ohm, 0-dBm oscillator at one end and terminate the far end in the proper AC termination impedance as shown in Figure 36.15.

3. On the second path, connect a frequency selective meter at one end of the path and terminate the far end in the proper AC termination impedance shown in Figure 36.15.

4. Adjust the oscillator for 1,004 Hz at 0 dBm and the frequency selective meter for a frequency of 1004 Hz.

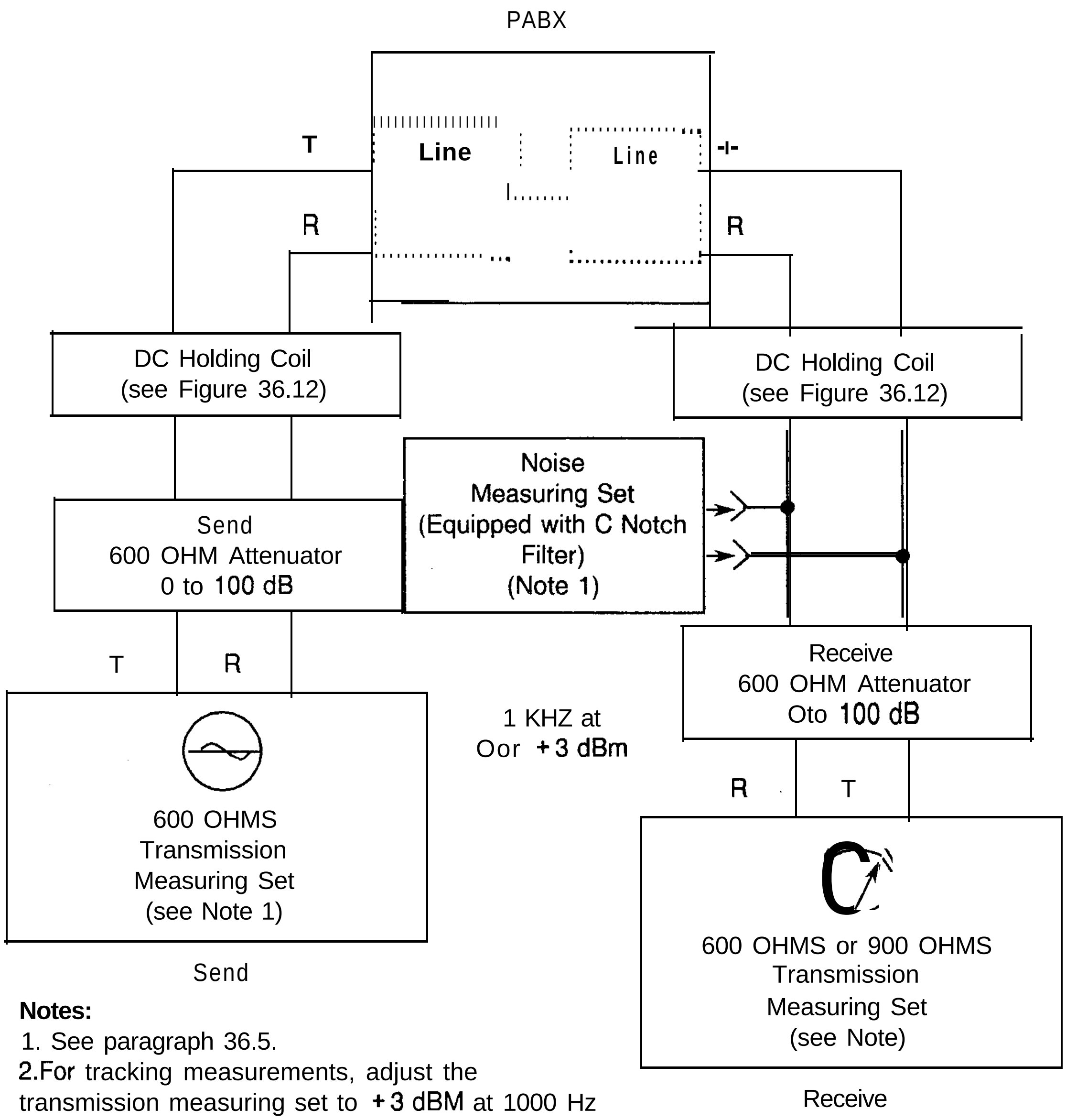
5. Measure the crosstalk attenuation.

Longitudinal Balance Attenuation

36.8.6 To measure longitudinal balance attenuation, proceed as follows (test objectives are listed in paragraph 36.6.3).

1. Establish a path to the system's quiet code termination or to a terminated line or trunk as shown in Figure 36.16. The required battery feed arrangement is applied only to trunks.

2. Using the longitudinal balance test set, measure the longitudinal balance at 204, 1004, and 3004 Hz.



Notes:
1. See paragraph 36.5.
2. For tracking measurements, adjust the transmission measuring set to +3 dBM at 1000 Hz and adjust the send and receive attenuator so that the combined loss equals -53 dB (E.G., send = 0 receive = -53;
send = -10, receive = -43; send = -25,
receive = -25; ect.)
3. For C-notch measurements, remove the receiver attenuator and frequency selective meter and terminate with the noise measuring set equipped with a C-notch filter

Figure 36.14 Level Tracking and C-Notch Distortion Measurement

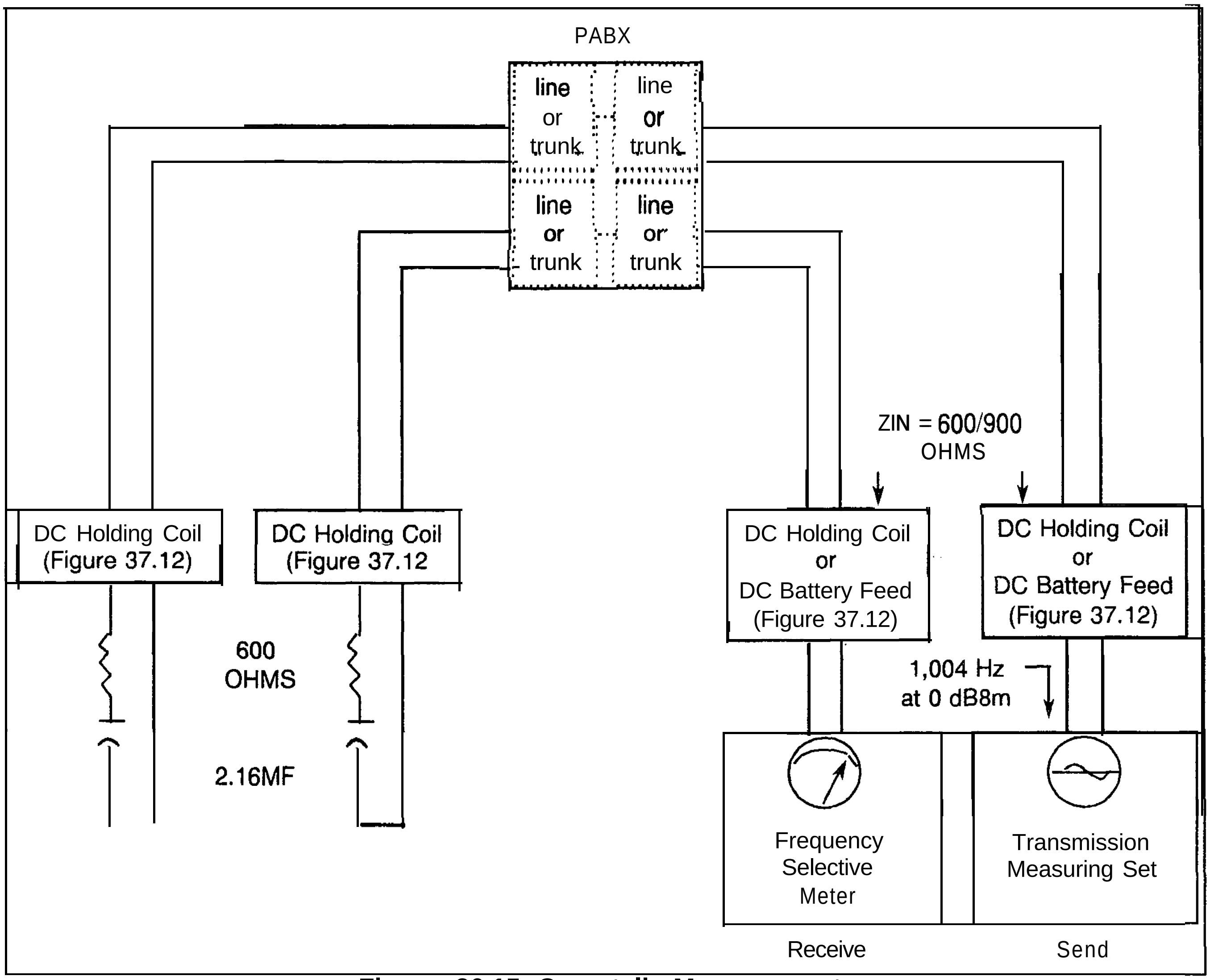


Figure 36.15 Crosstalk Measurement

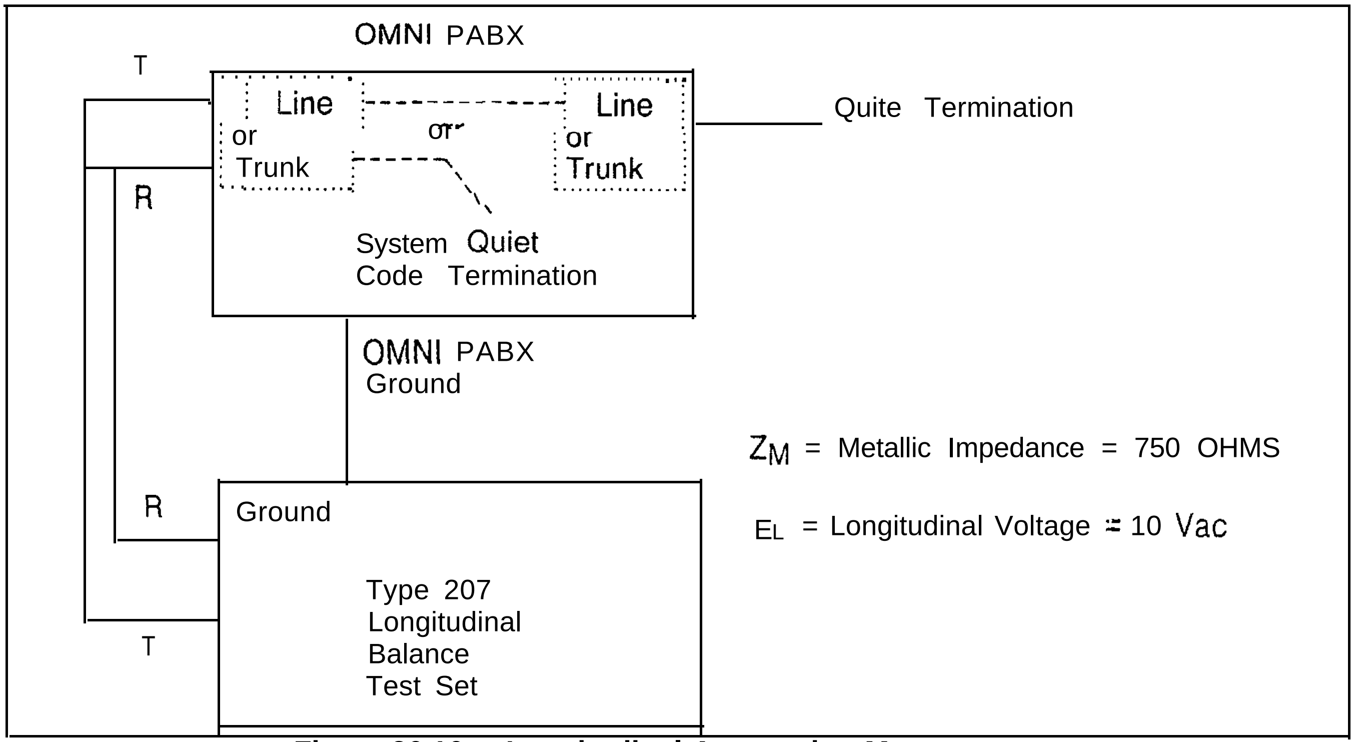


Figure 36.16 Longitudinal Attenuation Measurement

Longitudinal Susceptability

36.8.7 To measure longitudinal susceptability, proceed as follows (test objectives are listed in paragraph 36.6.14).

1. Establish a path to the system's quiet code termination or to terminated line or trunk as shown in Figure 36.17. The required battery feed arrangement is applied only to trunks.

2. Apply a longitudinal signal so as to create a tip or ring to ground voltage of 50 VAC maximum at 60 Hz or a longitudinal current of 10 ma maximum at 60 Hz.

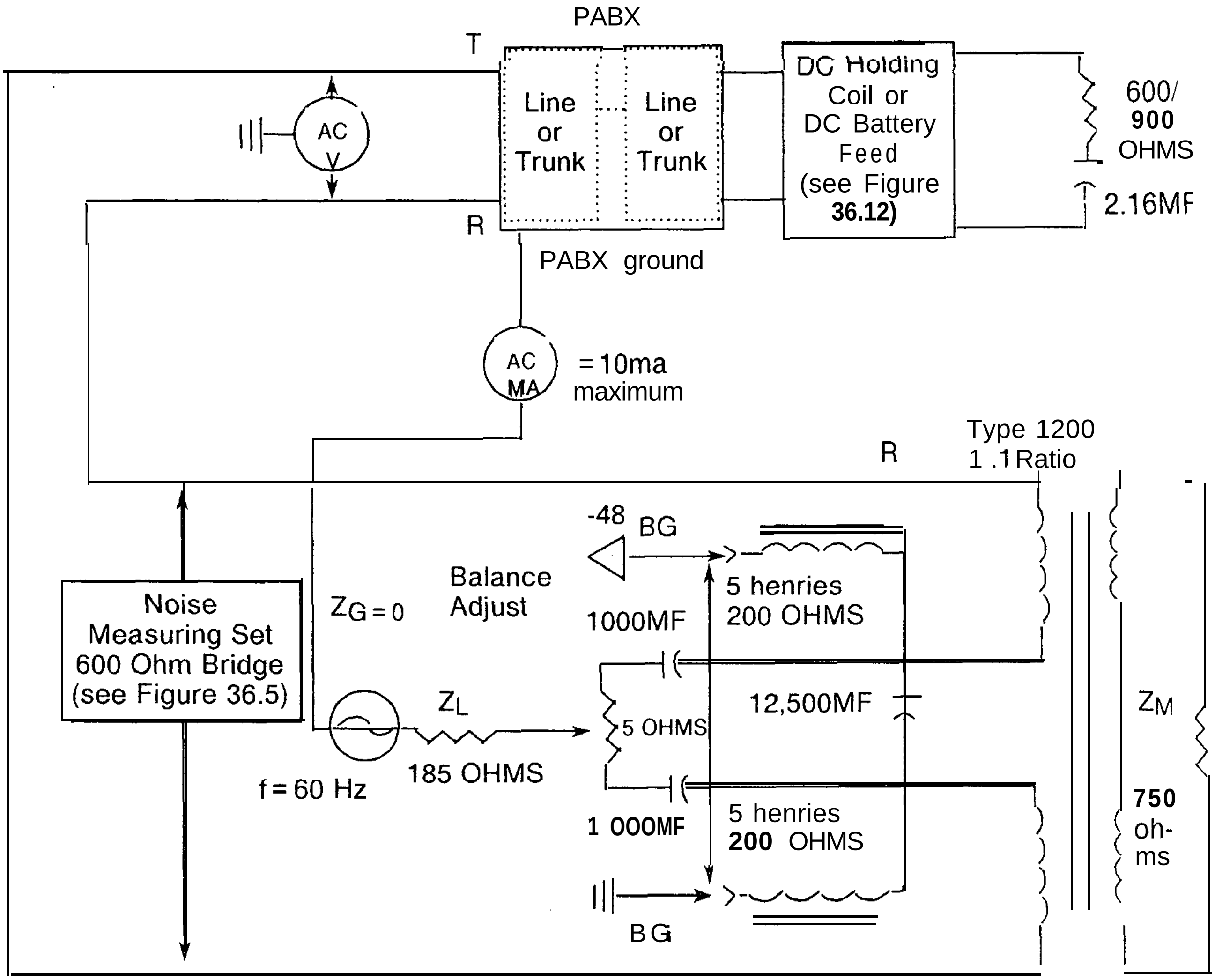
3. With a noise measuring set in the bridging mode, measure the metallic C-message noise.

Echo and Singing Return Loss

36.8.8 To measure ERL and SRL, proceed as follows (test objectives are listed in paragraph 36.6.73).

1. Establish a station-to-station, station-to-trunk, or trunk-'o-trunk path through the system. Provide DC holding for stations or incoming loop trunk circuits and provide a battery feed arrangement for CO-type loop trunk circuits as shown in Figure 37.12.

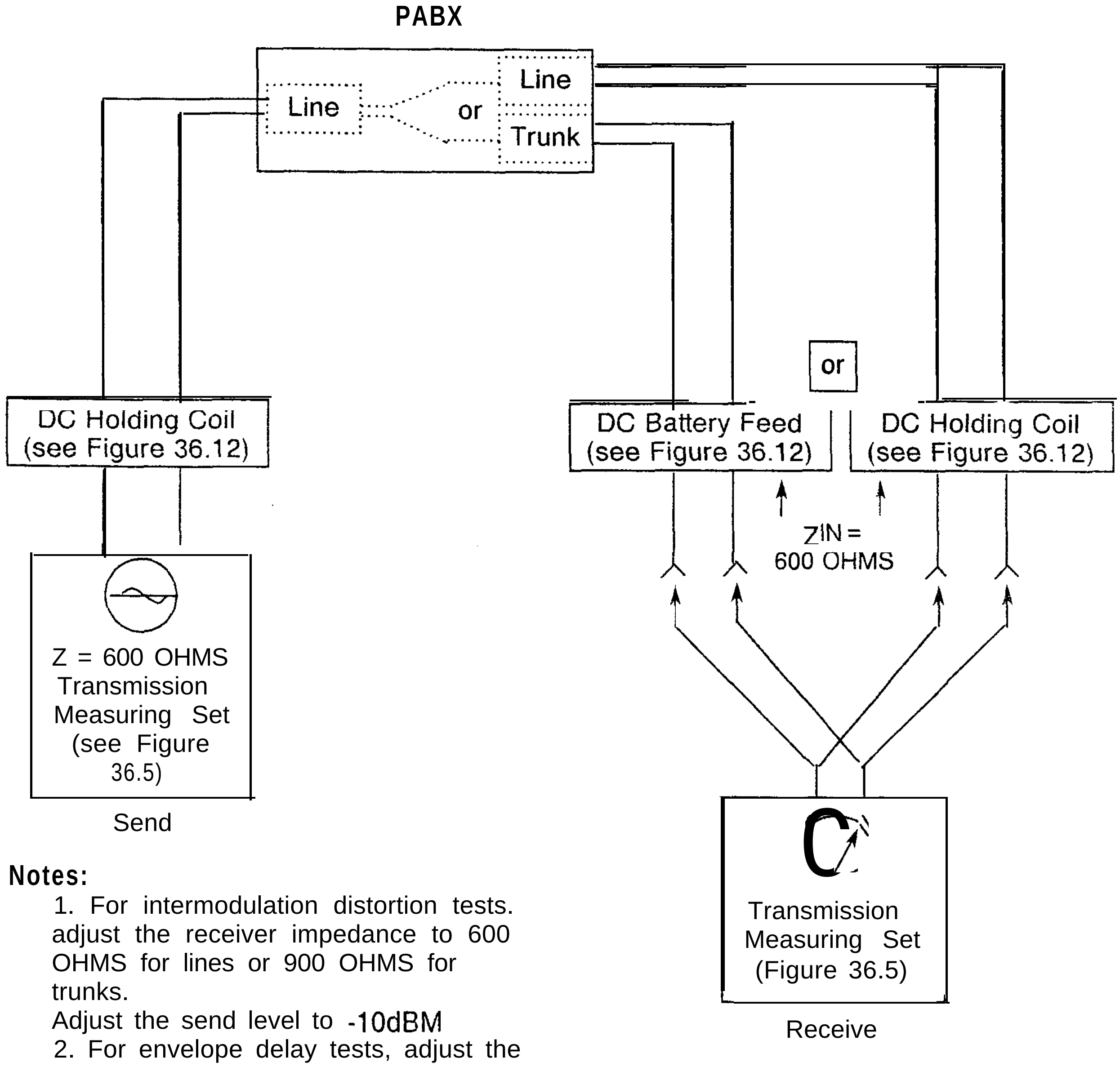
2. Connect a return loss measuring set and measure the echo and singing return losses of the path as shown in Figure 36.18.



NOTES:

1. Use LP strap for lines, BG straps for loop trunks, and NO straps for E&M trunks.
2. The longitudinal 60 Hz generator will be adjusted to provide 50 VAC measured from tip or ring to ground or 10ma of logitudinal current.

Figure 36.17 Longitudinal Susceptibility Measurement



Notes:

1. For intermodulation distortion tests. adjust the receiver impedance to 600 OHMS for lines or 900 OHMS for trunks.
Adjust the send level to -10dBM
2. For envelope delay tests, adjust the send portion for 1.804 Hz at -10dBm.
3. For ERL and SRL tests set the send portion to the quite termination position.

Figure 36.18 Intermodulation, Envelope Delay Distortion, Noise, and Return Loss Test Method

THIS PAGE IS INTENTIONALLY LEFT BLANK.

INDEX

A

B

INDEX

INDEX

D

INDEX

INDEX

H

I

K

L

M

INDEX

INDEX

R

INDEX

S

INDEX

INDEX