SBX IP 320

Features & Operation Guide

REVISION HISTORY

Release	Date	Documentation Changes	Page No.
2.0	07-08	A "Reply to Message" feature is added	245
		A "Company Directory" feature is added.	37
		A 911 feature is added.	2
		A "Mailbox Buttons" feature is added.	233
		A section is added that more clearly defines the indications as to whether new, unheard messages, or callback requests are waiting.	60
		A "DND Forward" feature is added.	41
		A "Security of Transit-Out Code with registered IP" feature is added.	140
		A "Net Firewall Routing" feature is added.	137
		A "Calling Party Number (CPN) Service" feature is added.	152
		A "DID/DISA Call Routing of DND Station or Station with Pre-selected Message Active" feature is added.	112
		A "DID Call routing with Incoming CLI" feature is added.	110
		A "Do Not Disturb (DND) with Pre-Selected Message" feature is added.	42
		A "CO Ring Assignment" feature is added.	126
		An "IP Phone Reroute Service" feature is added".	144
		The Weekly Time Table (PGM 233) is modified to include a " Lunch mode".	9
		A "Hunt Group Name Service" feature is added.	94
		A "CLI Transit" feature is added.	151
		A "Remote Mobile Extension Control" feature is added.	71
		A feature that allows the Secretary to activate the Executive/ Secretary feature from her station is added	153

REVISION HISTORY

Release	Date	Documentation Changes	Page No.
2.0	07-08	The Customer Call Routing (CCR) with VMIB feature is modified and the following sentence explains an added condition that explains the modification. "The current CCR announcement (current depth) will be provided again until the DISA retry count is over when the user does not press any digit, destination is busy, or in error cases."	108
1.0	01-08	Initial Release NOTE: that this document contains information on ISDN, DCOB, and SMS. These features are currently not supported. Information pertaining to DID pertains only to SIP Trunking.	--

Contents

System Features

System Features

This Programming Manual is designed to provide general system features operation and make references to associated Admin Programming using a Digital Key Telephone Unit (DKTU) and a PC for the SBX IP 320.

The following sections provide a functional listing of features with the description and operation of each. The features are divided into 3 parts as listed - a lead-in description preceding the following sections:

Operation: describes how to use the feature.

Conditions: explains any requirements or constraints of the feature related to its configuration.

Admin Programming: to make this feature operational, the admin programming listed in this section must be configured.

911 Feature

911 calls are placed on 911 marked lines. If all lines are busy, an existing call is dropped and the 911 call is placed.

Operation

A station user dials 911 or 9 + 911. The call is placed. If all trunks are busy, the system will drop an in-progress trunk, wait 1.5 seconds, and then place the 911 call.

Note: If the trunk access code is 9 (default), a station user must dial 9 + 911. If the System Administrator changes the trunk access code to other than 9, only 911 is dialed.

Any station user or Attendant who programs a flexible button for 911 ALERT will be alerted of internal stations placing 911 calls. The system can store the sixteen most recent 911 calls. Calling information includes the time/date of the call, as well as the station number from which the call was placed.

The initial 911 Alert indications include:

- Audible ringing tone
- Green flashing 911 ALERT flexible button LED
- Automatic LCD display of 911 call information

Programming

A CO line must have the 911 mark. PGM 141, button 11, 1/0 enable/disable.

911 feature button [TRANS/PGM + 7#] must be assigned to view the alert and log.

Conditions

- By default, CO line 1 is marked for 911.
- If a current call is dropped, the system will wait 1.5 seconds, and then place the 911 call.

Attendant Service

An attendant properly controls the incoming calls by transferring calls and accessing the unanswered calls, etc. Attendants can change simple settings of the whole system or intercom tenancy group (LCD date/time format, etc.).There are 2 types of attendants in the system. The types of Attendant and feature of each Attendant type are:

Main Attendant - a maximum of 5 stations can be defined as main attendants. Main attendants control the whole system and operation of main attendants effect the whole system. The first main attendant is called the system attendant. The system attendant can be changed, but cannot be removed.

Intercom Tenancy Group Attendant - each intercom tenancy group can have its own attendant. The intercom tenancy group attendant controls the stations belonging to the intercom tenancy group. Intercom tenancy group attendants can affect only the intercom tenancy group to which they belong.

Generally, the attendant of a station is the intercom tenancy group attendant to which the station belongs. If an intercom tenancy group attendant of the station doesn't exist, the main attendants will supply the station with attendant services.

Assign Attendant

By default, the first station (i.e., station100) is assigned as the system attendant, and other attendants are not assigned.

Main attendants can be assigned PGM 164. Intercom tenancy group attendants can be assigned PGM 120 - FLEX 1.

1. Attendant Assignment (PGM 164)
2. Intercom Tenancy Group - Attendant Assignment (PGM 120 - FLEX 1)

Condition

- An IP phone cannot be assigned as an attendant.

Attendant Call & Queuing

An Attendant call is an intercom call and a CO call to an attendant.

To make an intercom call to the attendant, a user enters the station number of the attendant or dials "0".

If a user dials "0", this rings at the assigned attendant station of the intercom tenancy group to which the station belongs. If there is no assigned station as attendant, it rings the main attendant station.

Call to any attendant will be queued, if the attendant is busy. Then, ring-back tone or MOH will be provided to the calling party (PGM 160 - FLEX 1).

Operation

To call an attendant:

1. Lift handset or press the [SPEAKER] button.
2. Dial "0".

   -or-

   Dial the station number of the attendant.

Condition

- When an attendant calls another attendant which is busy, the calling attendant will hear the busy tone and can camp-on to the called attendant.
- If an attendant activates unconditional call forward, the calls to that attendant will follow the call forward process.

Admin Programming

- Main Attendant Assignment (PGM 164)
- MOH Type (PGM 171 - FLEX 2)
- Intercom Group Attendant Assignment (PGM 120 - FLEX 1)
- Attendant Call Queuing RBT/MOH (PGM 160 - FLEX 1)

Attendant Forward

The attendant can forward (Unconditional Call Forward) a call to another station. The forwarded-to Station will substitute for the attendant temporarily, while the attendant is in the forwarded state.

Operation

It is the same procedure as the Unconditional Call Forward.

To activate Attendant Forward:

1. Lift the Handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial Call Forward Code 1 (Unconditional Call Forward).
4. Dial Station number.
5. Hang up the handset and go on-hook.

To deactivate Attendant Forward:

In an idle state, press the [DND/FWD] button.

-or-

In an off-hook state, press the [DND/FWD] button and dial #.

Condition

- If the attendant assigns unconditional call forward to an SLT or WHTU, the forwarded-to station only serves incoming calls as an attendant call or attendant recall. The forwarded-to SLT or WHTU cannot activate attendant features.

Attendant Intrusion

When an attendant has an urgent message for a station which is conversing with a CO party, the attendant can intrude upon the conversation and converse with the station and the CO line.

Operation

To intrude on a CO call while the attendant is receiving an intercom busy tone:

1. Press the programmed {ATD INTRUSION} FLEX button.
2. After the intrusion tone, converse with the station and/or the CO party.

To assign an {ATD INTRUSION} FLEX button:

1. Press [TRANS/PGM].
2. Press the FLEX button to be assigned
3. Press the [TRANS/PGM] button
4. Dial 86
5. Press the [HOLD/SAVE] button to accept changes.

Condition

- To use this feature, the Auto Privacy should be OFF (PGM 161) and Override Privilege (PGM 113 - FLEX 4) of the attendant should be Enabled.

Admin Programming

- Auto Privacy (PGM 161)
- Privacy Warning Tone (PGM 161)
- Override Privilege (PGM 113)

Attendant Override

A station in a DND state generally cannot receive an incoming call.

The attendant, however, can temporarily invalidate the DND state. Therefore, the Attendant can call and transfer to the station in a DND state.

Operation

To override a DND state at a Station while the Attendant is receiving a DND tone:

1. Dial *or the last digit of the dialed station number.

   -or-

2. Press the programmed {Camp-On} FLEX button.

   The DND warning tone will be changed to the Intercom ring-back tone at the Attendant Station.

3. The Attendant can then call a Station in the DND state.

To assign the {Camp-On} FLEX button:

1. Press [TRANS/PGM].
2. Press the FLEX button to be assigned.
3. Press the [TRANS/PGM] button.
4. Dial 85.
5. Press the [HOLD/SAVE] button to accept changes.

Conditions

- If the Attendant with a transferred CO call overrides a station in a DND state, and the station has a {CO} or {LOOP} button, the Attendant can transfer the CO call to the station in a DND state.
- If the transferred-to station has no {CO} or {LOOP} button, the CO call will be recalled to the Attendant immediately.

Attendant Recall

If the recalled CO call (transfer, hold) is unanswered by the destination station, the CO call will be directed to the Attendant. The Attendant will receive the recall ring for a time equal to the Attendant Recall Timer (PGM 180 - FLEX 1). If the Attendant doesn't answer the CO call for a time equal to the Attendant Recall Timer, the CO call will be disconnected.

Conditions

- If an Attendant for an intercom tenancy group is not assigned, the CO call is recalled to the System Attendant.
- When a call in exclusive hold is recalled to the Attendant, the call is placed on system hold.
- A Private CO line will not be recalled to the Attendant.

Admin Programming

- Attendant Recall Timer (PGM 180 - FLEX 1)
- Hold Recall Timer (PGM 180 - FLEX 5)

Change LCD Date/Time Display

The Attendant can change the LCD Date/Time display format for stations in the system.

Date: MM-DD-YY / DD-MM-YY (ex., August 4, 2008 -> 08-04-08 / 04-08-08)

Time: 12H / 24H (ex., eight thirty P.M. -> 08:30 PM / 20:30)

Operation

To change LCD Date format (toggle):

1. Press the [TRANS/PGM] button.
2. Dial 044.

To change LCD Time format (toggle):

1. Press the [TRANS/PGM] button.
2. Dial 045.

Admin Programming

- LCD Time/Date/Language Display Mode (PGM 169)

Day/Night Service

When a CO call comes into the system, the destination of the CO call can be changed according to the time of day.

There are 6 ring modes - Day mode, Night mode, Weekend mode, On-demand mode, Automatic Ring mode, and Lunch mode. The destination of a CO call can be set differently for each ring mode, while a User sets the destination of CO call with Admin programming. At Day mode/Night mode/Weekend mode, the User can set the appropriate destination of CO call according to the situation (day, night, or weekend).

On-demand mode is used to supply a different destination of CO call, except Day mode, Night mode, or Weekend mode.

Automatic Ring mode is classified as Day mode or Night mode or Weekend mode according to the Weekly Time Table (PGM 233).

Only attendant can change the ring mode. If a user presses the [DND/FWD] button at the attendant station.

Lunch mode can be used during the Day mode.

According to ATD ring mode setting, Lunch Mode can be applied in DISA service (PGM 140), Ring Assigned incoming call (PGM 144, PGM 145), DID service (PGM 231).

Operation

To change Day / Night / Weekend / On-Demand / Automatic Ring / Lunch mode:

1. Press the [DND/FWD] button at the Attendant Station to change the ring mode (1-6).
2. Press the [TRANS/PGM] button.
3. Dial 074.
4. Press the [HOLD/SAVE] button to activate ring mode.

   Note: In the Weekly Time Table, time can be set automatically.

Conditions

- The default value of the Weekly Time Table is as follows (entry number: 00). The first table entry (00) is for main Attendants and others (01-15) are for intercom tenancy group Attendants.

- Enter Start/End times in PGM 233.

DATE	DAY START TIME	LUNCH START TIME	LUNCH END TIME	NIGHT START TIME	WEEKEND START TIME
Mon	09:00	-:-	-:-	18:00	-:-
Tues	09:00	-:-	-:-	18:00	-:-
Wed	09:00	-:-	-:-	18:00	-:-
Thurs	09:00	-:-	-:-	18:00	-:-
Fri	09:00	-:-	-:-	-:-	18:00
Sat	-:-	-:-	-:-	-:-	00:00
Sun	-:-	-:-	-:-	-:-	00:00

- On-demand mode is not available in the Automatic Ring mode.
- The Attendants of an intercom tenancy group can change the ring mode as well as main Attendants. If an Attendant of an intercom tenancy group changes the ring mode, only the ring mode of the intercom tenancy group to which the Attendant belongs is changed. If a main Attendant changes the ring mode, the ring mode of the system will be changed.
- When the ring mode is set to Automatic Ring mode by the main Attendant, the ring mode of the system will follow the first table entry (entry number: 00) of the Weekly Time Table.
- If the system ring mode is changed from the night/weekend/on-demand/auto ring mode to day mode, the ring mode of all Intercom Tenancy Groups will change to the previous ring mode.
- The Lunch Mode default Value is not assigned.
- In PGM 144, Weekend mode, Lunch mode, or On-Demand mode, if Ring assigned destination is Station, Ring Delay Count is enable only 0.
- In Weekly Time Table, if input time is the same (for example, Day Start Time and Lunch Start Time are the same), the time table cannot work correctly.

Admin Programming

- DISA Attribute (PGM 140)
- CO Line Ring Assignment (PGM 144, PGM 145)
- DID Table (PGM 231)
- Weekly Time Table (PGM 233)

Disable Outgoing Access

The Attendant can take a particular CO line out of service. CO calls will not be able to be made through the CO line; incoming CO calls are not affected.

Operation

To set a CO line in/out-of-outgoing service from the Attendant:

1. Press the [TRANS/PGM] button.
2. Dial 073.
3. Press the desired {CO line} FLEX button; a confirmation tone will sound when the status (in or out-of-outgoing-service) of the selected CO line is changed.
4. Press the [SPEAKER] button to return to idle.

Conditions

- Any Attendant can use this feature.
- The LED of the {CO line} FLEX button which is out-of-outgoing-service is flashing in Attendant station but is fully lit in other stations.
- To release the-out-of-outgoing-service, press the flashing {CO line} FLEX button at the Attendant station.
- Though the desired CO line is busy, the Attendant can still make the CO line out-of-outgoing-service. The out-of-outgoing-service feature will take effect after the CO line goes to idle.

DSS/BLF Consoles

Attendants and other DKTUs may be equipped with DSS/BLF consoles which provide additional buttons for more convenient operation. The consoles are arranged as flexible mapped units. The DSS/BLF consoles are initially assigned with programming as one of 3 maps. All buttons of any map are programmable.

The DSS/BLF consoles each require a separate line connection to the KSU, and take up a station number.

Conditions

- There is no limit to the number of DSS/BLF consoles in a system.
- The default value for DSS/BLF is as follows:

Map 1	Flex 1 - Intrusion Flex 2 - All Call Page Flex 3 - Call Park 01 Flex 4 - Station Group 1 Flex 5 - Camp-on Flex 6 - Internal All Call Page Flex 7 Flex 8 Flex 9 Flex 10 Flex 11 Flex 12 Stations 100-135
Map 2	Stations 136-147
Map 3	Empty

Admin Programming

- Station ID Assignment (PGM 110 - FLEX 1)
- DSS/BLF ID Assignment (PGM 110 - FLEX 2)

ICM Box Music Selection

The Attendant can select the music channel source to provide to the intercom Box / Doorbox.

Operation

To select the music source from the Attendant:

1. Press the [TRANS/PGM] button.
2. Dial 075.
3. Dial the music source (00-12); music source will be heard, but if the music channel has no music source, no music will be heard.
   - Channel 00: Music is not used.
   - Channel 01: Internal music
   - Channel 02: External music
   - Channel 03: VMIB BGM
   - Channel 04-08: SLT MOH
4. Press the [HOLD/SAVE] button.

Admin Programming

- Intercom Box Music Channel (PGM 171)

Station Feature Cancel

The attendants can cancel features such as DND, Call Forward and pre-selected messages at other stations.

Operation

To disable active features at a station from the attendant station:

1. Press the [TRANS/PGM] button.
2. Dial 071.
3. Dial the desired station range.
4. Press the [HOLD/SAVE] button.

Call Control

Account Code

An account code is used to identify outgoing calls for accounting and billing purposes. The account code is appended to the SMDR Call record. A company uses an account code for each User Station so that the company can identify and bill (where applicable) calls made from each Station. An account code may use up to a maximum of 12digits (0-11).

Operation

To enter an account code before accessing a CO line:

1. Press the programmed {ACCOUNT CODE} flexible button.
2. Dial the account code (max of 12 digits) or the * key.
3. Intercom dial tone should be heard and a CO line is secured to make a call.

To enter an account code during a conversation with an external party:

1. Press the {ACCOUNT CODE} flexible button.
2. Dial the account code (max of 12 digits).
3. Press the {ACCOUNT CODE} flexible button. You will be reconnected to the external party.

To enter an account code without the {ACCOUNT CODE} flexible button during a conversation with an external party:

1. Press the [TRANS/PGM] button.
2. Dial 550.
3. Dial the account code (maximum 12 digits) or the * key, then reconnect to the external party.

To assign an {ACCOUNT CODE} flexible button to access the account code feature:

1. Press the [TRANS/PGM] button
2. Press the FLEX button
3. Press the [TRANS/PGM] button
4. Type 80.
5. Press the [HOLD/SAVE] button to accept changes.

Conditions

- While entering the account code, the current call is put on mute mode.
- The user may enter the account code before a call conversation is established.

Admin Programming

- Refer to the SBX IP 320 Installation Manual, SMDR Account Enter Code (PGM 106 - Flex 7)

Authorization Code

If the Station is programmed to enter the authorization code by ADMIN programming, the authorization code must be entered to access a CO line. An authorization code can be used for SMDR, DISA account code.

Note: Personal authorization codes should be kept secure by the System Attendant and individual Station Users to avoid fraudulent phone calls being made.

An authorization code is assigned as 3-11 digits as applicable; each station has a different authorization code.

Operation

To access a CO line using an authorization code:

1. When attempting CO line access, a DND warning tone will be heard. Enter the authorization code.
2. Press the # key (if 5 digits authorization code type is set according to PGM161-FLEX21, the # key is not needed). If valid, the CO line will be connected; if not valid, an error tone will be presented.

To register the authorization code on a Station:

1. Press the [TRANS/PGM] button.
2. Type 31.
3. Enter the authorization code, and press the # key (if 5 digits authorization code type is set according to PGM161-FLEX21, the # key is not needed).
4. Press the [HOLD/SAVE] button to accept changes.

To change the authorization code at a Station:

1. Press the [TRANS/PGM] button.
2. Type 32.
3. Enter the current authorization code.
4. Type in the new authorization code.
5. Press [HOLD/SAVE] to accept changes.

Conditions

- Up to 200 authorization codes can be programmed on the System.
- Duplicate authorization codes cannot be assigned to more than one Station.
- If Loop LCR ACNT is set on a Station, the authorization code is required when the station dials the Loop LCR CO Access code.

Admin Programming

- Authorization Code Table (PGM 227)
- DISA Account Code (PGM 141)
- CO Line Group Account (PGM 141)
- Loop LCR Account Code (PGM 111)
- 5 Digit Authorization Code Usage (PGM 161 - FLEX 21)
- Station Account (PGM 112 - FLEX 20)

Automatic Call Release

When a Station does not initiate dialing on an outgoing CO line or receives a no answer on an intercom call, the System will disconnect the call based on the assigned Auto Release Timer (PGM 180 - FLEX 14). If the User Station is in speaker phone or [SPEAKER] mode, the Station will return to idle, otherwise the station will receive an error tone if using the handset.

Conditions

- An Intercom call in H/F mode is considered answered and Station Auto Release will not be activated.
- When the Automatic Release time is assigned as 0, Auto Call Release is not activated.
- While making a call without lifting the handset and the Auto Release timer expires, the call will be canceled and the station will return to an idle state automatically.
- While making a call using the handset and the Auto Release timer expires, the call will be canceled and the station will receive an error tone.

Admin Programming

- Automatic CO Release Timer (PGM 180 - FLEX 14)
- Station Auto Release Timer (PGM 182 - FLEX 5)

Class Of Service (COS)

Each station and CO line may be assigned to have different classes to allow or restrict call service. The level of COS assignments are programmed at each Station and CO line. Applied dialing restrictions are the result of the interaction of COS assignments as listed in the Class of Service Table.

CLASS OF SERVICE

CO LINE COS						
S		1	2	3	4	5
T	1	Unrestricted	Unrestricted	Unrestricted	Canned Restricted	Unrestricted
A	2	Table A	Table A	Table B	Canned Restricted	Unrestricted
T	3	Table B	Unrestricted	Unrestricted	Canned Restricted	Unrestricted
I	4	Table A, B	Table A	Table B	Canned Restricted	Unrestricted
O	5	Canned Restricted2	Canned Restricted2	Canned Restricted2	Canned Restricted1	Unrestricted
N	6	Canned Restricted1	Canned Restricted1	Canned Restricted1	Canned Restricted1	Unrestricted
C	7	Intercom Only	Intercom Only	Intercom Only	Intercom Only	Intercom Only
O	8	Table C	Table C	Unrestricted	Canned Restricted1	Unrestricted
S	9	Table D	Table D	Unrestricted	Canned Restricted1	Unrestricted

Canned, Restricted1: Long distance call is not allowed (8 digits maximum)

Canned, Restricted2: Long distance call is not allowed (longer than 8 digits)

STATION COS

COS 1	There is no restriction to dial.
COS 2	Monitored by Exception Table A
COS 3	Monitored by Exception Table B
COS 4	Monitored by Exception Table A & B
COS 5	Long distance call is not allowed; longer than 8 digits
COS 6	Long distance call is not allowed; max. 8 digits may be dialed
COS 7	Only intercom, paging and emergency calls are allowed; no dialing allowed on CO lines
COS 8	Monitored by Exception Table C
COS 9	Monitored by Exception Table C

CO COS

COS 1	There is no restriction. Monitored by STA COS.
COS 2	Monitored by Exception Table A & STA COS 2/4.
COS 3	Monitored by Exception Table B & STA COS 2/4
COS 4	Long distance call is not allowed for all STA COS; max. 8 digits may be dialed
COS 5	Overrides STA COS 2, 3, 4, 5, 6, no COS restriction

CO TO CO LINE COS

COS 1	There is no restriction to dial
COS 2	Monitored by Exception Table A
COS 3	Monitored by Exception Table B
COS 4	Monitored by Exception Table A & B
COS 5	Long distance call is not allowed; longer than 8 digits
COS 6	Long distance call is not allowed; only max. 8 digits may be dialed
COS 7	Only intercom, paging and emergency call are allowed; no dialing allowed on CO lines
COS 8	The assignments in the Exception Table C are monitored for allow and deny numbers
COS 9	The assignments in the Exception Table D are monitored for allow and deny numbers

PBX Dialing Codes-There are 5 PBX access codes (2 digits) to enter the system and access a CO line via PBX. A CO line marked as a PBX line will not be governed by any station or CO line COS until a recognized PBX code is dialed.

Exception Table A & B-There are two exception tables with COS. Each table has 20 allow codes and 10 deny codes and a code may have eight entries.

Conditions

COS Rules

- In STA COS 7, no dialing is allowed to CO lines.
- In CO COS 5, STA COS 1-6 is ignored and there is no restriction to access to CO lines.
- In CO COS 4, STA COS 1~6 is ignored and long distance calls are not allowed; max. 8 digits may be dialed.
- In CO COS 1, it is restricted by STA COS.
- In CO COS 2 and STA COS 2/4, it is restricted by Exception Table A.
- There is no restriction in STA COS 1/3.
- In STA COS 5, long distance calls are not allowed; max. 8 digits can be dialed.
- CO line Allow/Deny Restriction Rules
- If there are no entries, no restriction is provided by the table.
- If there are entries in the Deny table, then the restriction is on a Deny Only basis.
- If there are entries in the Allow table, then the restriction is on an Allow Only basis.
- If there are entries in both the Allow and Deny Tables, the Allow Table is searched, if the dialed number matches an entry in the Allow Table, the call is allowed. If a match is not found, the Deny Table is searched and if a match is found in the Deny Table the number is restricted. Otherwise, the number is allowed.

General Conditions

- If Incoming CO Call Toll Check is set, the COS rule is applied when the station dials digits after answering incoming CO calls.

Admin Programming

- Station COS (PGM 116)
- CO line COS (PGM 141 - FLEX 2)
- CO-to-CO COS (PGM 166)
- Toll Exception Table (PGM 224)
- Canned Toll Tables (PGM 225)
- Incoming Toll Check (PGM 161 - FLEX 16)

System Speed Zone

Up to 10 speed number zones can be defined. Speed bins & Stations can be allocated to these zones. Toll checks based on COS can be applied to zones. Only Stations allocated to zones can access these bins. Speed bins not allocated to zones can be accessed by all Stations and no toll checks are applied.

Admin Programming

- Speed Dial Access (PGM 112 - FLEX 9)
- System Speed Zone (PGM 232)
- CO Dial Tone Detect (PGM 160 - FLEX 6)

Walking Class of Service (Walking COS)

This feature allows temporary override of the toll restriction and allows a toll call from previously toll restricted phones. The authorization code can be used as a verified account code for SMDR.

Operation

To activate Walking COS from a DKTU:

1. Press the [TRANS/PGM] button.
2. Dial 23; the confirmation tone sounds.
3. Enter the authorization code; the intercom dial tone sounds and the used extension COS is temporarily changed.
4. A CO line call can be placed only one time.

To program {Walking COS} on a flexible button:

1. Press the [TRANS/PGM] button.
2. Press the applicable FLEX button.
3. Press the [TRANS/PGM] button.
4. Dial 2 3.
5. Press the [HOLD/SAVE] button to accept changes.

Conditions

- Can be used on DKTUs and SLTs.
- This feature is available on a per-use basis only. While activating walking COS, hanging-up or pressing the [TRANS/PGM] button to hold the call and seize another line, the original programmed Station COS will be used.
- When a wrong number is dialed, press the [FLASH] button to dial again without changing to an idle CO line.
- The fee for a call with Walking COS will be charged according to the Station authorization code, not the actual Station.
- When a User tries to use Walking COS at a station set to COS 7 with temporary COS, the call will follow the original COS of the Station.

Admin Programming

- Authorization Code Table (PGM 227)

Call Handling

Absent Text Message

Custom Message

Each station can select from ten (11-20) available custom messages to display on the DKTU LCD. These messages are programmed by the System Attendant for System-wide use. Individual users may program message 00 as their own custom message.

When set, the selected message is displayed on the User Station LCD panel.

Operation

To program Custom Message 00 from a station:

1. Press the [TRANS/PGM] button.
2. Dial 52.
3. Enter the message, up to 24 characters.
4. Press the [HOLD/SAVE] button and a confirmation tone should sound.

. – 13 Q – 11 Z – 12 1 – 10	A – 21 B – 22 C – 23 2 – 20	D – 31 E – 32 F – 33 3 – 30
G – 41 H – 42 I – 43 4 – 40	J – 51 K – 52 L – 53 5 – 50	M – 61 N – 62 O – 63 6 – 60
P – 71 Q – 72 R – 73 S – 74 7 – 70	T – 81 U – 82 V – 83 8 – 80	W – 91 X – 92 Y – 93 Z – 94 9 – 90
Blank – ∗1 : – ∗2 , – ∗3	0 – 00	

To program Custom Messages 11-20 by the System Attendant:

1. Press the [TRANS/PGM] button.
2. Dial 053 and the message number (11-20).
3. Enter the message, up to 24 characters.
4. Press the [HOLD/SAVE] button and confirmation tone is heard.

To activate LCD Messages (Custom/Pre-selected) by a Station user or by a System Attendant:

1. Press the [TRANS/PGM] button.
2. Dial 51.
3. Dial the 2-digit message code 00 or 11-20.
4. Press the [HOLD/SAVE] button.

To cancel LCD Messages (Custom/Pre-selected) by a station user:

1. Press the flashing [DND/FWD] button or, Press the [TRANS/PGM] button.
2. Dial 51 and the desired number.
3. Press the [HOLD/SAVE] button.

To cancel LCD Messages (Custom/Pre-selected) by the System Attendant:

1. Press the [TRANS/PGM] button.
2. Dial 052.
3. Press the [HOLD/SAVE] button.

Admin Programming

- Refer to the SBX IP 320 Installation Manual, Numbering Plan (PGM 106)

Pre-selected Message

A User can choose from pre-selected messages (1-10) to be shown on the calling party LCD. Detail information is entered by each User (e.g. Time, Date or Station number).

The pre-selected messages include:

Message 01: LUNCH, RETURN AT HH:MM

Message 02: ON VACATION /RETURN AT DATE MM:DD

Message 03: OUT OF OFFICE/RETURN TIME HH:MM

Message 04: OUT OF OFFICE/RETURN AT DATE MM:DD

Message 05: OUT OF OFFICE/RETURN UNKNOWN

Message 06: CALL: (Telephone No: Up to 17 digits)

Message 07: IN OFFICE: STA XXXX

Message 08: IN A MEETING/RETURN TIME HH:MM

Message 09: AT HOME

Message 10: AT BRANCH OFFICE

Operation

To activate LCD Messages (Custom/Pre-selected) from a Station or from the System Attendant:

1. Press the [TRANS/PGM] button.
2. Dial 51.
3. Dial the 2-digit message code 00 or 01-10.
4. Press the [HOLD/SAVE] button.

To cancel LCD Messages (Custom/Pre-selected) from a Station:

1. Press the flashing [DND/FWD] button

   -or-

   Press the [TRANS/PGM] button.
2. Dial 51 and the desired number.
3. Press the [HOLD/SAVE] button.

Conditions

- If a station assigns call forward while a pre-selected message is activated, the pre-selected message is canceled automatically).
- Users can leave a message at a DKTU or SLT. When leaving a message at the SLT, a DND warning tone will sound when the handset is lifted as a reminder of the message waiting.
- When a pre-selected message is selected in a DKTU, the [DND/FWD] button will flash.

Admin Programming

- Refer to the SBX IP 320 Installation Manual, Numbering Plan (PGM 106)

Alarm

The system can be programmed to recognize the status of an external contact from a relay (open or closed). When activated, the System will signal programmed Stations with a single tone repeated per 1-min. interval or a continuous tone. This capability is commonly employed to provide remote alarm signals. When used as an alarm, the assigned Stations receive the programmed signal. To stop the signal, the alarm should be deactivated (reset) from a Station assigned to receive the alarm signal.

Operation

Upon detecting the Alarm contact, the system sends the appropriate alarm signal to assigned stations.

To terminate the alarm signal while the line is in an idle state:

1. Dial 565; a confirmation tone sounds.
2. The alarm signal will be terminated at all assigned stations.

Conditions

- An external contact should be connected to the alarm input.
- The alarm contacts should be dry (no voltage/current source connected).
- SLT cannot receive alarm signals.
- Alarm Reset can be programmed on a flexible button.

Admin Programming

- Alarm Enable (PGM 163 - FLEX 1)
- Alarm Contact Type (PGM 163 - FLEX 2)
- Alarm Mode (PGM163 - FLEX 3)
- Alarm Signal Mode (PGM 163 - FLEX 4)
- Station Alarm Attribute (PGM 113 - FLEX 10)

Analog CLI Display

The SBX IP 320 System supports the following CLI protocol specifications:

- Bellcore GR-30-CORE & SR-TSV-002476 & ETSI ETS 300 659, ETSI ETS 300 778
- Denmark, TDK-TS 900 216
- Sweden, Telecom specification 8211-A112, Standard SS 63 63 25
- India DTMF, TEC Standard No. S/ASF-01/02
- Brazil DTMF, 220-250-713 (STANDARD) Issue 01, November 1993

Operation

When an incoming CO call is received at a CO line board:

The DSP chip on MBU detects CID signal and CLI information is distributed to station.

Conditions

- For CLI display on the SLT, a Ring Phase (PGM 182 - FLEX4) more than 4 seconds is recommended.
- If the ring phase is less than 4 seconds, some of the SLTs will not receive CLI display signal.

Admin Programming

- CO CID ATTR (PGM 147)

Automatic Fax Transfer

The system will determine if an incoming call from the preprogrammed CO line is for a FAX or for a speech terminal by detecting the tone of the call (1100Hz, 0.5s ON/3s OFF repeated). When the system detects a FAX tone from the incoming line within the predetermined time, the System will transfer the call to the appropriate FAX Station. If the FAX tone is not detected within the predetermined time, the System will transfer the call to the appropriate Station(s).

Conditions

- Only one CO line can be programmed as a FAX CO line. If the FAX CO line is not programmed, Automatic Fax Transfer will not be activated.
- Station 17 is used as the FAX Station. The FAX machine should be connected to the port for Station 17 to use this feature.
- If the FAX CO line is not answered within the FAX CO call time, the incoming call will be disconnected.
- An outside caller connected to the FAX CO line will hear a ring back tone while the system is detecting a FAX tone.
- To transfer calls from the FAX to a FAX station, do not assign CO ring to the FAX Station 17.
- If a CO line is programmed for DISA and for Automatic FAX Transfer as well, incoming calls from that CO line are served as DISA calls. So, if a user wants to call the FAX station, just call station 17 by exploiting DISA call.
- When the FAX machine goes idle after a FAX call, the associated CO line is released.
- If the FAX CO line is disconnected during a FAX call, the CO line is released and the FAX machine will return to an idle state.
- Only Analog lines are enabled for using this feature.

Admin Programming

- Auto FAX Transfer CO (PGM 161 - FLEX 17)
- FAX tone Detect Timer (PGM 182 - FLEX 13)
- FAX CO Call Timer (PGM 182 - FLEX 14)

Automatic Privacy

Automatic privacy allows a Station User to suspend automatic privacy for an existing CO line conversation without invitation. By default, all conversations that take place on CO lines, the Intercom, and Conferences are protected by privacy (Automatic privacy).

Note: ADMIN programming (PGM 161 - FLEX 5) is required to enable or disable this feature.

If automatic privacy is enabled, when pressing a busy CO line button, a busy tone will be heard.

If automatic privacy is disabled, when pressing a busy CO line button, the Station is connected to the conversation in progress.

Conditions

- When Automatic Privacy is disabled, privacy is still activated for Intercom and Conference calls.
- A Station can only override a privacy-disabled Station.
- The Station will present an Intrusion tone when another Station accesses the line.

Admin Programming

- Auto Privacy (PGM 161 - FLEX 5)
- Privacy Warning Tone (PGM 161 - FLEX 6)
- Override Privilege (PGM 113 - FLEX 4)

Barge In

Barge in permits an authorized extension to intrude into other existing outside/internal calls. Barging In establishes a conference call.

There are Two Barging In operations.

Monitor

The intruding extension can listen to the existing conversation.

Speech

The intruding extension can join the existing conversation.

Operation

Monitor:

1. Call the busy station and receive the busy tone.
2. Press the "MONITOR" soft button.
3. Caller can listen to the existing conversation. The others hear a "Warning Tone".

Speech:

1. While you are listening to the existing conversation, press the "JOIN" soft button. The others hear an "Intrusion Tone".
2. The Supervisor can cancel the conversation with "DROP" soft button.
3. The Supervisor can exit from the barge-in by hanging up the phone.

Conditions

- This feature is only supported for DKTUs with 3 soft buttons.
- This feature is only available for a called party in talk state (with CO Line or Station).
- Emergency supervisor is connected only to attendant station when Executive station is called.

Admin Programming

- Barge In mode (PGM 113 - FLEX 13)

Background Music (BGM)

A user can listen to Background Music (BGM) through the speaker while handset is on-hook and the line is in an idle state.

Music from the source is heard over the station speaker and will be automatically shut-off when a call or paging announcement is received, or when the station is off-hook.

Operation

To assign background music at a station, use the following procedure:

1. Press the [TRANS/PGM] button.
2. Dial 73; the music will play.

To transmit background music on an external page port at the Attendant Station, use the following procedure:

1. Press the [TRANS/PGM] button.
2. Dial 076 (External port).
3. Dial the background music channel number; the music will play.
4. Press the [HOLD/SAVE] button. After hearing the confirmation tone, the station should go to an idle state, and the selected background music will be transmitted on an external port.

To transmit background music through the intercom box at an Attendant Station:

1. Press the [TRANS/PGM] button.
2. Dial 075.
3. Dial the background music channel number and the selected background music is heard.
4. Press the [HOLD/SAVE] button.
5. After hearing the confirmation tone, the station should go to an idle state, and the selected background music will be transmitted over the intercom box.

Conditions

- When lifting the handset or pressing the [SPEAKER] button at a Station, the music is automatically shut-off.
- When external music is assigned, the music source should be connected to the MBU.
- The same music source can be used with MOH.
- Press the [VOLUME] button to adjust volume while the background music is heard.
- The BGM may be blocked in an intercom box / Doorbox by pressing the [DND/FWD] button.

Admin Programming

- Background Music Type (PGM 171 - FLEX 1)

Call Log

The Call Log feature enables the 7208D & 7224D telephone User to view a log of the last (15-50) incoming and outgoing calls. The User can scroll through the list of numbers stored, select the desired number, and activate a redial to that number.

The log includes the CLI (or dialed number), the time, the date and Station/System Speed name of the call. It is stored on the MBU and is retained if the Station is unplugged or replaced.

The Call-log for incoming /Outgoing/Lost calls is available if a flexible button is programmed to be used for {CALL LOG}.

Operation

To program a Call Log flexible button:

1. Press the [TRANS/PGM] button.
2. Press the flexible button to be assigned.
3. Type the Call Log code, then press the [TRANS/PGM] button.
4. Dial 57.
5. Press the [HOLD/SAVE] button.

To use the Call Log feature of the 7208D & 7224D:

1. Press the programmed {CALL LOG} button.
2. You can scroll through the numbers by pressing the up/down navigation buttons to locate received calls, dialed calls, and lost calls.
3. When the CLI of the desired number is displayed, press the OK key. The System will establish a call to that number.
4. Select the Lost Call Menu to view CLI MSG-Wait numbers. A number in the Lost Call list can be answered, deleted, or saved by pressing the Select soft button.

Conditions

- To use this feature, a flexible button must be programmed for (CALL LOG).
- A User can program the Call Log List number, in the range of 15-50, per Station (PGM 160 - FLEX 19)
- When the Call Log List number is programmed, the All Call Log database will be initialized.
- The Maximum Call Log List per System is 500.
- The System assigns the Call Log list from the first available port in order.
- Must enable CLI MSG-Wait to be able to use the Lost Call Log.

Admin Programming

- Call Log List Number (PGM 160 - FLEX 19)
- CLI MSG-Wait (PGM 114 - FLEX 4)

Camp-on

When you call a busy Station, the busy tone sounds. You can give a signal to the busy Station. The busy Station (off-hook or on speakerphone) is notified of the call waiting by a camp-on tone and the [HOLD/SAVE] button LED flashes.

An SLT User may notify a busy Station of an outside CO call or internal call waiting (camp-on tone).

Operation

To activate Camp-on while receiving the Intercom busy tone, use the following procedure:

Dial * or the last digit of the busy station.

-or-

Press the busy DSS flexible button

-or-

Press the flexible button assigned as {Camp-On}.

To answer a Camp-on call while receiving the Intercom busy tone, use the following procedure:

1. Press the [HOLD/SAVE] button; the active CO line is placed on exclusive hold and the call waiting is connected.
2. Press the [HOLD/SAVE] button, to alternately talk with both parties.

To activate Call Waiting on an SLT while receiving the Intercom busy tone, use the following procedure:

1. Go Off-hook, then dial * or the last digit of the dialed Station number.
2. When answered, the call should be announced.

   -or-

   Replace the handset and go on-hook.

To answer the call waiting/camp-on, use the following procedure:

1. When the SLT station is busy, the camp-on tone is heard indicating a camp-on call.
2. Hook-flash and dial 560.
3. The SLT Station should be connected to the camp-on call.
4. Hook-flash and dial 56 again, to connect to the original Station call.

To assign the {Camp-On} button as a flexible button:

1. Press [TRANS/PGM].
2. Press the FLEX button to be assigned.
3. Press the [TRANS/ PGM] button.
4. Type 85.
5. Press the [HOLD/SAVE] button to save changes.

Conditions

- During a conference or paging, Call Waiting is not activated.
- Camp-on is not applied to a station which is in DND mode.
- The Attendant can override a Station using the Camp-on feature.
- If the Stop Camp-on Tone (PGM 112 - FLEX 15) is set to ENABLE, the camp-on tone will not be heard.

Admin Programming

- Stop Camp-on Tone (PGM 112 - FLEX 15)
- Voice Over (PGM 113 - FLEX 6): Voice over is also applied to SLT.

Change Ring Type

The ring tone signal used to notify Stations of an incoming call can be changed in ADMIN Programming to provide distinctive ringing on a per CO line basis. A distinctive ring tone can be programmed for each CO line that is used to ring each Station.

Admin Programming

- CO Distinct Ring (PGM 142 - FLEX 5)
- Ring Frequency (PGM 422)

Chime Bell

If the Chime Bell Activate Station presses Chime Bell button, Chime Bell Receive Station starts to ring.

To program a flexible button as the "Chime Bell button":

1. Press the [TRANS/PGM] button and the flexible button to be assigned.
2. Dial the code ([TRANS/PGM] + *9).
3. Press the [HOLD/SAVE] button.

Operation

To Activate the Chime Bell:

Press the Chime Bell button at the Chime Bell Activate station

To program a flexible button as the "Chime Bell button":

1. Press the [TRANS/PGM] button and the flexible button to be assigned.
2. Dial the code ([TRANS/PGM] + *9)
3. Press the [HOLD/SAVE] button.

Conditions

- The ring stops when Chime Bell Timer expires.
- Chime Bell Ring cannot be answered.
- If Chime Bell Receive station is busy or in an Off-Hook state, the station gets Mute ring instead.
- The Chime Bell master (active station) and slave (receive station) station should be keysets.

Admin Programming

- Chime Bell Station Pair (PGM 184 - FLEX 1)
- Chime Bell Relay (PGM 184 - FLEX 2)
- Chime Bell Timer (PGM 184 - FLEX 3)
- Chime Bell Frequency (PGM 184 - FLEX 4)

Company Directory

Description

This feature allows a caller reaching CCR menu to utilize the DTMF keys to "spell" the name of a subscriber and be directed to their extension.

Operation

When an outside caller presses the programmed CCR digit, the system will prompt "Press one to search by first name", "Press two to search by last name", then "Enter the first 3 characters of the persons first name" or "Enter the first 3 characters of the persons last name"

Once 3 characters have been entered, the system will check the entered characters for a match to the programmed name in the VMIB field in PGM 110, Flex 3 for first name, Flex 4 for last name.

If a match is found, the system will play "transferring to [programmed name]" and transfer the call to the station.

If more than one match is found, the system will play:

- "for [subscriber name], press 1"
- "for [subscriber name], press 2"

When the caller enters the desired digit, the system will route the caller to that station.

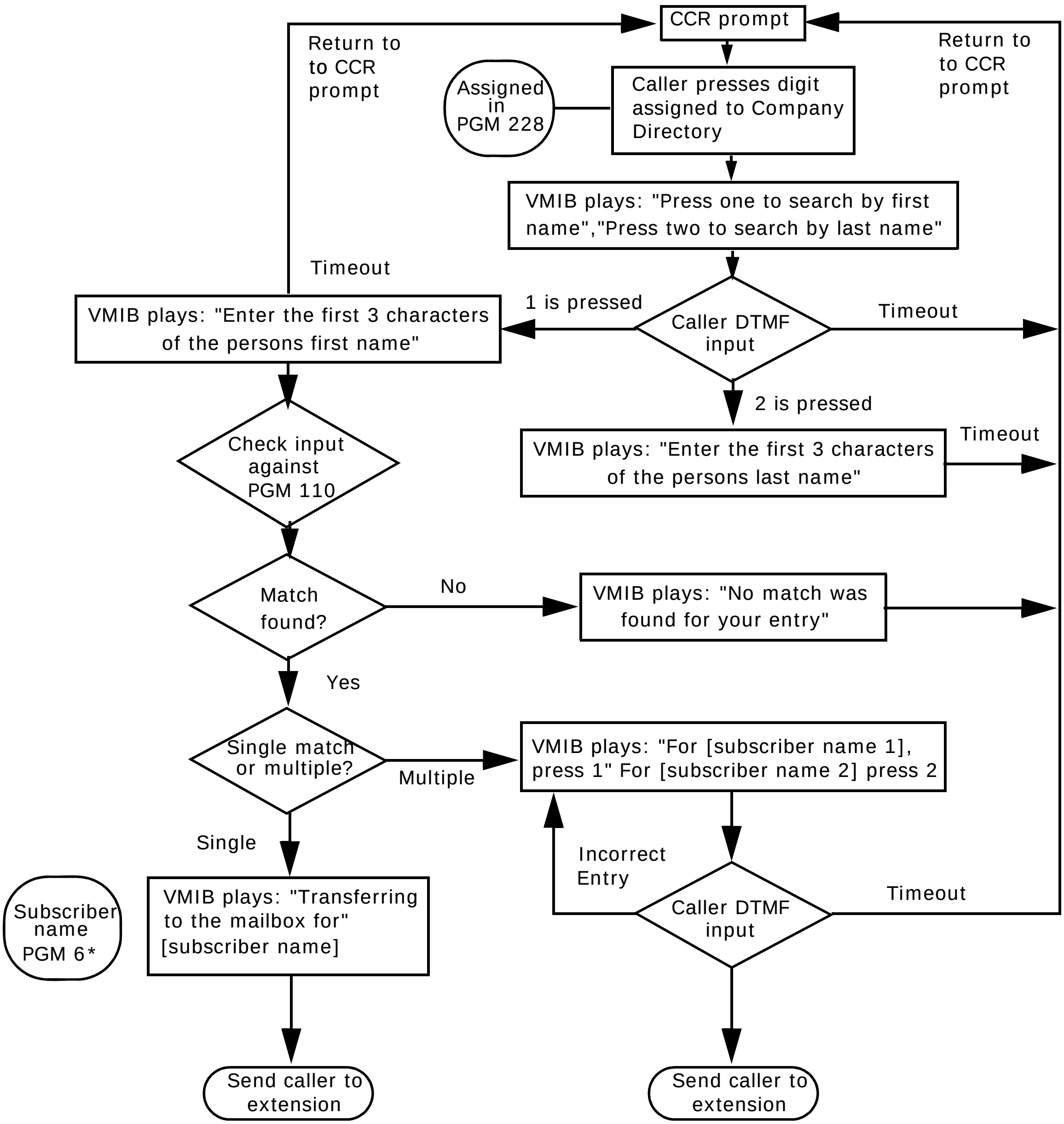
Conditions

- The field for station user name, PGM 74, is a user-settable option and will not affect the searchable name directory.
- The first name and last name fields will contain a maximum of 12 characters each.
- To record a VMIB Subscriber Name prompt, use 6* in station programming.

Admin Programming

- PGM 110 - FLEX 3 for first name
- PGM 110 - FLEX 4 for last name
- PGM 228 - CCR option 12 (Company Dir)

Company Directory Flow Diagram



Data Line Security

Once Data Line Security (PGM 111 - FLEX 4) is set on a Station, communication between the Station and another party is protected from signals such as overriding and Camp-on to the Station.

This status should primarily be used for modems and fax stations.

Conditions

- If an analog extension (Data, Fax, Modem), assigned data line security, makes an external call via ISDN line, 3.1KHz Setup message will be sent to an ISDN CO line instead of the speech Setup message.
- If a busy station which is assigned data line security, receives an incoming CO call, the call is disconnected regardless of the DID/DISA Destination setting.

Admin Programming

- Data Line Security (PGM 111 - FLEX 4)

Dialing Security

This feature allows you to prevent he dialed phone number from being displayed on the LCD of the called station when calling with a speed dial number.

Operation

To program a speed dial number with security dialing:

Note: Speed dial starting with * is operated with dialing security.

1. Press the [TRANS/PGM] button.
2. Press the [SPEED] button.
3. Dial a speed dial bin number.
4. Press the * button.
5. Dial the desired speed number.
6. Press the [HOLD/SAVE] button.

To activate a speed dial with security dialing:

Press the [SPEED] button + dial the Speed dial bin number which was programmed with dialing security.

Conditions

- Dialed phone numbers with Dialing Security are included in SMDR information.
- This feature applies to transferred or recalled CO calls.
- This feature applies to redial calls.

Do Not Disturb (DND)

Placing a station in DND eliminates incoming outside line ringing, intercom calls, transfers, and paging announcements. While in DND, the station will not receive calls on CO lines. The Attendant can override a station in DND. Stations in DND can continue to make normal outgoing calls. Station users can individually have the ability to place their station in DND (PGM 114). By default, DND is available at all stations.

Operation

To activate DND from a DKTU:

Press the [DND/FWD] button.

To deactivate DND from a DKTU:

Press the [DND/FWD] button again.

Conditions

- Pressing the [DND/FWD] button during call forward or while a pre-selected message is active, will not activate DND; call forward or the pre-selected message will be released.
- When a Station assigned to preset call forward is in DND mode, and an incoming call is received at the next station by ring assignment.
- An Attendant can override a station which is in DND by using Camp-on or Intrusion.
- An Attendant may cancel DND for one or all stations.

Admin Programming

- Do-Not-Disturb (PGM 111)

Do Not Disturb (DND) Forward

This feature modifies the operation of calls presented to a station in the DND condition.

Operation

When an internal or external call is presented to a station in DND, the system will check a programmable parameter and then send the call to the appropriate destination.

Conditions

- When the subscriber's mailbox is on the VMIB, the subscriber must record a mailbox greeting in order for calls in DND state to forward to the mailbox.
- If the subscriber's mailbox is on the VMIB and there is not greeting recorded, calls to a station in DND will be forwarded to the attendant.
- When DND forward is enabled, under no condition should an incoming CO call be dropped.

Admin Programming

- PGM 113 - FLEX 19 (DND Forward to Voice Mail) 0 or 1 disable/enable. The default value is enabled.

Do Not Disturb (One Time DND)

One Time DND allows a Station User to turn off muted ringing that occurs when off-hook (handset or [SPEAKER]) while on another call. The station user, while off hook, depresses the DND button which eliminates muted ringing. When the station goes on-hook, the DND button is extinguished and DND is cancelled.

Operation

To activate One-time DND from a DKTU:

While in an off-hook state or connecting a CO line or intercom call, press the [DND/FWD] button.

The station is in DND; the [DND/FWD] button LED lights.

When the station goes to an idle state, DND is released at the station; the [DND/FWD] button LED will be extinguished.

Admin Programming

- Do-Not-Disturb (PGM 111 - FLEX 3)

Do Not Disturb (DND) with Pre-Selected Message

A station can be programmed with a pre-selected message option of [P-MSG DND] Admin - ringing or activating as DND.

If [P-MSG DND] Admin is ON for a station, then CO and ICM ring is treated just like the DND feature:

- A pre-selected message will be shown to a calling station with a busy tone for an ICM call.
- A busy tone with a busy message will be provided to a caller of an external DID user.
- For DISA CO, an external user will hear a "Do not disturb" voice message and they can retry again later.

Operation

[P-MSG DND] is set to OFF:

1. Station receives ring for incoming call.
2. Caller hears ring-back tone.

[P-MSG DND] is set to ON (default):

1. Station does not receive ring for the incoming call.
2. Caller hears a busy tone.

Conditions

- This feature cannot be used when the CO direct Ring Assign feature to station is set (PGM 144).

Admin Programming

P-MSG DND (PGM 113 - FLEX 16)

Emergency Intrusion

If an Emergency Supervisor Station calls another station and the Supervisor receives a busy tone, the Supervisor can dial the Emergency Intrusion code and converse directly with the called station.

Operation

To activate Emergency Intrusion:

Dial the Emergency Intrusion code '#' while encountering a busy tone.

Conditions

- An Emergency Intrusion call is only possible when the calling station is set to be an Emergency Supervisor in Admin programming.
- If the station was talking to another station or CO line, the call is disconnected when the station receives an Emergency Intrusion call.
- The busy station is connected to the Emergency supervisor after a short beep tone indication.
- An Emergency Intrusion call is not possible to a station in a DND state.
- An Emergency Intrusion call is not possible to an Attendant.
- An Emergency Intrusion call is not possible to a Net DSS.
- An Emergency supervisor is connected to the Master station of a Linked pair if both the Master and Slave are busy.
- An Emergency supervisor is connected to the Secretary Station instead of Executive station.

Admin Programming

- Emergency Supervisor (PGM 112 - FLEX 24)

Extend CO-to-CO Connection

When a call is made between two analog CO lines using DISA or off-net call forward, the call duration is limited by the unsupervised conference timer. After the unsupervised conference timer expires, the call will be dropped by the SBX IP 320 system. This feature is enabled to extend the unsupervised conference time for as long as the caller wants. If this feature is activated, the DTMF receiver device is assigned to caller-side CO lines.

Operation

When the CO-to-CO unsupervised conference timer extend feature is activated, two analog CO lines on a conversation will hear a warning tone 15 seconds before the unsupervised conference timer expires and the call will be disconnected.

To extend call duration, use the following procedure:

1. Dial the unsupervised conference timer extension code number, and the extension time multiple digit data (1-9).
2. The SBX IP 320 system will re-assign the unsupervised conference timer to the multiple of the entered digits.

   Ex.) The unsupervised conference timer is set to 10 minutes and the entered digit is 3, then the timer will be extended to 30 minutes.

Conditions

- To use this feature, at least one IDLE DTMF device must exist.
- This feature is only available on analog CO-to-CO calls using DISA or off-net call forward.

Admin Programming

- Unsupervised Conference Timer Extend Enable (PGM 160 - FLEX 18)
- Unsupervised Conference Timer Extend Code (PGM 109 - FLEX 6)
- Unsupervised Conference Timer (PGM 182 - FLEX 6)

Flash

CO Flash - Provides station users with the ability to terminate an outside call or transfer a call without hanging up. A [FLASH] button is located on each DKTU.

The flash type and duration of each CO line are assigned by the system.

Flash on ICM Call - This feature enables Station Users to utilize the [FLASH] button for terminating pages and intercom calls. While paging or on an intercom call, press the [FLASH] button to terminate the call and return to the intercom dial tone.

Operation

To perform a Flash while on a CO line call:

Press the [FLASH] button.

To generate a flash while on a CO line from a SLT:

1. Press the hook-switch slightly.
2. Dial 551.

Conditions

- The Flash command is not activated on ISDN CO lines.
- A Station that is not permitted to access CO line cannot initiate a flash.
- During a flash, the LED of the CO line will flash.

Admin Programming

- Refer to the SBX IP 320 Installation Manual, Flash Command to CO Line Code (PGM 106 - FLEX 8)
- Flash Type (PGM 141 - FLEX 7)
- CO Flash Timer (PGM 142 - FLEX 12)
- SLT Hook-switch Bounce Timer (PGM 182 - FLEX 1)
- SLT Maximum Hook-switch Flash Timer (PGM 182 - FLEX 2)
- SLT Minimum Hook-switch Flash Timer (PGM 182 - FLEX 3)

Flexible Buttons

Flexible buttons are customized by either ADMIN or station programming.

The programmable Flexible buttons include:

- CO Line - Automatically accesses the assigned line (User Programmable).
- DSS/BLF - Automatically indicates the assigned station and provides BLF for off-hook and DND (User Programmable).
- Flexible Numbering Plan Code - Any Feature with a dialing code (Paging, Account code, Call Park, etc.) can be assigned to a flexible button (User Programmable).
- Speed Dial - Automatically dials Speed Number (System, Station, Saved Number Redial, Last number Redial) (User Programmable).
- Group Access - Hunt Group pilot number (User Programmable).
- Pool Group Access - Some or all outside lines can be grouped; pressing this button accesses the highest numbered unused CO line in that group (User Programmable).
- Loop - Used to answer a transferred call on l line for which a user does not have a button assigned (User Programmable).
- Station Assignment - Allows assignment of stations and complete flexibility within the system numbering plan. A station can be assigned a number (100-399).
- 4/8 button - Feature code assign.
- Telephone Number - Automatically dials an outside Telephone Number (ex. 7208D/7224D.)

Operation

The Flexible buttons are programmable individually at each keyset, and are used by pressing the applicable FLEX button.

To assign a flexible button:

1. Press the [TRANS/PGM] button.
2. Press the FLEX button to be assigned.
3. Press the [TRANS/PGM] button.
4. Type the Feature Code (refer to the Flexible Feature Code table and Station Programming codes table at the end of this feature).
5. Press the [HOLD/SAVE] button to save changes.

To assign a direct button (i.e., CO or DSS button):

1. Press the [TRANS/PGM] button.
2. Press the FLEX Button to be assigned.
3. Type the desired Code.
4. Press the [HOLD/SAVE] button.

To assign a Telephone Number (7208D/7224D DKTU):

1. Press the [TRANS/PGM] button.
2. Press the FLEX Button to be assigned.
3. Press the [TEL NUM] soft button.
4. Type the desired telephone number.
5. Press the [HOLD/SAVE] button.

Conditions

- A flexible button that is not assigned as a {CO Line} button is considered a Feature button and is programmable at each Station.
- When ADMIN program 112 - FLEX 6 (CO Line Programming) is enabled, the flexible button that is assigned as a {CO line} button can be programmed.

Admin Programming

- Flexible Button Assignment (PGM 115)
- CO Line Programming (PGM 112 - FLEX 6)

FLEXIBLE NUMBERING PLAN		
Number	Item	Comments
100-147	Intercom Call	-
620-629	Group Pilot Number	-
501-510	Internal Page Zone	-
543	Internal All Call Page	-
544	Meet Me Page	-
545	External Page Zone	-
549	All Call Page (Int & Ext)	-
550	SMDR Account Code Enter	SLT
551	Flash Command to CO Line	SLT
552	Last Number Redial	SLT
553	DND (Toggle On/Off)	SLT
554	Call Forward	SLT
555	Speed Dial Programming	SLT
556	Message Wait/Callback Enable	SLT
557	Message Wait/Callback Return	SLT
558	Speed Dial Access	SLT
559	Cancel DND/FWD/Pre-MSG	SLT
560	SLT Hold	SLT
563	Programming Mode Enter Code	SLT
564	ACD Reroute	-
565	Alarm Reset	-
566	Group Call Pickup	-
568	UCD DND	-
569	Night Answer	-
601-610	Call Parking Locations	-
7	Direct Call Pickup	-
801-824	CO Line Group Access	-
8801-8836	Individual CO Access	-
8*	Retrieve Held CO Line	-
8#xx	Retrieve Held Individual CO Line	-
9 (or 0, based on nation code)	Access CO Line in the 1st available CO Line Group	-
0 (or 9, based on nation code)	Attendant Call	-

FLEXIBLE NUMBERING PLAN		
Number	Item	Comments
#*1	1st Door Open	-
#*2	2nd Door Open	-
#*3	3rd Door Open	-
#*4	4th Door Open	-
*8	VM Message Waiting Enable	-
*9	VM Message Waiting Disable	-

STATION PROGRAMMING		
Number	Item	Comments
11	Differential Ring	Keyset
12	Intercom Answer Mode (1 HF / 2 TONE / 3 PV)	Keyset
13	SMS Message Display	LDP Keyset
14	Enblock Mode	LDP Keyset
15	SMS/ Notice Display	LDP Keyset
16	Scroll Speed	LDP Keyset (Not supported in SBX IP 320)
17	Ear-Mic Headset	LDP Keyset
18	ICM Ring	LDP Keyset
19	CO Ring	LDP Keyset
21	Station COS Down	-
22	Station COS Restore	-
23	Walking COS	Keyset
31	Authorization Code Registration	-
32	Authorization Code Change	-
33	Registration Mobile - Extension	-
34	Active Mobile - Extension	-
35	Register Mobile-Extension CLI	-
36	Voice Msg Wait Notice To Mobile-Extension	-
41	Wake-up Time Registration (One-time/ Continuous)	-
42	Wake-up Time Cancel	-
43	Active Conference Room	-
44	Deactive Conference Room	-

STATION PROGRAMMING		
Number	Item	Comments
451	Call Coverage Mode	-
452	Call Coverage Delay Ring Cycle	-
51	Pre-selected MSG Activation	-
52	Set Custom Message	-
61	Record VMIB User Greeting	-
62	Listen VMIB Time & Date	-
63	Listen VMIB Station Number	-
64	Listen VMIB Station Status	-
65	Record VMIB Page Message	-
66	Erase VMIB User Greeting	-
67	Erase VMIB Page Message	-
71	LCD Display Mode (English/Domestic Language)	Keyset
72	MPB Version Display	Keyset
73	Background Music	Keyset
74	Station User Name Registration	-
75	Headset/Speakerphone Mode	Keyset
76	Headset Ring Mode	Keyset
77	WTU Station Number Receive	Keyset (not supported in SBX IP 320)
78	Serial No/SW Packages	Keyset with LCD
79	PC – Phone Lock Key	-
**	HOTDESK Logout	-
*0	HOTDESK Login	-
*1	Relocation Out	-
*2	Relocation IN	-
*3	Register Bluetooth	Not supported in SBX IP 320
*4	Bluetooth Usage	Not supported in SBX IP 320

Forced Hands-free Mode

A DKTU caller can temporarily change the answering mode of the called party DKTU (tone mode -> hands free mode).

Operation

To activate forced hands free mode, use the following procedure:

1. During a tone mode call to a DKTU, when you hear a ring back tone, dial the forced hands free code number.
2. The called party DKTU will stop ringing and the speaker and microphone will be activated (operate as hands free mode).

Conditions

- If the DKTU is changed to hands-free mode, the connection tone will be provided and the voice path is connected.
- If the called party DKTU is set to linked-pair station, the answer mode of the called party will be not changed.
- If the calling party DKTU is set a linked-pair station, the calling party DKTU can use the forced hands free mode.
- If called party DKT is set as a Mobile-Ext, this feature will be not applied.
- Only the calling party DKT that is designated in PGM 111 - FLEX 19 (with a setting of ON) can use this feature.
- After connection, related feature with hands free mode is not applied.(ex. call back)

Admin Programming

- Forced Hands-Free Mode (PGM 111 - FLEX 19)

Forced Trunk Disconnect

Forced Trunk Disconnect feature allows an authorized user (Emergency Supervisor) to disconnect another extension's active outside call, and then the user can make an outgoing call on the released line.

This feature is used in an emergency when no other line is available.

Note: This feature is used in emergency case, so only an authorized user can use it. Station base program is implemented for this feature.

Operation

To activate the Forced Trunk Disconnect feature:

1. Seize a CO line.
2. You hear a busy tone and then dial the "Forced Trunk Disconnect" code, #. The busy line goes to idle and you hear a dial tone.

Conditions

The Forced Trunk Disconnect Feature is only possible when the station is set to Emergency Supervisor.

The Forced Trunk Disconnect Feature is only possible for an Analog line.

Admin Programming

- Emergency Supervisor (PGM 112 - FLEX 24)

Headset

An industry-standard headset can be connected to a Station instead of using the handset. The Station is programmed for headset operation in place of speakerphone operation.

Operation

To change the Station mode between Speakerphone or Headset:

1. Press the [TRANS/PGM] button.
2. Dial code 75.
3. The Station Answer mode should be displayed on the LCD.
4. Dial code 1 (Speaker mode) or 0 (Headset mode).

To change the Headset Ring mode:

1. Press the [TRANS/PGM] button.
2. Dial code 76.
3. The Station Answer mode should be displayed on the LCD.
4. Dial the applicable code

   1 = Speaker ring only

   2 = Headset only

   3 = Both

Conditions

- The intercom signaling mode (HF/TN/PV - [TRANS/PGM] + 1 + 2) can be set in Headset and Speaker phone mode.
- In Headset mode, the User can select an incoming ring mode to hear ringing with the speaker, headset, or both by Admin programming.
- The station will receive paging with the Station speaker.
- To answer an intercom call in Tone mode, the User should press the [SPEAKER] button.
- When an intercom call is received in Privacy mode, the station will be muted automatically. User should press the [MUTE] button to answer the intercom call.

Admin Programming

- Headset Ring Mode (PGM 111 - FLEX 10)
- Speakerphone/Headset (PGM 111 - FLEX 11)

Hot Desk

A hot desk enables the user to dynamically select a Station using a login / logout operation without having a fixed station. For example, in a call center or marketing department people can share work stations with one another.

Operation

To activate hot desk operation at a dummy station, use the following procedure:

1. Go Off-hook or press the [SPEAKER] button.
2. Dial the authorization code, then press the # key.
3. The Station restores the User's database (station number, COS, ring assign, etc.) and can receive incoming calls.

To logout, use the following procedure:

1. Press the [PGM] button.
2. Dial the User Logout code or press the programmed [Agent Logout] flex button.
3. Select the call forward type (refer to values) using the volume up/down keys.
    - Off-net speed 000
    - Mobile extension
    - VMIB
    - VM group
4. Press the [HOLD] button. The User database is saved and the Station returns to the dummy state.

Conditions

- A dummy station will display "DUMMY STATION (xxx)" (xxx = the physical Station number).
- A dummy station only allows login operation; all other operations are not allowed.
- Logout operation is only allowed for dummy Stations where a user has logged-in.
- Total number of Users is restricted by the System Station capacity. The total number of hot-desk Users = the total number of Stations - Installed number of Stations - 1.
- A hot-desk User must have his own password.

- Saved User database information includes:
    - Station Number
    - Station Attributes (PGM 111 - 124)
    - CO Routing (Ring Assign, DID routing)
    - Hunt Group Membership
    - Voice Mail
- If user tries to log-in at another station without logging-out of a dummy Station, the previous used Station returns to the dummy state automatically.
- The hot-desk will automatically log-off if there is no activity at the DKTU within the Auto Log-Out Timer.
- The button map of the hot-desk will not be changed even though a User logs-on to a different type of DKTU (it is recommended that the same type of DKTU is used for the Hot-Desk Station).
- Only DKTUs (with more than 12 buttons) and WKTs can be used as dummy stations.
- The modem associated with the Station cannot log in (PGM 170).
- When the System reset happens, all login agents are automatically logged-out.

Admin Programming

- Dummy Station ON/OFF (PGM 112 - FLEX 23)
- Number of Agents (PGM 250 - FLEX 1)
- Assign Station Number of Agents (PGM 250 - FLEX 2)
- Agent Auto Logout Timer (PGM 250 - FLEX 3)

In-Room Indication

A supervisor presses In-Room Indication button and [HOLD/SAVE] button at the idle state. Then each LED of In-Room Indication buttons of all members are turned ON.

10 groups can be programmed. Each group has at most 20 members excluding the Supervisor.

To program a flexible button as the "In-Room Indication button":

1. Press the [TRANS/PGM] button and the flexible button to be assigned.
2. Dial the code ([TRANS/PGM] + *8)
3. Press the [HOLD/SAVE] button.

Operation

To Activate or Deactivate the In-Room Indication button:

1. Check if Supervisor station is the idle state and In-Room Indication button is programmed.
2. Press the In-Room Indication button and press the [HOLD/SAVE] button.

Conditions

- If the Supervisor station is not Idle, the In-Room Indication button does not work.
- If a station presses the In-Room Indication button but the station is not a supervisor, an error tone is heard.
- If the [HOLD/SAVE] button is not pressed within 5 seconds after the In-Room Indication button is pressed, the station goes back to an Idle state.

Admin Programming

- In-Room Indication Supervisor (PGM 183 - FLEX 1)
- In-Room Indication Member (PGM 183 - FLEX 2)

Intercom Signal Mode

Users can control the method by which they receive intercom calls and signals.

Stations equipped with a speakerphone can select one of the available three signaling modes:

HF - Hands Free. The station user, upon hearing a tone burst and voice announcement over the speaker, can reply hands free.

TN - Tone. A standard tone ring notifies the party of an incoming intercom call. The called party answers by lifting the handset or pressing the [SPEAKER] button.

PV - Privacy. The station user receives a tone burst and a voice announcement over the speaker. The microphone is deactivated for privacy. The called party must lift the handset or press the [MUTE] button to answer the call.

Operation

To assign the Intercom Signal mode, use the following procedure:

1. Press the [TRANS/PGM] button.
2. Dial 12; the confirmation tone sounds.
3. Dial the signal mode type (1 = HF, 2 =TN, 3 = PV).
4. Press the [HOLD/SAVE] button and confirmation tone is heard.

Conditions

- In Message Wait/ICM Queuing/Call Transfer/ Attendant Override, the ring is received with TN mode regardless of the assigned Intercom Signal mode.
- Intercom signal mode will not affect the voice announcements from internal/all call paging.

Intercom Tenancy Group

A station can be assigned to one of the Intercom Tenancy Groups. Each Intercom Tenancy Group can be operated independently and Stations in the group can be assigned an individual CO Group to use. A maximum of five Intercom Tenancy Groups can be assigned.

Each group can be assigned by the Attendant and can be programmed to allow or deny calls to other groups. Stations in a Group are allowed access to other Stations based on the allowed access of the calling Group.

When a call to another intercom Tenancy Group is allowed, CO line or Station calls, pick-up and transfer features are activated.

Conditions

- When a call to another Intercom Tenancy Group is denied, call pick-up cannot be activated.
- It is not allowed for stations to have the same station numbers regardless of whether they belong to different intercom tenancy groups.
- The attendant of an Intercom Tenancy Group can be any station in the system, and it is not affected by Intercom Tenancy Group access.
- When the Attendant of an Intercom Tenancy Group sets the Day/Night/Weekend mode, it will affect only their assigned Intercom Tenancy Group.

Admin Programming

- Intercom Group Number (PGM 111 - FLEX 13)
- Intercom Tenancy Group (PGM 120)

Message Wait / Call Back

Message Wait - A station user can notify another Station User that he wishes to talk to them. The notified station user can return the call or a message left at the station. When responding to the Station, the User can answer messages left on the Station in sequential order (up to 5 messages).

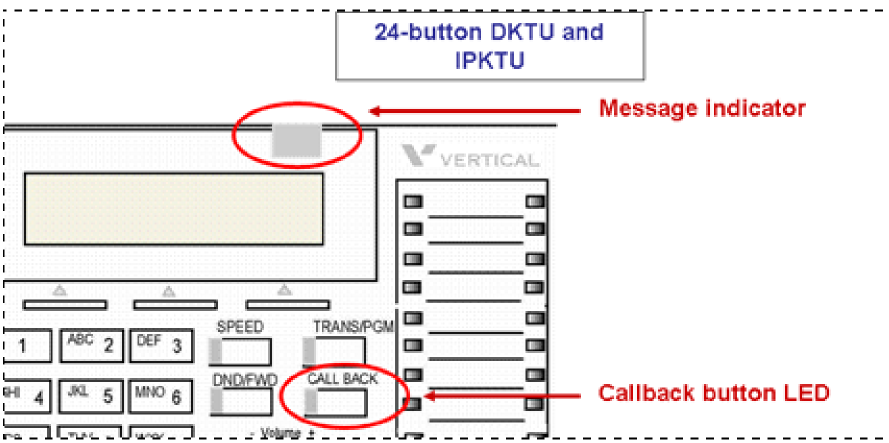
Call Back - A station user can initiate a call back request on a busy station. Once that station becomes idle, the station that left the call back request will be signaled.

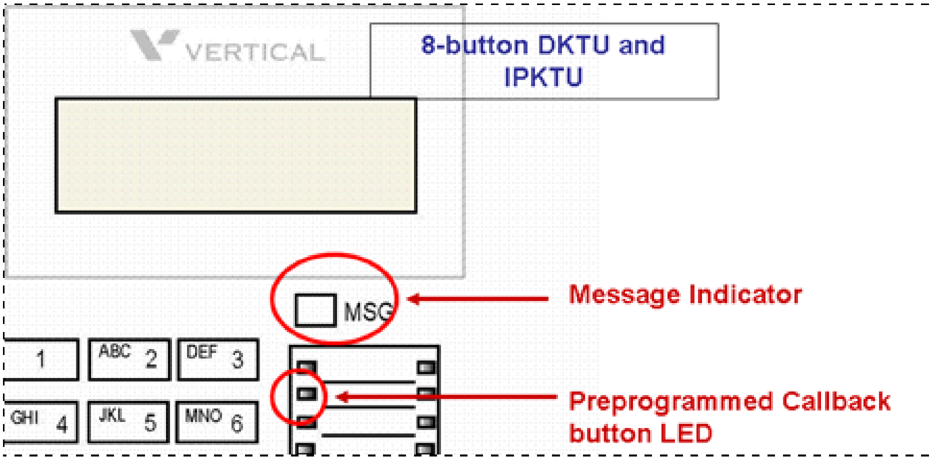
A station with a message waiting can receive periodic audible reminders of a message waiting. This tone is sent to Stations only while idle and is presented over the speaker.

Message Wait Indicator Lamps

Subscribers must be able to clearly distinguish whether new, unheard messages, or callback requests, are waiting.

The following diagrams will be used to specify the location of the Message Indicator and Callback button LEDs:





- A new message for the VMIB or external voice mail is defined as a message in the mailbox which has not yet been played to the subscriber.
- A saved message is defined as a message in the mailbox which has been played to the subscriber, but has not been deleted or moved.
- A station message wait request is defined as the MWI request left by another intercom station without a voice message, by pressing the Call Back button when ringing a busy or unanswered station.

8-button DKTU and IPKTU

- Ringing state: An incoming call flashes the Message Indicator lamp at the rate of 1 flash per 500 ms until the call is answered, forwarded, or abandoned. This occurs regardless of whether the mailbox has a message, or whether a callback request is active on the station.
- Idle State with new messages: When a new message is in the VMIB or external VM, the Message Indicator lamp, and the Call Back button (if programmed) are flashing at the rate of 1 flash per 1000 ms.
- Idle State with message wait request and no new voice message: When a station message wait request is active on the station and there is no new voice message in the VMIB or external VM, the Message Indicator lamp is extinguished, and the Call Back button (if programmed) is flashing at the rate of 1 flash per 1000 ms.

- Idle State with no callback request and no new message: When no station message wait request is active on the station and no new message is in the VMIB or external VM, the Message Indicator lamp, and the Call Back button (if programmed) are extinguished.

24-button DKTU and IPKTU-

- Ringing state: An incoming call flashes the Message Indicator lamp at the rate of 1 flash per 500 ms until the call is answered, forwarded, or abandoned. This occurs regardless of whether the mailbox has a message, or whether a callback request is active on the station.
- Idle State with new messages: When a new message is in the VMIB or external VM, the Message Indicator lamp, and the Call Back button are flashing at the rate of 1 flash per 1000 ms.
- Idle State with message wait request and no new voice message: When a station message wait request is active on the station and there is no new voice message in the VMIB or external VM, the Message Indicator lamp is extinguished, and the Call Back button is flashing at the rate of 1 flash per 1000 ms.
- Idle State with no callback request and no new message: When no station message wait request is active on the station and no new message is in the VMIB or external VM, the Message Indicator lamp, and the Call Back button are extinguished.

Mailbox buttons:

- In the idle state, on any DKTU or IP-KTU, including Soft Phone, all pre-programmed flex buttons assigned as mailbox buttons will flash red at the rate of 1 flash per 1000 ms when there is an un-played (new) message in the assigned mailbox.
- When there is no new, unheard message in the assigned mailbox, the mailbox button assigned will be extinguished.

Operation

To leave a message wait at an idle station that does not answer or is in DND mode, use the following procedure:

Press the [CALLBK] button; a confirmation tone sounds. The [CALLBK] button LED at the receiving station will flash.

To answer a message wait, use the following procedure:

Press the flashing [CALLBK] button. The station that left the message will receive an intercom ring.

To leave a call back at a busy station, use the following procedure:

1. Press the [CALLBK] button; a confirmation tone should be heard.
2. Replace the handset.
3. When the receiving station hangs up the current call, the station leaving the message will receive an intercom ring.
4. When the intercom call is answered, the callback request will be removed.

To leave a message wait at an SLT:

1. Hook-flash.
2. Dial 556; a confirmation tone sounds
3. Replace the handset.

To answer the message wait:

1. Lift the handset; an indication tone will sound.
2. Dial 557. The station leaving the message will receive an intercom ring.

Note: If a voice message is recorded on a pulse-type SLT (PGM 110), the recorded message will be played by dialing the message wait retrieval code (557). The played message will be deleted automatically after being played. When more than one message is recorded, after the first message, a warning tone will be heard to indicate remaining messages. To retrieve another message, dial 557 again. The callback will ring with Tone mode regardless of intercom signaling mode.

To queue a call back on a busy SLT:

1. Hook-flash.
2. Dial 556 while busy tone is heard.
3. Confirmation tone sounds; replace the handset.
4. When the SLT returns to an idle state, the intercom ring is received.

Conditions

- A Station can leave only one callback or message; a new request will override the previous one.
- Message wait data will be protected with power failure.
- When dialing the station number instead of pressing the [CALLBK] button to answer a message wait, the message wait will be canceled in the calling station.
- Message wait reminder tone is programmable from 00 to 60 min. If you don't want to present the tone, the timer may be set to 00.
- Message wait reminder tone is not heard at a busy station.
- Message wait reminder tone will continue until all the messages are retrieved.
- When a station attempts to leave a message at a station which has already 5 messages and one of those is not equal to the attempting station, error tone will be heard. When VMIB access is allowed in the station, after recording VMIB message, it turns to a normal message in the station. In this case, an error tone will not be provided in the attempting station.

Message Wait Indication (MWI) - SLT Feature

If the SLT station receives a message from another User, then the SLT LED will flash indicating a message is waiting.

Conditions

- When lifting handset, user will hear a DND warning tone indicating message waiting.
- When a message waiting indication prompt is recorded in the System greeting 097, a voice announcement will be heard instead of the DND warning tone.

Admin Programming

- Station ID Assignment (PGM 110 - FLEX 1)

Mobile Extension

A mobile user is able to use the phone as an extension of the SBX IP 320 system to receive incoming and make outgoing calls when the mobile phone number is registered to the SBX IP 320 System. Mobile Extension is only for SIP trunks.

Operation

To register a mobile extension number to a DKTU.

1. Press the [TRANS/PGM] button.
2. Dial 33.
3. Dial the mobile phone number that is being registered.
4. Press the [HOLD/SAVE] button.

To activate (deactivate) a mobile extension, use the following procedure:

1. Press the [TRANS/PGM] button.
2. Dial 34.
3. Dial 1 to activate the extension, or 0 to deactivate it.
4. Press the [HOLD/SAVE] button.

To register the mobile extension CLI number to a DKT:

1. Press the [TRANS/PGM] + 35.
2. Dial the mobile CLI number.
3. Press the [HOLD] button.

To use the mobile extension in case of Hunt group member:

1. Press the [TRANS/PGM] + 36.
2. Dial "1" to activate, or "0" to deactivate.
3. Press the [HOLD] button.

Incoming call through DID, ICM or Transferring call:

1. The call to the extension which has the preprogrammed mobile extension is routed to both an extension and a mobile extension.
2. When an incoming call is answered at an extension or at a mobile extension, the other will stop ringing.
3. If mobile is answered, the extension goes to Mobile-Ext In Used State [IN USE AT MOBILE EXT].

4. Under the Mobile-Ext In Used State, the extension cannot receive any digits.
5. If another extension tries to call the extension of Mobile-Ext In Used State, a busy tone sounds. Call Waiting is not supported in this case, but the user can leave a message wait to the extension of Mobile-Ext In Used State.

Transferring a call from the mobile extension through the SBX IP 320 system:

1. Dial the transfer code (*) during the talk state in Mobile extension. The mobile extension user hears the internal dial tone and the calling party hears the MOH.
2. Dial the extension number.
3. The call will be transferred after the mobile extension user goes on-hook state.
4. The mobile extension user can reconnect the call by pressing the code (#) in the case that the extension does not answer.
5. Mobile Extension can transfer a call to another station's mobile extension.

To make outgoing calls from the mobile extension through the SBX IP 320 system:

1. The mobile extension user dials his DID number of the SBX IP 320 system. The internal dial tone will sound.
2. The mobile extension can proceed to make an internal or outgoing call with CLI number of extension.

Conditions

- If the extension is busy, forwarded, or in the DND state, calls will not be routed to the mobile extension.
- When making an external call from a mobile extension through the SBX IP 320 system, it sends the CLI number of the Station extension; the mobile extension cannot use the transfer feature.
- After the no answer timer expires, a call will be routed to the DID no answer destination (the no answer timer of the SBX IP 320 system is shorter than GSM).
- This feature applies only to ISDN DID Lines, ICM calls, and transferred calls.
- The message wait or call back feature is not supported on mobile extensions.
- When on a mobile extension conversation through the SBX IP 320, the DTMF receiver is dedicated to the mobile extension.
- A mobile extension can transfer the call to another extension. If he tries to seize a CO line or to dial the Hunt Group or dial another destination except an extension, an error tone will be provided.

- If the "Transfer recall" timer expires, a call is not routed to the extension associated with mobile extension.
- An outgoing call from a mobile extension is restricted by the internal station's COS.
- A mobile extension can receive the Calling Party number or his own extension CLI number as CLI (according to PGM 143-FLEX7-CLI Transit: ORI / CFW)
- To use the "Transferring call / outgoing call from mobile extension" feature, the CO line connected with the mobile extension should be set to a DID line.
- When a mobile extension calls his own DID number of the SBX IP 320 system to make a CO call, the SBX IP 320 system checks the CLI of the mobile extension number. If the CLI matches the "Mobile Extension Number" or "Mobile CLI" which is registered to his extension, the system provides CO dial tone and allows him to make a CO outgoing call through the SBX IP 320 system.
- To assign the {Mobile Extension Activation} button:

  [TRANS/PGM] + {FLEX} + [TRANS/PGM] + 34 + [HOLD/SAVE]
- If the extension is a member of a Hunt Group (except Ring Hunt Group), a call is routed to the mobile extension.
- To use the mobile extension in case of Hunt group member, Hunt call should be programmed in PGM 236, FLEX 5.
- A mobile extension can transfer a call to another station's mobile extension.

Admin Programming

- Mobile Extension Register (PGM 236)

Music On Hold (MOH)

When a CO call is placed on hold, the external party will hear music.

MOH is supplied through various music sources. Music can be played to any party on hold. The following music sources are available:

- Internal Music
- External Music
- SLT MOH

Operation

The following values are associated with MOH:

0 = Not assigned

1 = Internal Music

2 = External Music

3 = Reserved

9 = Hold Tone

Conditions

- Only 1 MOH channel is supported.
- SLT ports connected with MOHU can provide MOH channels.

Admin Programming

- CO Line MOH (PGN 142 - FLEX 6)
- MOH Type (PGM 171 - FLEX 2)
- Internal MOH Type (PGM 171 - FLEX 8)

Mute

During a conversation, pressing the [MUTE] button disables the handset microphone or the speakerphone for privacy while continuing to listen to the other party on the phone through the handset or speaker. Pressing the [MUTE] button again, will reactivate the microphone.

Operation

To mute the transmitting audio, while on a call:

Press the [MUTE] button. The [MUTE] button LED will illuminate (the connected party will not be able to hear the voice on the muted Station).

To restore transmission:

Press the illuminated [MUTE] button. The LED of the [MUTE] button extinguishes and transmission is restored.

Conditions

- When changing from the speakerphone to the handset, mute is released.
- When pressing another DSS button, the mute state will not be changed.

On-Hook Dialing

A station can make a call without lifting the handset by using the speakerphone or monitor mode.

If this feature does not operate, verify if the speakerphone is enabled or disabled. On-hook dialing is not available on all keysets.

Operation

To use On-hook dialing, use the following procedure:

1. Verify that the Auto Speaker Selection is ON.
2. Assign a CO line/Intercom/Speed Dial to a flexible button.
3. Press the flexible button.

-or-

1. Verify the Auto Speaker Selection OFF.
2. Assign a CO line/Intercom/Speed Dial to a flexible button.
3. Press the flexible button. LIFT HANDSET will display on the LCD.
4. Lift handset or press the [SPEAKER] button to operate.

Admin Programming

- Auto Speaker Selection (PGM 111 - FLEX 1)

Remote Mobile Extension Control

A Mobile Extension user can change their mobile extension settings remotely. This feature can only be used when SIP trunks have been configured.

Operation

1. Dial mobile user's DID number of the SBX IP 320 system, an internal dial tone is received.
2. Enter "the remote control menu" with the code (remote MEX control code) and a confirmation tone will be provided.

4. Select the desired menu (1-6), a dial tone is received.

Remote Control Menu

1: Activate Mobile extension

2: Deactivate Mobile extension

3: Fwd to DVU

4: Cancel Fwd to DVU

5: Activate UCD DND (Agent login)

6: Deactivate UCD DND (Agent log out)

Admin Programming

- Flexible Numbering Table C (PGM 109 - Flex 7: Remote MEX Control)

Station Call Coverage

This feature permits a user (a covering station user) to receive ring and answer calls directed from a covered station. A Flex button is assigned at the covering station for calls to the covered station. A station can have multiple Call Coverage buttons, each covering a different station. Multiple stations can have a Call Coverage button for a covered station.

When a covered station rings, the {CALL COVERAGE} button LED will flash and the covering station may receive ring (immediate or delayed) for the call. The covering station can answer the call using the {CALL COVERAGE} button, terminating ring at other stations. Once answered, the LED of {CALL COVERAGE} buttons for the station at other covering stations will extinguish.

Operation

To assign Call-Coverage button toa flexible button (program by call coverage station):

1. Press the [TRANS/PGM] button.
2. Press a flexible button to program the Call Coverage button.
3. Press [TRANS/PGM] button and dial 46.
4. Enter Station number to cover.
5. Press the [HOLD/SAVE] button.

To set Call-Coverage Attributes (station program by call coverage originated station):

1. Press the [TRANS/PGM] button.
2. Dial 45.
3. Dial 1 to program call coverage mode.
4. Enter 1 to activate call-coverage (0 to disable).
5. Press [HOLD/SAVE].

To set Call-Coverage Delay Ring Cycle (station program by call coverage originated station):

1. Press the [TRANS/PGM] button.
2. Dial 45.
3. Dial 2 to enter call coverage delay ring cycle.
4. Dial delay ring cycle (0-9).
5. Press [HOLD/SAVE].

Once the Button is assigned on the station and a call arrives at the covered station:

- The coverage station receives audible and visual indications after a programmable delay when there is a ring on the originating station.
- The ring tone is the normal internal or external ring tone cadence and the oldest call coverage button flashes faster than the others and the others flash the same as the incoming ring rate.
- The coverage station user presses the flashing Coverage Flex Button on the station, or presses the ON/OFF Button, or lifts the handset. The call will be answered and will cease to ring and LEDs cease to flash at any other stations that may have the same coverage appearance.

Conditions

- Multiple coverage stations can have the same remote ringing station(s) programmed on their keysets.
- Once a coverage station answers the call, all the related rings are stopped and the call is picked-up by the coverage station.
- The station user assigns the call coverage button at their station via the Flex Button programming sequence.
- This type of button can be set up on DSS consoles as well as keysets.
- A station user can cover an SLT extension. An SLT terminal cannot perform the Call Coverage function.
- If the coverage station is in DND, no audible ring will be heard.
- Call coverage does not support hunt group members.

Station Name

Each station may be assigned a name of up to 7 characters, and a System and Station speed dial number of up to 12 characters.

The System will allow Station Users to dial Station numbers by entering a name that has been programmed for the Station (via intercom). When the names are programmed in the digital display keyset, the user may select a Station or Speed dial number by the name.

Operation

To register a Station Name:

1. Press the [TRANS/PGM] button.
2. Type 74.
3. Type the Name (up to 12 characters).
4. Press [HOLD/SAVE] to accept changes.

. – 13 Q – 11 Z – 12 1 – 10	A – 21 B – 22 C – 23 2 – 20	D – 31 E – 32 F – 33 3 – 30
G – 41 H – 42 I – 43 4 – 40	J – 51 K – 52 L – 53 5 – 50	M – 61 N – 62 O – 63 6 – 60
P – 71 Q – 72 R – 73 S – 74 7 – 70	T – 81 U – 82 V – 83 8 – 80	W– 91 X – 92 Y – 93 Z – 94 9 – 90
Blank – ∗1 : – ∗2 , – ∗3	0 – 00	

To access Dial by Name:

1. Press the [SPEED] button twice.
2. Dial the desired directory (1-3, refer to values).

   1 = Intercom

   2 = Station Speed Dial

   3 = System Speed Dial

A confirmation tone should be heard. The stored names in the speed bin are displayed in alphabetical order.

Note: The up and down arrows can be used to locate the desired name.

To search a name by entering a character, use the following procedure:

1. While two names are displayed on the LCD, enter alphanumeric data.
2. The LCD displays 2 names which start with the entered character (the cursor will point to the first name in the LCD).
3. When entering more alphanumeric data, the LCD will display names that start with the updated input.

Note: To delete the last letter of input, press the [CALLBK] button. The up and down arrows also can be used to locate the desired name.

4. When the appropriate name is displayed, move the cursor to point at the name.
5. Press the [HOLD/SAVE] button to make a call.

To search for a name by scrolling:

1. While two names are displayed in the LCD window.
2. Use the up and down arrow keys to locate the desired name.
3. When the appropriate name is displayed, move the cursor so it points directly at the name.
4. Press the [HOLD/SAVE] button to make a call.

To register the name of an SLT:

1. Lift the handset.
2. Dial 563; a confirmation tone sounds.
3. Dial 74 (Name Register code).

4. Enter the name (up to 12 letters).
5. Hook-flash to save.

Conditions

- Dial by name is only available on a DKTU with LCD.
- The name must be registered to use Dial By Name.
- If an invalid Group is dialed, an error tone will be heard.
- In SLT and DKTU with no LCD, only the station name can be entered.
- The speed dial without name will not be listed by Dial By Name.

Station Port Blocking

The station port can be blocked by admin program. No calls can be made from/to the blocked station port.

Conditions

- Attendants cannot be blocked by admin programming.
- An ADMIN programming-enabled station cannot be blocked.
- A port blocked station cannot be set as part of a linked pair.
- If Station A (110) and B (111) are a linked pair, and these stations are set to port block, the two stations are port blocked at the same time, but if the linked pair relationship is terminated, these two stations are still port blocked at each station.

Admin Programming

- Station Port Block (PGM 113 - FLEX 15)

Station User Programming

The system supports multiple hierarchical menus based on station programming. Users can program Stations by selecting the desired menu (refer to table). The Attendant also can program a station and the Attendant Station the same way.

MAIN MENU	SUB MENU	OPTIONS	REMARK
[1] RING	[1] TYPE	1, 2, 3, 4	DKTU
	[2] ANSWER MODE	H(1)/T(2)/P(3)	
	[3] SMS MSG DISPLAY		
	[4] ENBLOCK MODE	1: ON/ 0:OFF	
[2] COS	[1] COS DOWN	ICM/COS7	
	[2] COS RESTORE	ENTER AUTHO CODE	
	[3] WALKING COS	ENTER AUTHO CODE	
	[4] COS CHANGE		India Only
[3] AUTHORIZE / MOBILE - EXT	[1] AUTH. REGISTER		
	[2] AUTH. CHANGE		
	[3] REG MOBILE-EXT		
	[4] ACTIVE MOBILE-EXT		
[4] TIME	[1] SET WAKE UP TIME	ONCE/PERMANENT	
	[2] WAKE UP DISABLE		
[5] MESSAGE	[1] SET PRESELECTED MSG	00-10	
	[2] SET CUSTOM MSG	None	
[6] ANNOUNCE	[1] RECORD USER GREETING		
	[2] LISTEN TIME / DATE		
	[3] LISTEN STA NUMBER		
	[4] LISTEN STA STATUS		
	[5] RECORD PAGE MSG		
	[6] ERASE USER GREETING		
	[7] ERASE PAGE MSG		
	[8]OUTBOUND NOTIFY	dial 1, ON;dial 0, OFF	
	[9]OUTBOUND NUMBER	dial outbound telephone #	
	[*] REC USER NAME		
	[#] ERASE USER NAME		

MAIN MENU	SUB MENU	OPTIONS	REMARK
[7] SUPPLEMENTARY	[1] LCD DISPLAY LANGUAGE	DOMESTIC/ENGLISH	DKTU
	[2] MBU VERSION DISPLAY		
	[3] BGM		
	[4] REGISTER STA NAME		2/8 FLEX/SLT
	[5] SPK/HEADSET	SPEAKER/HEADSET	
	[6] HEADSET RING MODE	SPEAKER/HEADSET/BOTH	
	[7] WTU STA NUM RCVR		
	[8] SERIAL NUMBER		
	[9] PC-PHONE LOCK KEY		
[0] ATTENDANT			ATD ONLY
[*] SYSTEM	[#] ENTER ADMIN		ADMIN ONLY
	[1] RELOCATION OUT		
	[2] RELOCATION IN		
	[3] REGISTER BLUETOOTH	(Not supported)	
	[4] BLUETOOTH USAGE	(Not Supported)	
	[0] HOTDESK LOGIN		
	[*] HOTDESK LOGOUT		

Operation

To enter the programming mode:

1. Press the [TRANS/PGM] button.
2. The Main Menu should display.
3. Use the Up and Down arrow keys to view other choices in the Main Menu, Sub-Menu and the corresponding options.

To select a menu, use the following procedure:

1. Dial the number of the desired menu item.
2. If the selected menu is a programming item.
3. If there is an available sub-menu, a selectable menu is displayed on the LCD.

Note: Press the [TRANS/PGM] button to move the top menu. Press the [REDIAL] button to move the previous menu.

Conditions

- After a menu is programmed, the previous menu list is displayed on the LCD.
- Pressing a flexible button in the Main Menu mode, will activate the flexible button programming mode.

Admin Programming

- Refer to the SBX IP 320 Programming Guide, Station Programming (Chapter 3, Quick Reference Admin Programming Tables)

Station Relocation

The Station Relocation Feature lets a user unplug their station and plug it into another location. Dialing a code followed by the old station number brings all the station attributes including station number, button mapping, speed dial, and class of service to the new location.

Operation

To store the station attributes to a temporary buffer, use the following procedure:

1. Dial the feature code [TRANS] * 1 (Station Relocation Backup).
2. Unplug the station.

To retrieve stored station attributes, use the following procedure:

1. Plug the phone in at another properly wired jack.
2. Dial [TRANS] * 2 (Station Relocation Retrieve). The station will be relocated; all Station attributes are copied to the current Station location.

Conditions

- All information for the port of the destination Station will be retained so that it may be copied or relocated to another port.
- If a different Station type is plugged in at a location, preprogrammed {DSS} buttons are not guaranteed.
- DKTUs must be relocated to another digital port. DKTUs cannot be relocated to an SLT port.

Station Serial Call

Using DSS flexible buttons, users can place consecutive Intercom calls without returning the line to an idle state (no need to hang-up) between calls.

Operation

To use serial calling:

Press the appropriate DSS flexible button; the old call will be disconnected and a new call will be established.

Time & Date Setup (Digital Network)

In an outgoing call, the ISDN network will send the appropriate time & date in the CONNECT message when the called party answers.

Operation

To set the time & date using the Digital Network:

1. Press the [TRANS/PGM] button.
2. Dial 046 and 1 (Yes).
3. Press the [HOLD/SAVE] button.

Conditions

- This feature may be set at the Attendant Station and using Admin Programming.
- The time & date can be set to a default value or changed based on seasonal time changes.

Voice Over

This feature provides voice announcements at a busy station without interrupting the existing conversation. The announcement is received over the existing conversation so that only the busy station hears both incoming parties. The user can alternately talk back to both parties.

Operation

To use Voice Over:

1. A busy station is called and camped-on by a new caller. The busy station hears a warning tone over the current voice path indicating the camped-on call.
2. The busy station is connected with both the current caller and a new caller (the busy station can send and receive voice to both simultaneously; the new caller and a current caller cannot send and receive voice to each other.
3. Press the flashing [HOLD/SAVE] button of the busy station; the current caller will hear MOH if available, and the busy station will be connected with the new caller.
4. Alternate between calls by pressing the [HOLD/SAVE] button.

Conditions

- Placing a station in DND disables the voice over function.
- The [HOLD/SAVE] LED flashes at 60 ipm at the busy station during the Voice Over.
- An Attendant can activate voice over at the station in DND mode.
- After voice over is activated, both calls will be dropped if the busy Station that receives the calls hangs up. If either of the callers hangs up, the [HOLD/SAVE] LED extinguishes and the remaining call will be connected in the normal talking state following the presentation of a confirmation tone.
- The holding party will receive MOH if provided.
- The recall timer is not activated during voice over.
- Every time the busy Station switches between the callers, a confirmation tone is provided.
- If the busy Station is using the handset, voice over will be activated via the handset. Likewise, if busy Station is using the speaker, voice over will be activated using the speaker.
- Voice over is operated when busy station has an allowance to receive voice -over (PGM 113 - FLEX 6) and a new caller has an privilege to make voice-over (PGM 111-FLEX 22).

Admin Programming

- Voice Over (PGM 113 - FLEX 6)
- Caller Voice Over (PGM 111 - FLEX 22)

Wakeup

Each attendant or station user can set an alarm as a wake-up call or reminder. This feature can be programmed to activate one time or programmed to repeat daily. If the user goes off-hook during the alarm, a special dial tone will be presented.

Operation

To register a wake-up time alarm from the Attendant Station:

1. Press the [TRANS/PGM] button.
2. Dial 042.
3. Dial the station range to be alerted. If a single station is to receive the alarm, enter * instead of a second station number.
4. Dial the desired time the alert should display (2-digit hour and minute, 24-hour mode.
5. Dial # to have the alarm alert once only.
6. Press the [HOLD/SAVE] button.

To cancel a wake-up alarm from the Attendant Station:

1. Press the [TRANS/PGM] button.
2. Dial 043.
3. Dial the station range that was to be alerted, if a single station is to receive alarm, enter * instead of a second station.
4. Press the [HOLD/SAVE] button.

To register wake-up time alarm from a Station:

1. Press the [TRANS/PGM] button.
2. Dial 41.
3. Dial the desired 2-digit hour and minute for alerting.
4. Dial # to have the alarm alert once only.
5. Press the [HOLD/SAVE] button.

To cancel a wake-up time from a Station:

1. Press the [TRANS/PGM] button.
2. Dial 42.
3. Press the [HOLD/SAVE] button.

Conditions

- If a wake-up alarm is registered at a station, a * is displayed in front of the present time on the LCD.
- If a VMIB is installed, the voice message for the wake-up time is heard 3 times and then MOH is presented.
- If the wake-up ring is not answered after 3 times, it is canceled.
- If the System Attendant dials to wake-up fail station to erase wake-up fail ring, the fail ring will disappear with confirmation tone. But when the Wake-up Fail Ring timer expires, confirmation tone will not be provided and Fail Ring will disappear.
- If the [Wake-up Fail Ring Timer] is 0, wake-up fail ring will not disappear automatically.
- If the [Wake-up Fail Ring Timer] is 99, the fail ring is not given to attendant station.
- If the [Wake-up Fail Ring Timer] is 1-98, after Wake-up Fail Ring Timer expires, the wake-up fail ring will disappear automatically.

Admin Programming

- Wake-up Fail Ring Timer (PGM 182 - FLEX 7)

Conference

A station user (Supervisor) can make a call with intercom stations and CO Lines. A Supervisor invites station users or CO Line users one by one with the [CONF] button. Connected users can speak and hear with each other at the same time.

In a Multi-Line conference, up to 15 parities (intercom/CO Line) can enter a conference. Up to 14 CO lines can be used to converse with one intercom station.

An Unsupervised Conference is one that continues even though the conference initiator (supervisor) exits the conference.

To establish an Add-on Conference:

1. While on a call, press the [CONF] button; the existing call will be put on HOLD and the intercom dial tone will be heard.
2. Dial the digits to connect the second internal party.
3. When the call is answered, press the [CONF] button.
4. Repeat steps 1-3, until all parties are added to the conference.
5. When all parties have been called, press the [CONF] button again; all parties will be connected to the call.

To make a Multi-line Conference:

1. While on a CO line call, press the [CONF] button; the existing call will be put on HOLD and the intercom dial tone will be heard.
2. Dial the digits to connect the second party.
3. When the call is answered, press the [CONF] button.
4. When all parties have been called, press the [CONF] button again; all parties will be connected to the call.

To make Unsupervised Conference:

1. During a conference, the supervisor presses the [CONF] button.
2. The conference is still connected and the supervisor’s [CONF] button LED should flash.
3. To re-enter the conference, the supervisor lifts the handset and presses the flashing [CONF] button.

Conditions

- In an unsupervised conference, it is restricted to Unsupervised Conference timer if there is no internal station in the conference (Default: 10min.).
- The Unsupervised Conference timer will be reset if the internal party re-enters the conference.
- Up to 15 parties (internal/external) can enter a conference.
- In a Multi-line conference, up to 14 CO lines can be on a conference with one internal party.
- If the supervisor in a conference receives an error or busy tone from the internal party while making a conference, he can receive intercom dial tone again by pressing the [CONF] button.

Admin Programming

- Unsupervised Conference Timer (PGM 182 - FLEX 6)
- Multi-line Conference (PGM 160 - FLEX 9)

Conference - SLT (Brokers Call)

A single line telephone user can initiate a 3-way conference with any combination of CO line or internal users.

A single line telephone user can alternate between two calls, maintaining private conversations with both parties. The parties may be either internal (stations connected to the system) or external CO line calls, and may be incoming or outgoing.

Operation

To set up a conference from an SLT:

1. Make the first call.
2. Hook switch and the intercom dial tone will sound; the existing call is placed on exclusive hold and the recall timer is activated.
3. Place the second call and announce the conference.
4. Hook switch and connect to the first call; within 2 seconds, hook switch again to establish a conference.

Conference Room

This feature allows internal users or CO callers to join a conference without being invited by the conference supervisor. This conference feature has conference join codes, and each conference room has its own join code (room number). A DID/DISA and transferred CO call can be a member of the conference. This feature terminates when the deactivation code is dialed or Forced Delete PGM code is used by the Attendant.

When a user enters a conference room, members that have previously entered receive a warning tone. This warning tone is provided to all members, including CO Conference Room Members. When a user goes out of the conference room, a warning tone is also provided.

If this tone is not desired, PGM 160 - FLEX 11 [CONF WARN TONE] option can be set to OFF.

Each Conference Room can be assigned to a DSS button. This Conference Room button flashes or is steady depending on the number of conference room members. This LED flashing is enabled all stations, as well as at Attendant Stations.

Operation

To Activate the conference room feature:

1. Press the [TRANS/PGM] button.
2. Dial 4 + 3.
3. Dial the Conference Room Number (1-9).
4. Dial the 5-digit password (optional).
5. Press the [HOLD/SAVE] button to save changes.

To deactivate a conference room:

1. Press the [TRANS/PGM] button.
2. Dial 4 + 4.
3. Dial Conference Room Number (1-9).
4. Press the [HOLD/SAVE] button.

To join a conference room in case of internal call:

1. Dial the activated conference room number (571-579).
2. Enter the 5-digit password for entering conference room (optional).

To join a conference room in the case of a DID/DISA call (in using of DID type 2, DID destination is assigned conference room):

DID/DISA call is routed to conference room.

To Transfer CO Call to conference room:

1. Press the [TRANS/PGM] button at the Attendant Station.
2. Dial the activated Conference Room Number (571-579).
3. Dial the password for entering the conference room (optional).

To check Conference Room status from the Attendant Station:

1. At the Attendant Station, dial a conference room status code [TRANS/PGM] + dial 0 + 4 + 7.
2. Enter conference room number (1-9) to monitor.
3. The Attendant LCD will show the number of members joining the conference room.

To delete room forcedly by Attendant:

1. Dial a conference room forced delete code, [TRANS/PGM] + 048.
2. Enter the Room number and press [HOLD/SAVE].
3. When erasing Conference room at ATD station, password is not needed.

Conditions

- The Maximum number of conference rooms is 9.
- A maximum of 15 members can enter each room.
- Assigning and entering a password is Optional.
- CONF ROOM status can be checked by ATD (how many members are joining the conference room).
- For CO party, only ISDN line can enter a conference room.
- If system attendant has a conference room button, she can check the status of conference room by checking the LED indication.

  - ON - The conference room is activated, but there is no members in the room.

  - OFF - The conference room is deactivated.

  - Flash 60 IPM - The number of members are 1 to 3.

  - Flash 120 IPM - The number of members are 4 to 6.

  - Flash 240 IPM - The number of members are more than 7.
- In case of Analog Line which is set to a valid "Open Loop timer" (PGM 142 - FLEX 13), a DISA and transferred CO call can be a member of conference.
- PGM 160 - FLEX 11 option is used as a Conference feature. Entering and Leaving tones are controlled by PGM 160 - FLEX 11.

Admin Programming

- Conference Warning Tone (PGM 160 - FLEX 11)

Paging Conference

During a page by conference page zone, the second originator can page along with the first originator.

Operation

When a Conference Page is being activated:

Keyset User

Lift the handset and press the [CONF] button.

SLT User

1. Lift the handset to answer the page.
2. Hook-flash and dial 58 (conference page join code).

Conditions

- The Page Timer is not applied to Paging Conference Group.
- If there is a second page originator, it is impossible to "Meet Me Page."
- If the first originator goes on-hook, the conference group paging connection is released.
- The second originator can make paging regardless of page access privilege.

Admin Programming

- Paging Warning Tone (PGM 161 - FLEX 4)
- Paging Access (PGM 111 - FLEX 8)
- Conference Page Zone (PGM 119)
- Refer to the SBX IP 320 Installation Guide, SLT Conference Page Join Code (PGM 109 - FLEX 5)

External Device Control

Door Open

A maximum of four relays in can be used for the Door Open feature.

Operation

To use the Door Open feature:

Dial the Door Open code or press the programmed {DOOR OPEN} button.

To make a {DOOR OPEN} button:

1. Press the [TRANS/PGM] button.
2. Press the desired FLEX button to be assigned).
3. Type # + * + 1 (1st Door Open).

   #*1 = 1st Door Open

   #*2 = 2nd Door Open

   #*3 = 3rd Door Open

   #*4 = 4th Door Open
4. Press the [HOLD/SAVE] button to accept changes

Admin Programming

- External Control Contact (PGM 168)
- Door Open Timer (PGM 181 - FLEX 5)

Doorbox

A convenient intercom Doorbox can be connected to the system. The intercom Doorbox can receive page announcements and intercom calls. The Doorbox can signal assigned stations in the system. Any combination of DKTUs or Doorboxes can be arranged in the system.

Operation

To call a Doorbox:

1. Lift the handset or press the [SPEAKER] button.
2. Dial the station number of the Doorbox or press the flexible button for the Doorbox.
3. After hearing a warning tone, announce the call.

To place a call from a Doorbox:

Press the [CALL] button and the assigned station will ring.

To answer an intercom call at a station assigned for Doorbox signals:

DKTU

Press the flexible button for the Doorbox.

SLT

Go Off Hook.

To place the Doorbox in DND mode:

Press the [DND] button.

Conditions

- The Doorbox cannot attend a conference.
- A CO call cannot be received at a Doorbox.
- A Doorbox can be a member of a page zone group.
- To receive a Doorbox call at an SLT, set Doorbox Signaling value to ON (PGM 111-FLEX 6).
- If Nation code is TELKOM or ISRAEL, a DSS button for the Doorbox should be assigned to the SLT (PGM 115).
- The SLT and WHTU can receive only one call from the Doorbox.

Admin Programming

- ICM Box Signaling (PGM 111 - FLEX 6)
- Station ID Assignment (PGM 110)
- ICM Box Music Channel (PGM 171 - FLEX 3)
- ICM Box Timer (PGM 181 - FLEX 6)

Loud Bell

The Loud Bell Control (LBC) contacts are activated when the assigned station receives ringing from an incoming CO line (if assigned ring), transferred CO line, or intercom call.

Conditions

- Two LBC contacts can be assigned individually to the station. All the contacts may be assigned to the same station but, only the first contact will be activated in the intercom call.
- The LBC 1 can be programmed to be operated as an external night ring contact as well as a LBC contact.
- In the night mode, LBC 1 will follow UNA ring assignment and will ignore the station ring.
- An external ringing device should be attached to the contacts.

Admin Programming

- External Night Ring (PGM 160 - FLEX 7)
- Universal Night Answer (PGM 141 - FLEX 8)
- External Control Contacts (PGM 168)

Hunt Groups

Stations can be grouped so incoming calls may be routed to an idle station in the group. The number of Hunt groups and the number of Station in a Group are:

HUNT GROUPS	STATIONS/HUNT GROUP
10	26

Several VMIB announcements may be provided to each Hunt group. If a call is not answered when the 1st Announcement Timer expires, the second announcement will be provided if the call continues to wait. The second announcement may be repeated until the call is answered or disconnected by the User.

A Hunt Group may be assigned as one of the following 5 types:

CIRCULAR	A call is routed to the hunt group. If the first destination is unavailable or the call is unanswered, the call is routed to the next station in the group.
TERMINAL	If the call is unanswered or the destination is unavailable, it is directed to the next listed station in the group. The call will continue to be routed until reaching the last station in the group.
UCD (UNIFORM CALL DISTRIBUTION)	Calls are routed to the station in the group that has been idle for the longest time.
RING	All stations in the group will ring when a call is received.
VM (VOICE MAIL)	This group is assigned for Voice Mail and only the SLT assigned as the member of the VM group.

Common Hunt Group Features

Common Hunt Group features include:

VMIB Announcement - The SBX IP 320 system supports first and secondary VMIB announcements for the Hunt Group. When a call is received at the hunt group, the pre-assigned VMIB announcement will be played to the caller if the VMIB announcement is set and the timer expires. If the call is not answered when the second announcement timer expires, this announcement will be played. Also, the secondary announcement can be repeated as its programming.

Overflow Destination - If the overflow timer expires and the call is unanswered, the call will be diverted to the overflow destination. The overflow destination can be an extension, a Group, VMIB announcement, or a system speed dial bin number.

Wrap-up Timer - When a member of the Hunt Group goes idle, the system will not distribute calls to that member. After the wrap-up timer expires, the member Station returns to a real idle state, and ready to receive CO calls.

Music Source - The SBX IP 320 system supports up to 8 kinds of music sources for Hunt Groups not including the ring back tone. When a call goes to queue, a caller will hear the pre-assigned music source.

Multiple Member Assignment - A station can be a member of one or more Hunt Groups of the same type.

Conditions

- A station in DND/Call Forward state will not receive Group calls.
- Transferred calls to a Hunt group are not recalled.
- When a call is received in a Hunt Group, the call will be in the ring process before receiving the VMIB Announcement for the duration of the Hunt Group Announcement timer. If no Hunt Group announcement is assigned, the timer is ignored. If the timer is set to 0, the call will receive the announcement prior to the ring process.
- When a Hunt Group has guaranteed announcement (the 1st announcement timer is set to 0).
- Overflow timer is started and ring is provided after the announcement finishes playing.
- Only the 1st announcement may be used for a guaranteed announcement.
- If all stations in the Group are busy when a call is received, the call continues to wait for an available station in the Group. If queued, the call will receive MOH until the call is answered or disconnected.
- If there is no available member in a Group because every member sets DND, ACD DND, or call forward, all new calls to the Group and all queued calls in the Group are rerouted to another destination as programmed:
  - Overflow destination, if assigned.
  - Alternative destination, if the group is an ACD group and it is assigned.
- If a call is not answered when the Overflow Timer expires, it will be sent to the overflow destination while the VMIB announcement is being played.
- If overflow destination is not assigned, the call will be dropped when the overflow timer expires.
- If an Announcement Timer is set and no VMIB number is assigned, the announcement will be ignored.

- When the number of queued incoming calls are over the pre-defined amount in an ACD group, incoming CO calls will be dropped.
- The Pick-up Hunt Group is reserved for Intercom calls only.
- ISDN phones can be a member of a Hunt Group, but will only work when answering a hunt call.
- Group pick-up doesn't work with a call of Hunt Group pilot number; an error tone will be presented.
- ISDN phones cannot be the first Hunt Group member; two number entrance for one ISDN phone is not permitted.

Admin Programming

- Hunt Group Assignment (PGM 190)
- Hunt Group Attribute (PGM 191)

Hunt Group Name Service

Incoming calls to Circular/Terminal, UCD, and Ring Hunt Group types can display the name of the Hunt Group when programmed in Hunt Group Attributes (PGM 191).

Operation

Incoming CO Call -- When an incoming Hunt group call is established and a group name exists, the Hunt name will be displayed on the1st LCD field (instead of LINE RINGING).

ICM Call -- When an ICM Hunt group call is established and a group name exists, the Hunt name will be displayed on the 3rd LCD field (instead of HUNT CALL).

Conditions

- The Group Name of an ICM Hunt Call can be cut off because of LCD field length.
- If a Hunt group name is not assigned, the group number will be displayed.
- The maximum length of a Hunt Group Name is 12 characters.

Admin Programming

Hunt Attributes (PGM 191)

- FLEX 18: Circular/Terminal group
- FLEX 15: Ring group
- FLEX 23: UCD group

Automatic Call Distribution Groups (ACD Groups)

A separate supervisor or common supervisor can be assigned in an ACD group. The supervisor can monitor the status of the group. When a call is queued to a group for longer than a predefined time or when a predefined number of calls are queued, the supervisor's LCD will indicate the number of calls in queue, and the queued time for the longest queue. The supervisor can change the overflow destination and timing. The system will provide traffic and on line status reports, based on the supervisor's request for the ACD group, including the following group statistics:

- Total calls
- Number of unanswered calls
- Average and the longest queued calls
- Number and total time when all agents are busy
- Average ringing time before answer
- Average service time after answer

An ACD Supervisor can activate two-way recording when monitoring an agent's conversation.

Queue Information of Queuing Call count is automatically displayed at the Supervisor’s LCD, as well as at the Agents’ LCDs.

Operation

<u>VMIB Announcement</u>

The VMIB announcements can be assigned to provide different messages to each ACD group. The system ACD groups can be programmed to provide announcements to incoming calls to a group where all station are busy. Both primary and secondary announcements will be available and a guaranteed announcement may be assigned.

Agents

The following features are available to agents in an ACD group.

Agent Login/Logout - It is assigned by the Admin Program. An ACD agent can be assigned to more than one ACD group.

Agent On/Off Duty - Agent can be On/Off duty by dialing the ACD DND Code ([TRANS/PGM] + 87 + Hunt Group Number) on the dial pad or by pressing the pre-programmed flexible button.

Alternate ACD Group

An alternate ACD group can be programmed so that if stations are busy, the alternate will be checked for an available station.

Overflow Station Assignment

An overflow station may be assigned to route callers in queue to a designated station after a specified time. The overflow station may not be one of the ACD group stations. The overflow station can be forwarded, if enabled in Admin programming.

Display Queued Calls

LCD Display Queued call count.

```
QUE Group NO (Queue count)
```

LED Indication

The DSS LED of the {ACD Group} will flash at a greater frequency as the number of queued calls increases.

0 : Off, 01-3 : 60 IPM, 4-6 : 120 IPM, more than 7 : 240 IPM

Supervisor

Supervisor Login/Logout - It is assigned by Admin Programming. Each ACD group can be assigned a separate supervisor.

Supervisor Monitor - The supervisor monitor provides a means for an ACD supervisor to monitor an agent's call in progress or to provide assistance. When used, a supervisor may intrude into an agent's call in a listen only or in a true conference mode. This feature is available with or without warning tone.

Reroute Queued Call - The supervisor can reroute a queued call to another destination with or without answer.

Database Assignment - The supervisor can change the overflow destination, overflow time, and wrap-up time.

DKT with flexible buttons must be programmed [Supervisor Status] button.

[TRANS/PGM] + [FLEX] + [TRANS/PGM] + 8* + Group NO + [HOLD/SAVE]

ACD Statistics Report

1. Press the [Supervisor Status] button.
2. The Status & Control Menu is shown as below:

```
[1] ACD STATUS
[2] ACD DATABASE
[3] ACD DUTY
[#] ACD PRINT
```

ACD call queue status display

Dial "1" for ACD call statistics

-ACD Group-
Total Calls
Average Call Time
Average Ring Time
Busy Count and Time
Number of calls in queue
Average and Longest Queued Time
Unanswered Call Count

Access to the ACD Routing Database

Dial "2" for ACD Database Code,

Select the database item, scrolling with the [VOL UP]/[VOL DWN] button.

Overflow Destination- station/group

Overflow Time- xxx seconds

Wrap-Up Time- xxx seconds

Enter new data.

Press the [SAVE] button.

Agent Status and Control

Dial "3" for Agent Status.

```
[1] DUTY STATUS
[2] DUTY ON/OFF
[3] DUTY PRINT
```

To view Agent Status:

Dial Agent Status Code "3 + 1 (duty status)"

```
AGENT XXXX
TOTAL CALLS
AVE CALL TIME
AVE RING TIME
UNANSWERED CALLS
```

Press the '*' key or '#' key to select the next agent.

Press [VOL UP] or [VOL DOWN] key to view:

- Number of ACD calls served
- Average ACD call service time after answer.
- Average ring time before answer
- Number of unanswered ACD Calls

To control Agent duty status:

Dial ACD Agent Duty feature Code "3 + 2(duty on/off)".

Press the '*' key for agent selection.

Dial 0 or 1, (0: Off Duty, 1: On Duty).

To output Agent Statistics Report:

Dial ACD Statistics Reporting Code "3 + #(duty print)".

Press the '*' key or '#' key for a next agent selection.

Press the [MUTE] button to initialize the database; this eliminates overlap of future reports.

Below information is an example of Each Agent Statistics

```
=====================================================================
 ACD GROUP INFORMATION (*620)        Date: 07/15/08 Time: 05:42:24
 AGENT_NO | TOTAL_CALL | UNANSWERED_CALL | AVE_RING_TIME | AVE_SVC_TIME
 =====================================================================
    1001 |          0 |               0 |  00Min 00Sec |  00Min 00Sec
 =====================================================================
```

If PGM 161 - F9 ACD PRINT ENABLE is ON, ON Demand print format is not used.

Only the Periodic format can be printed as below.

~	1	=	2	=	3	=	4	=	5	cr	lf

Fields	Meanings
~ (tilde)	Means start of ACD statistics and is always located at first column
= (equal)	Delimiter between each meaningful data
1	Each Agent number
2	Total call counter
3	Unanswered call counter
4	Average ringing time (in sec)
5	Average service time (in sec)
cr	Line Feed (0x0A)
lf	Carriage Return (0x0D)

To print ACD Statistics:

1. Press a programmed {ACD Status} button:

   1(Status)/2(Dbase)/3(Duty)/#(Print)

2. Dial ACD Statistics Reporting Code "#".

3. In the reporting status (1), the supervisor can initialize all stored reporting database by pressing the [MUTE] button before hanging-up. When you press the {ACD} button in any sub state, the station goes to (2) state.

Below is an example of Group Statistics.

====== ACD GROUP INFORMATION (*620 GRP) ======

Date: 07/15/08 Time: 05:40:07

- Total Calls: 2

- Unanswered Calls: 2

- All STA Busy Count: 2

- Average Ringing Time: 00Min 00Sec

- Average Service Time: 00Min 00Sec

- All STA Busy Time: 01Min 17Sec

- Calls in Queue, Now: 0

Periodic Print ACD Statistics

This ACD Statistics format is used when PGM 161 - FLEX 9 ACD PRINT ENABLE is ON.

If this option is ON, ON Demand print format is not used. Only the Periodic format can be printed.

~	1	=	2	=	3	=	4	=	5	=	6	=	7	=	8	=	9	=	0	cr	lf

Fields	Meanings
~ (tilde)	Means start of ACD statistics and is always located at first column
= (equal)	Delimiter between each meaningful data
1	ACD Group number
2	Total call counter
3	Unanswered call counter
4	All busy counter
5	Average ringing time (in sec)
6	Average call service time (in sec)
7	Total busy time (in sec)
8	Number of current queued calls
9	Longest queued time
0	Average queued time
cr	Line Feed (0x0A)
lf	Carriage Return (0x0D)

To go to the Main Menu

Press the [HOLD/SAVE] button, then go to the procedure below.

To re-route queued calls with answer by the supervisor:

1. Assign a flexible button as an {ACD Group} button.
2. Press the flashing {ACD Group} button.
3. The first queued incoming CO call is routed to the supervisor.
4. If the supervisor lifts handset, the preferred line will be answered.

The following operation follows the normal operation for the DSS button feature assigned to "station group".

To re-route queued calls without answer by the supervisor:

1. Assign a flexible button as an {ACD Group} button. The DSS LED of the {ACD Group} button flashes when there are queued calls.
2. A supervisor presses "No answer reroute code" 564 and destination station number or {DSS} to reroute the first queued call.

To monitor an agent's conversation at the ACD supervisor station:

1. Call the busy agent and receive a busy tone.
2. Press the {ACD} flexible button.
3. The supervisor can monitor the agent, but will not send audio to the agent.

To Record an agent's conversation at the ACD supervisor station:

1. Call the busy agent and receive a busy tone.
2. Press the {ACD} flexible button.
3. The supervisor can monitor the agent, but will not send audio to the agent.
4. Press the [Two Way Recording] button.
5. When Two Way Record is stopped, the recorded VMIB message is left at the supervisor station.

To activate ACD-DND manually:

1. Assign {ACD-DND} with ACD-DND feature code [Trans] + 87 + Group NO on station.
2. An ACD member user presses the {ACD-DND} button in order to go into the ACD-DND mode. The {ACD-DND} button turns steady on when in the ACD-DND mode.

To deactivate ACD-DND:

Press the illuminated {ACD-DND} button.

Conditions

- The supervisor should assign a flexible button for {ACD}.

  [TRANS/PGM] + Flex. BTN + [TRANS/PGM] + 8* + ACD Group No + [HOLD/SAVE]
- The user can see the same group status as seen by the ACD supervisor or by printing periodically to RS-232C print.

- To print ACD statistics periodically, set the ACD PRINT ENABLE(PGM 161-F9) and the ACD Print Timer (PGM 161-FLEX 10: 10sec base).
- The agents can also print and view the same ACD statistics as the supervisor.
- The agent can login and logout using Hot Desk feature.
- If "ACD PRINT ENABLE (PGM 161 - FLEX 9)" is set to ON, only periodic statistic is printed.
- A guaranteed announcement is obtained by assigning 0 seconds to the first announcement.

Admin Programming

- Hunt Group Assignment (PGM 190)
- Hunt Group Attribute (PGM 191)
- ACD Print Enable (PGM 161 - FLEX 9)
- ACD Print Timer (PGM 161 - FLEX 10)
- ACD Clear Database after Print (PGM 161 - FLEX 11)

Circular Group

In Circular Hunt, calls to a station in the group or a pilot number will go to the station or an idle station in the group. If unavailable or unanswered in the hunt no answer time, the call is directed to the next station in the group. The call will continue to route until each station in the group has been tried. The call will remain at the last station in the group or will be passed to the assigned overflow station or the assigned overflow group. A circular Hunt group can be assigned with a pilot number (Hunt group number) and only calls to the pilot number will hunt.

Ring Group

All the stations in the group receive ring simultaneously for a call of Hunt group until one of stations received ring answers the call. If call is not answered until overflow timer, it will be sent to an overflow destination, if assigned.

Terminal Group

Calls to a station in the group or a pilot number will go to the first station in the group. If unanswered or unavailable, the call proceeds to the next listed station in the group. The call will continue to be rerouted until reaching the last station in the group where the call will remain or can be sent to an overflow station/group. A terminal Hunt group can be assigned with a pilot number (Hunt group number) and only calls to the pilot number will hunt.

UCD Group (Unified Call Distribution)

Calls are sent to the group by dialing the pilot number (Hunt group Number) or assigning CO lines to directly terminate to the group. Calls are directed to the station in the group that has been idle for the longest time. If all stations in the group are busy when a call is received for the group, the call may be routed to an alternate location, or may continue to wait (queue) for an available station in the group. Based on programming, the queued call may be taken out of the group and directed to an overflow station.

The member of UCD group can assign DND. A station on UCD-DND will not receive calls.

Operation

To assign a UCD-DND button:

1. Press [TRANS/PGM].
2. Press FLEX button.
3. Press [TRANS/PGM].
4. Dial 87.
5. Enter the Hunt Group Number.
6. Press the [HOLD/SAVE] button to save changes.

Note: Do not make a loop with UCD group alternative destination.

Condition

- The SBX IP 320 system supports VMIB announcements for hunt groups. When a call is received at the secondary VMIB announcement, the caller can be connected to another station by the number entered with CCR service (PGM 228).

Voice Mail Group

This group is assigned for voice mail and only SLTs can be assigned as members of this Hunt group. When the VM group is called, the system will search for an idle member in the calling VM group with Terminal type or Circular type hunting.

Incoming Call Pickup

CO Line Name

This feature allows the capability to name each CO Line. Stations with an LCD interface screen, including the Attendant Station, will display the programmed CO Line Name in place of the default "LINE XXX" display.

Conditions

- This applies to all conditions where the LINE XXX message is displayed. However, SMDR will display the line number in place of the programmed name.
- A CO Line Name can be assigned to each CO Line.
- Each CO Line Name can contain up to 12 characters.
- If the CO Line Name Display is set to OFF at PGM 142 - FLEX 1 selected, the CO Line Name is not displayed even if the name is programmed.

Admin Programming

- CO Line Name Display (PGM 142 - FLEX 1)
- CO Line Name Assignment (PGM 142 - FLEX 2)

Customer Call Routing (CCR) with VMIB

Customer Call Routing (CCR) is the incoming CO Call type of DISA/DID where the user can route the destination by pressing only one digit. If a user presses a certain digit, the corresponding VMIB announcement is played. When the user presses the desired digit again, call routing is established.

A user also may access the desired destination directly by dialing the Station or Hunt Group number, or VMIB announcement.

Operation

When a call is answered by a system programmed with CCR, a VMIB announcement should be heard by the caller. VMIB announcement gives a choice of destination; the caller may select a destination based on the information presented in the VMIB Announcement.

- If a user chooses a destination, and is routed to the destination, but that destination is busy, the call is rerouted to BUSY destination (PGM 228 - FLEX 11).
- If a user does not choose any destination until the DISA retry count is over or the destination is an error, the call is rerouted to the TIME OUT /ERROR destination (PGM 228 - FLEX 12).
- If a user chooses a destination, and is routed to the destination, but that destination is no answer, the call is rerouted to the NO ANSWER destination (PGM 228 - FLEX 13).

To program the CCR Table, use the following procedure:

1. Select CCR Table 01 to match with the VMIB announcement number at PGM 228.
2. Press [FLEX 1] to set the Sales Dept. The Flexible button number should be the same as the dialing digit number.
3. Select Destination Type as Hunt Group by dialing 2. Enter the desired Hunt Group Number 620.
4. Press [FLEX 2] to set the Technical Dept.
5. Select destination type as VMIB by dialing 2 or 3 and enter Announcement Number 02.

Note: If type 3 (VMIB Drop) is selected, the call will be dropped after the Announcement is played. If Type 2 (VMIB) is selected and the appropriate CCR Table 02 is programmed, the call will be routed to the Technical Dept.

6. Press [FLEX 10] to set the Operator. Select the destination type as Station by dialing 1 and enter the desired Station Number 101.

To use DISA CCR:

1. Verify the CO Service Type is set to Normal at PGM 140.
2. Press [FLEX 1].
3. Verify the DISA Service is set to ON at PGM 140.
4. Press [FLEX 2].
5. Set VMIB Message Number to 01 PGM 140 and press [FLEX 2].

To use DID CCR:

1. Verify the CO Service Type is set to DID/MSN at PGM 140.
2. Press [FLEX 1]
3. Verify the DID Digit Receive Number is set to 3 at PGM 146.
4. Set DID Conversion Type as 2 (using Flexible DID Table) at PGM 143
5. Press [FLEX 4]
6. Set the Destination Type as 3 (VMIB)
7. Select Announcement Number 01 for the Flexible DID Table at PGM 231.

Conditions

- To use the CCR feature for DID, VMIB should be assigned for Flexible DID Destination (PGM 231).
- The CCR feature is only supported for DID and DISA.
- If a caller dials a full destination number, the call will be directly routed to the desired destination by the system numbering plan.
- If a caller dials one digit then pauses, the SBX IP 320 system will compare the digit with the CCR Table. If a matching digit is found on the CCR Table, and the bin number is the same as the VMIB Announcement, the call will be routed to the programmed destination.
- If the dialed digit is invalid, the caller can attempt to redial up to 3 times (the DISA Retry Counter is also programmable). When the DISA Retry Counter is exceeded, the call will be routed to the recall destination or disconnected following an error tone.
- VMIB announcement 01-70 may be used for CCR.
- Call routing will be operated with previously programmed VMIB announcement.
- The maximum CCR depth is 10.

- The external user can dial alternate digits while the VMIB announcement is being played or the digits should be entered within the Inter-digit Time (5 sec) after the announcement is ended.
- If a user presses the # button while CCR is in operation, CCR will return to the first step of Operation.
- If a user presses the * button while CCR is in operation, CCR will return to the previous step.
- The call will be dropped directly after the VMIB announcement, if VMIB Drop is selected at the CCR Table.
- If a call is routed to System Speed Dial, the call will be routed to the applicable Speed Dial destination. If the CO Call is assigned System Speed Dial, the routing will be the same as Incoming CO Off-net Call Forward.
- The current CCR announcement (current depth) will be provided again until the DISA retry count is over when the user does not press any digit, destination is busy, or in error cases. The CCR announcement retry count is calculated by each CCR announcement depth.
- Each PGM 228 - CCR table (each VMIB announcement) must have a value. If PGM 228 - CCR table 01 (system announcement 01) isn't assigned any value, the call will be routed to the DID/DISA destination (PGM 167).
- VMIB prompt usage option is set in PGM 167 - Flex 5.
- If destination is No answer, after DID/DISA no answer timer expires, the call is routed to the No answer destination directly and does not follow the retry count.

Admin Programming

- Flexible DID Table (PGM 231)
- DID Digit Conversion Table (PGM 146 - FLEX 5/ FLEX 6)
- DISA Retry Counter (PGM 160 - FLEX 4)
- CCR Inter-digit Timer (PGM 180 - FLEX 15)
- Inter-digit Timer (PGM 181 - FLEX 8)
- DID/DISA Destination (PGM 167)
- Custom Call Routing Table (PGM 228)

Direct Inward Dialing (DID)

This feature allows CO incoming calls to access a specific destination. This feature enables the caller direct access to a desired Station, Hunt group, VMIB announcement, Speed, or Page, bypassing the Attendant.

There are 3 types of DID Conversions that can be set by Admin Programming (PGM 143 - FLEX 4):

Type 0 - In an incoming DID call, select some digits which are received digits by Admin Programming. The selected digits will be converted by DID Conversion type (PGM 146 - FLEX 5/FLEX 6).

Type 1 - The incoming DID digits are the destination number. There is no conversion.

Type 2 - With result of DID Conversion type 1, convert by the Flexible DID table (PGM 231) additionally.

Operation

Ex. 1: To make a DID call by DID Digit Conversion:

1. Check if DID Digit Receive Number is set to 3 at PGM 146.
2. Set DID Conversion Type to 0 (using Digit Mask) at PGM 143 and press FLEX 4.
3. Set DID Digit Mask by pressing #1**. The first digit 2 is ignored and second digit 6 is converted to 1 and the last two digits are bypassed.

Ex. 2: To make a DID call using the Flexible DID Table:

1. Verify that DID Digit Receive Number is set to 3 at PGM 146.
2. Set DID Conversion Type as 2 (using Flexible DID Table) at PGM 143 and press FLEX 4.
3. Program Flexible DID Table as shown in the PGM 231 Flex DID Conversion Table.

Conditions

When a DID call is received at a busy station, the call is automatically handled according to the DID/DISA Answer Timer.

- Destination of calls handled can be Station, Hunt group, VMIB announcement, Drop after VMIB announcement, System Speed, Internal Page, External Page or, Internal/External/All Call Page.
- If the call is not answered or the number was invalid, the call is routed by DID/DISA Destination.

- DID calls to a busy station can be placed on a waiting stage according to admin programming to KTU and SLT.

Admin Programming

- DID Conversion Type (PGM 143 - FLEX 4)
- DID Digit Conversion Mask (PGM 146 - FLEX 2)
- Automatic Speaker Selection (PGM 111 - FLEX 1)
- DID/DISA Destination (PGM 167)
- DID Call Wait (PGM 114 - FLEX 17)
- DID Restriction (PGM 114 - FLEX 16)

DID Call routing with Incoming CLI

A DID incoming call destination can be assigned by the incoming CLI. If a CLI number is registered at the Incoming CLI Destination Table, all DID calls with these CLI will be routed to a registered specific place.

The ICLID Table (Incoming CLI Destination Table) has two fields, one is the CLID field, and the other is the Destination Index field. The Destination Index field represents an index number of the DID Conv Table (PGM 231).

If CLI from an external call matches with the CLID field, the call will reference the Destination Index field, and then look at the registered destination of the index from the DID Conv Table.

Operation

To set the Incoming CLI and Index Tables:

1. Press PGM 237 and choose a table number.
2. Register a CLI number at Button 1
3. Assign an Index Table number at Button 2.

Rerouting destination by Incoming CLI:

1. DID incoming call has own CLI
2. This call will be analyzed with ICLID Table.
3. If CLI of this call matches a registered CLI field, then the system looks up the DID Conv Table with the saved index.

4. This call will be delivered to the specific destination designated by the ICLID Index Table.

Conditions

- This feature is supported only when the CO type is DID.
- This feature is executed first, when system receives a DID call with CLI.
- The ICLID Table is an auto-sorted table. So when the Administrator enters the CLI and Conv Table index number, this field is saved automatically in a sorted order.
- If a saved CLI number is the same as some previous field's CLI, the system recognizes this purpose is to change the Conv Table index field.
- When the Administrator erases a CLI field, the Conv-Table index will be erased.
- The ICLID Table can be initialized by PGM 450 - FLEX 11 (Initialization Other Tables)
- If the ICLID Usage option of a CO line is off, this CO line will not be checked with ICLID tables. Only a CO line with the ICLID Usage option ON, will be applied with the ICLID feature. (PGM 143 - FLEX 18)
- A wild-card number "*" can be used in the CLI field. If "*" is registered, only registered numbers before the "*" will be checked with the received CLI number. If all numbers match before the "*", this call will be handled by the ICLID feature.

Admin Programming

- PGM 237
- PGM 143 - FLEX 18

DID/DISA Call Routing of DND Station or Station with Pre-selected Message Active

A DID/DISA call can be routed to a specific destination when a called station is in DND or in Pre-selected message modes.

There are three destinations (Tone/Attendant/Hunt Group). The flow is the same for DID/DISA as the other Destinations in PGM 167. DND/P-MSG rerouted flow is the same as the busy reroute flow.

Operation

Rerouting destination is Tone (PGM 167 - FLEX 4, then FLEX 1)

1. Call to Station XXX and Station XXX is DND or Pre-Selected MSG Set state.
2. Caller hears busy tone.

Rerouting destination is Attendant (PGM 167 - FLEX 4, then FLEX 2)

1. Call to Station XXX and Station XXX is DND or Pre-Selected MSG Set state.
2. This call will be routed to the Attendant.

Rerouting destination is Hunt Group (PGM 167 - FLEX 4, then FLEX 3)

1. Call to Station XXX and Station XXX is DND or Pre-Selected MSG Set state.
2. This call will be routed to the assigned hunt group.

Conditions

- If [P-MSG DND] Admin is off, the route operation doesn't work even though the station is Pre-Selected MSG state.

Admin Programming

- PGM 167 - FLEX 4

Direct Inward System Access (DISA)

The DISA feature allows incoming CO calls to access a specific destination, bypassing the attendant station. Unlike DID, there is no digit conversion in DISA.

On accessing an incoming CO Line, the system will give the pre-recorded VMIB announcement or dial tone. The caller then is able to dial additional digits to access their desired destination on the system.

Operation

To program DISA Line Assignment, use the following procedure:

1. Press [TRANS/PGM] and enter 140.
2. Select the CO Line Range to be assigned to DISA Line.
3. Press [FLEX 2], and select the Ring type and press [FLEX 1].
4. To select the Ring Type, press [FLEX 1] for Day, [FLEX 2] for Night, [FLEX 3] for Weekend, or [FLEX 4] for On-demand.
5. Dial 1.
6. Press [HOLD/SAVE] to activate the DISA Line Assignment.
7. Press [FLEX 2], select the Ring Type.
8. Press [FLEX 2] then enter 01-70 for the VMIB Greeting Assignment.
9. Press [HOLD/SAVE] to accept changes.

To use DISA Line Assignment, use the following procedure:

1. Select the DISA Line you wish to use.
2. When the tone or announcement is heard, dial the desired station/ hunt group number.
3. After a connection with the system is made, dial the CO Access Code (Ex., 8801) to call again outside of the system by securing another CO Line.

Conditions

- You can assign the VMIB announcement instead of the intercom dial tone on a DISA line.
- If the DISA Authorization Code is enabled for a DISA Line, a DND warning tone or VMIB announcement is heard, guiding the user to enter the DISA Authorization Code, the dial tone then should be heard.
- Each DISA Line may be assigned as full-time DISA or Night Mode Only.

- Night mode DISA operates as a normal CO Line during Day mode.
- If the VMIB announcement number is stored with #, the CO Line will be dropped after the VMIB announcement is played.
- If the DISA Authorization Code is disabled, the permission is determined by CO to CO COS & CO COS.
- If the DISA Authorization Code is enabled, the Authorization Code should be entered to access an outgoing CO Line. If the Authorization Code is matched with the Authorization Code of the station, the user may access the CO Line depending on STA COS & CO COS. If the Authorization Code is matched with the Authorization Code of system, it is determined by CO to CO COS and CO COS.

Admin Programming

- DISA Line Assignment (PGM 140 - FLEX 2)
- DISA Account Code (PGM 141 - FLEX 3)
- DISA Retry Counter (PGM 160 - FLEX 4)
- CO-to-CO COS Assignment (PGM 166)
- Weekly Time Table (PGM 233)
- DISA Restriction (PGM 114 - FLEX 10)

Preferred Line Answer (PLA)

If Preferred Line Answer (PLA) service is enabled and there are several incoming CO Calls (Transferred, Recalled, Queued, or Normal Incoming Call) at the same time, the first answered call can be chosen by setting the PLA priority.

Note: The default setting for answer order is:

Transferred Call >Recalled Call >Normal Incoming Call > CO Line Queued Call

Operation

If there are multiple CO Calls ringing at a station and the call is answered at one of the stations, the call with the highest priority will automatically be answered first.

Conditions

- Automatic Speaker Select feature should be enabled.
- The Priority of CO Line for PLA can be changed by Admin Programming.

Admin Programming

- Preferred Line Answer (PGM 112 - FLEX 7)
- Automatic Speaker Selection (PGM 111 - FLEX 1)
- PLA Priority Setting (PGM 173)

Ring Assignment

A pre-assigned destination receives incoming calls through the CO Line. The destination can be a Station, Hunt Group, or VMIB Announcement. If a destination station is busy, the incoming call returns a muted ring signal, so the station user can pickup the incoming CO Call as needed.

Operation

Ex. 1 When there's an incoming CO Call through CO Lines 1-8 during Day mode, the Stations 101-105 (as available) start to ring instantly. If one of the stations answers the call, other stations stop ringing. After 9 seconds, if the call is still not answered, station 100 (Attendant) starts to ring.

1. Set CO Service Type as Normal at the PGM 140 menu item.
2. At PGM 144, select CO Line Range 01-08 and press FLEX 1 for Day Mode.
3. Dial 1 for the Station, and enter the station range 101105.
4. Press 0 to make the stations ring instantly when there is an incoming call.
5. To save changes, press the [HOLD/SAVE] button.
6. Press FLEX 1 for Day Mode again without exiting PGM 144, and press 1 for the Station again.
7. Enter Station Range as 100100, and dial 9 as the delay value. Press [HOLD/SAVE] again.

Ex. 2 When there's an incoming CO Call through CO Lines 1-8 during Night mode, the Hunt Group starts to ring. The ringing station is decided by the Hunt Group type.

1. Verify the CO Service type is set to Normal at PGM 140.
2. Check if Hunt Group 620 is assigned properly at PGM 190.
3. At PGM 144, select CO Line range 01-08 and press FLEX 2 (Night Mode).
4. Dial 2 for the Hunt, and enter Hunt Group Number 620.
5. To save changes, press the [HOLD/SAVE] button.

Ex. 3 When there's an incoming CO Call through CO Lines 1-8 during Weekend mode, the VMIB announcement played. If # is pressed, the line will be released.

1. Check if CO Service type is set to Normal at PGM 140.
2. Check if VMIB Announcement 01 is recorded properly at the System Attendant Station.
3. At PGM 144, select CO Line range 01-08 and press FLEX 3 for Weekend Mode.
4. Dial 3 for VMIB, and enter the VMIB Announcement Number 01.
5. To make the CO Line release after VMIB announcement, press the # key.
6. Save the changed setting by pressing the [HOLD/SAVE] button.

Condition

- Any CO Line Ring Assignment can be programmed for multiple stations, or all stations. And each ring to a station can be delayed by ADMIN programming. The ring assignment is individually applied to ring modes Day, Night, weekend or On-demand. Every CO Line must be assigned to an Attendant Station by default.
- To receive incoming CO Line Calls, the DKTU should have a {CO} or {LOOP} button.

Admin Programming

- CO Service Type (PGM 140 - FLEX 1)
- CO Ring Assignment (PGM 144)
- Weekly Time Table (PGM 233)
- Hunt Group (PGM 190)

Universal Night Answer (UNA)

If the CO Line is programmed for Universal Night Answer (UNA), any user can pick up incoming CO Calls during Night Mode by dialing the Night Answer code 569, regardless of the pick-up group.

If there's incoming CO Calls during Night Mode, Station B can pick up the call even though Station A and B do not belong to a pick up group.

Operation

To pick-up a call in Night Mode:

1. Lift the handset or press the [SPEAKER] button. The intercom dial tone sounds.
2. Dial 569, the Universal Night Answer code. The call is connected.

Conditions

- UNA feature is activated when the Ring Mode is Night.
- If there isn't an incoming CO Call when the Night Answer Code is dialed from a station, an error tone will sound.
- The connected CO Line may be transferred or disconnected similar to Day Mode call handling.
- If External Night Ringing is set to On, the call is routed to External Page by LBC1.

Admin Programming

- Universal Night Answer (PGM 141 - FLEX 8)
- External Night Ring (PGM 160 - FLEX 7)

Internet Protocol (H.450)

The SBX IP 320 System supports H.450 over IP (Internet Protocol) networking. The Networking System can link telephone systems together so that they behave like a unified communication network, providing service transparency, cost-efficiency and adaptability to your organization's needs.

Network Configurations

The System can support various interconnected corporate network configurations via the Internet and up to 72 systems (including itself) can be combined on the network.

Networking Protocols

The System can support H.450 over IP - for the basic networking functions and proprietary protocol for advanced networking functions.

- IP standard protocolH.323 for Call Control, H.450.1 - H.450.12
- LG-NORTEL CO. LTD. Proprietary protocol for advanced Function

Requirements

To use the networking features, software lock-key installation is required. Each SBX IP 320 system has its own unique software lock-key. To get the software lock-key, contact the distributor of your SBX IP 320 system.

Numbering Plan

Unified Dialing Plan (UDP) - In the SBX IP 320 networking system, the UDP numbering plan is used. In the UDP networking, the stations of each system can have a unique station number from 2 digits up to 7 digits. The unique station number is assigned according to the numbering plan of each SBX IP 320 system.

Automatic Routing Service

A dialed number is analyzed and routed to access the correct destination according to the Net Numbering Table. The System supports an alternative route when the main path fails. In order to maximize the use of Networking and efficiency of the private network, the Net Numbering Table provides a simple accessibility to the end User.

Absent Text Message

Absent Text Message is available within the networking connection environment.

If a Station User leaves his desk with an Absent Text Message active at the Station, an incoming call to this Station from another Networking System will receive the absent text message (displayed on the calling Station LCD).

Operation

When Absent Text Message is active at a Station, operation of the feature is automatic when a call is received from a Station from another Networked System.

Condition

- A text message will be displayed on the LCD of the networked calling station; the called station still will receive the ring signal.

Attendant Call Service (CAS)

An Attendant call from a Station on a Networked System can be routed to the Centralized Attendant (CA) of Call Attendant Service (CAS) master system. This call will be queued when all centralized attendants are busy, like to the normal attendant call and queuing operation.

Operation

1. Assign Net DSS to the attendant (except System Attendant).
2. System Attendant presses ATD DND button.
3. Dial Attendant code at any station in the system, then the call will be routed to the Net Attendant. The system provides Ring Back Tone to calling station and the [NET DSS] button lights.

Conditions

- ICM Call is routed to Net attendant if system attendant is in ATD_DND mode.
- CO Call can be routed to Net attendant only if DID/DISA destination is ATD and the ring is not assigned to any station.

Admin Programming

- Networking CAS Enable (PGM 320)
- Flex Button Assignment - Station Programming Button (PGM 115)
- Attendant Assignment (PGM 164)
- DID/DISA Destination (PGM 167)
- CO Ring Assignment (PGM 144)

Busy Lamp Field (BLF)

A Station on a Networked System can program the appearance of a Busy Lamp Field (BLF) status on other Stations in the Networked System. The BLF button can also be used to make a Net Call to another Networked Station.

To use BLF service, the manager software, and PC application, must be installed from a PC. The Gate Keeper is the PC server that is installed with the BLF manager software for operational purposes.

The BLF manager software periodically receives the status of Station from whole system.

Note: The UDP port will be used to send the status information, and TCP port will be used to send other information.

The BLF manager software sends the broadcast message to the whole System when the status changes.

Conditions

- The BLF manager software should be installed on one System for the whole Networked System.
- The number of Net DSS can be restricted according to the capability of each System.
- When a flexible button on a Station is assigned as the [NET DSS] button of another system, the System serves as local BLF to indicate the status of the Station.
- CO BLF is not supported, and also the ringing signal does not update the status of that Station - ICM / CO / Transfer / CO Recall ring.
- When a Station is in DND mode, the [NET DSS] button will flash.
- If the BLF manager does not exist, BLF services can be activated between only two Systems. The address of the Gate Keeper should be registered on each System.

Admin Programming

- Networking Supplementary Attributes (PGM 321)
- TCP Port Assign (PGM 321 - FLEX 2)
- UDP Port Assign (PGM 321 - FLEX 3)
- BLF Manager IP Address Assign (PGM 321 - FLEX 4)
- Duration of BLF Status (PGM 321 - FLEX 5)

Call Completion

Call Completion is the same service as Call Back, except that it is conducted within a networking connection environment, and is used at H.450 protocol standard specification.

There are two kinds of call completion as follows:

Completion of Calls to Busy Subscribers (CCBS) - After calling a user in another system using basic call and encountering a busy tone. A station user can be notified when the busy destination of another system becomes idle. If the user wants to make a call to the destination on that notification, the call can be reinitiated to the destination of another system again.

Completion of Calls on No Reply (CCNR) - After calling a user in another system using basic call and encountering no reply. The caller can be notified when the destination becomes an idle status after some actions. If the caller wants to make a call to the destination, the call can be reinitiated to the destination again.

Operation

To activate CCBS, use the following procedure:

1. Dial a Station on another networking System.
2. If the busy tone is provided, press the [CALLBK] button; a confirmation tone will sound, and the call will be disconnected.
3. When busy Called Station returns to Idle, the Caller will receive a call-back ring.
4. When the Caller answers the call-back ring, a new call will be initiated to the Calling Station.

Conditions

- A new Call Back request left at a Station will cancel any previous callback messages.
- A voice message cannot be left even though the VMIB is installed in a local system.

- When the originator does not answer the call back ring within net timer, the call will be cleared.
- There are two CCBS modes-Connection Mode and Disconnection Mode. This can be selected at (PGM 320).

Admin Programming

- Networking Basic Attribute (PGM 320)

Call Offer

Call Offer is the same service as Camp-On, except that the Camp-On is executed in a Networking connection environment at the H.450 protocol standard specification.

When a Station User initiates a Net Call to a busy Station that is located on another networking system, a busy tone will be heard. At that time, the caller can give a signal to the busy station by using the Call Offer service. The busy station (through the receiver or on speakerphone) is notified of the call waiting by a camp-on tone and flashing LED of the [HOLD/SAVE] button.

Operation

To activate Call Offer:

1. Dial a valid Station number on another networking system; if busy, the caller will hear a busy tone.
2. Press the Camp-On code (*); the busy Station will receive an off-hook muted ring (the Calling Station will hear a ring-back tone instead of a busy-tone).

To answer the Call Offer:

1. Press the flashing CO line button while receiving a muted ring.

   -or-

   The muted ring is changed to normal CO ring when you go on-hook state.
2. Then Station User can answer the offered call.

Conditions

- Call Offer is only applied to a station that is in talk status.
- During a conference or page, Call Offer is not operative.
- The SBX IP 320 System does not support the path reservation mode of a standard QSIG specification.

Admin Programming

- Networking Basic Attribute (PGM 320)

Centralized SMDR for Transit Call

The SMDR call records from the Master System will include the station number from the Networked Systems (NET-Number) on CO Transit-In and CO Transit-Out calls.

Operation

The NET-Number automatically will be included in the Centralized SMDR output for CO Transit-In and CO Transit-Out calls.

Conditions

- Up to 4 digits can be displayed in the Station column.
- Only applies to ISDN CO lines on the Master system.

Example SMDR Printout from the Master System:

```
---- Site Name : MASTER     ----------------------------------------------------

 NO   STA   CO   TIME       START        DIALED      ACT  CNT   COST  ACCOUNT CODE

------  -----  ---  ------------ ----------- ------ ----------------  ----  ----  -------------

0027   200   001  00:00:05  02/02/08 05:46  O85620140    0     0

0026   200   011  00:00:10  02/02/08 05:46  I RING 00:00


0029 CO006   011  00:00:02  02/02/08 05:46  O200        0     0

0028 CO011   006  00:00:03  02/02/08 05:46  I RING 00:01

---------------------------------------------------------------------------------
```

Lines 0026 & 0027 show the resultant printout for a CO Transit-Out call from Station 200 on the Networked System. The System used ISDN Line 001 and VOIP Line 011 between Systems.

Lines 0028 & 0029 show the resultant printout for an incoming CO Transit-In call to Station 200 on the Networked Systems. ISDN Line 006 was used for the incoming call and VOIP Line 011 was used between Systems. The NET-Number (200) is displayed as the Outgoing Number from the Master System to the Networked Systems.

Centralized VMS

All voice mail within Networked Systems can be recorded in an external VMS.

Conditions

- The centralized VMS should be assigned in Networked Systems, and the number of the centralized VMS should use the representative number of mail accesses created in the Master System.
- The Numbering Plan including the representative of mail access assigned in Master System should be included in the Numbering Plan of QSIG group in Networked System.

Admin Programming

- Network Destination MBU IP (PGM 324)
- Station Group Assignment - Centralized VMS Assign (PGM 190)

CO Ring Assignment

A user can assign a CO ring to one of the networked stations. Only one Net-Station can be assigned by Net-Destination.

If all network CO lines are busy, an incoming call will be routed to an error destination (PGM167 - FLEX 2).

Operation

To set Incoming CO Ring Assign to a Net-Station:

1. Dial PGM 144 and choose the desired CO line (01-12).
2. Select Net-Station type (4) and assign to a Net-Station number.

    Incoming calls to the CO line specified will be routed to the assigned network user.

Call sent to destination

- If ring assignment is set to Net-Station, the system will search for an idle Net CO line.
- If the Net CO line is idle, the incoming call will be connected to the assigned Net-Destination-Station.
- If the Net CO line is busy, this call will be routed to the Error-Destination specified.

Conditions

- This feature is based on the Network feature.

Admin Programming

- Network Station is Added for Assign Destination (PGM 144)

CO Transit - In

The CO Transit-In automatically re-routes incoming DID/MSN calls from another networked System.

1. An outside caller makes a DID call to station 202 and the DID call request arrives at the master system. The networking system of station 202 is not connected to the PSTN directly, but it can be connected to the PSTN through another networking system (master system).
2. The master system checks the received DID call destination. If the DID destination matches a station of its system, the DID call is routed to the station. If the DID destination does not match, the master system searches the network numbering plan table to determine whether the destination matches with a station in another networking system. In this case, the master system transfers the received DID call request to the found networking system.
3. The CO transit-in DID call rings station 202. The caller can converse when station 202 answers.

Operation

- When an incoming call is received from the PABX, the call is automatically routed, and a Network CO line is secured according to the DID conversion.
- The incoming call will ring with CLI at the Station being called; the Caller will hear a ring-back tone.
- Once the call is answered at the Station, it will be connected.

Conditions

- There are no timers affiliated with CO Transit-In operation.
- The Caller will hear a busy tone if a networking path is not available during transit.

Admin Programming

- Networking Routing Table - Numbering Plan (PGM 324)
- DID Conversion Type (PGM 143)
- Flexible DID Table - Assign Slave Extension (PGM 231)

Additional Programming

PGM 143 - FLEX 4 (DID Conversion Type) = 2

CO Transit - Out

The CO Transit-Out increases efficiency of CO lines and reduces call costs by routing outgoing CO calls to the nearest appropriate point on the networked System. The System should be provided sufficient digit translation and string analysis options to enable the switch to route the call correctly.

1. Station 202 of the slave system dials the CO TRANSIT-OUT code, that is registered in the network numbering plan table (PGM 324). The term of CO TRANSIT-OUT code is that a station of a slave system can seize the PSTN CO line of the master system.
2. The slave system seizes a CO line that is preprogrammed for transmitting the CO TRANSIT-OUT code to the master system. And, the dialed code is transmitted to the master system.
3. The master system seizes a CO line that is preprogrammed to serve the lend request by the CO TRANSIT-OUT code. After seizing the PSTN CO line, the master system sends the digit information to the PSTN that is dialed from the station of the slave system.
4. The called user of PSTN receives ringing. The caller and called users can converse when the called user answers.

Operation

To activate the CO Transit-Out:

1. Dial the CO Transit-Out code from a Station on another networked System.
2. The Networked System will secure a CO line that is preprogrammed for transmitting the CO Transit-Out codes to the Master System
3. The Master System will secure a CO line that is preprogrammed to receive the CO Transit-Out codes.
4. After seizing a PSTN CO line, the Master System will send the digit information to place a call.
5. The Called Station will ring, and the call will be connected when it is answered.

Conditions

- To use CO Transit-Out, the networked system user must dial the CO transit-out code; the CO Transit-Out service is not activated by pressing a CO Line button.
- The Station COS of the networked system is applied for toll restriction.
- At the Master System, the Attendant must have CO access authority to make a public connection.

- For CO Transit-Out, any code will be available using NET Routing Table (PGM 324). If there is a conflict between NET Routing Table and System Numbering Plan, the System Numbering Plan will have the highest priority.
- If the CO Transit-Out code is programmed in the Network Numbering Plan Table, then the type value of this entry must be set to PSTN (PGM 324).
- If the Networked System isn't connected to the Master System directly, the CO Transit-Out code can be transmitted through the other networked Systems transparently by setting the PGM 324.
- At the Master System, the PSTN CO lines must be set (PGM 322).

Admin Programming

- Networking CO Line Type (PGM 322 - FLEX 2)
- Network Routing Table - Numbering Plan (PGM 324)

Do-Not-Disturb (DND)

Net Calls to a Station can be rejected using the DND mode; the Calling Station will hear a busy tone.

Operation

To activate DND mode:

1. Press the [DND] button at the Station.
2. When a call is received at the Station, the Caller will hear a busy tone; the Station in DND mode will not receive any notification of the call.

Condition

- When a Station is in DND mode, the [NET DSS] of the DND Station will flash if the BLF (Busy Lamp Field) manager is activated.

Admin Programming

- DND Attribute (PGM 111)

Identification Service

Between Networking Systems, a Station Name can be transmitted via the networking signaling messages.

Calling Name Identification Presentation (CNIP) - the name of a Station is transmitted when an outgoing call is made.

Connected Name Identification Presentation (CONP) - the name of a station is transmitted when an incoming call is answered. If the opposite side Networking System receives the name of the station, then it is displayed on LCD.

Operation

CNIP and CONP operation is executed whenever the station of networking system makes a call and it answers the incoming call.

Condition

- To use the CNIP and CONP services, the name of the Station and the related PGM 320 - FLEX 3 & 4 must be set properly.

Admin Programming

- Networking CNIP and CONP Enable (PGM 320)

Message Waiting Indication (MWI)

The Message Waiting Indication (MWI) is the same service as the Calling Line Identification (CLI) message wait.

A Station can leave a MWI message when a station receives a No Answer on another Networking System. If MWI is enabled at the called Station, software will cause the Station lamp to flash when a messaging is waiting.

Operation

To retrieve MWI messages:

1. Press the flashing [CALLBK] button.
2. MWI contents (CLI number, date and time, the calling count from the same CLI) will be shown on the LCD.
3. Press the volume up/down button; the previous or next MWI is displayed, depending on which volume button is pressed.

To delete the current CLI Message and see the next one:

1. Press the [CONF] button.
2. The current MWI message will be deleted; a confirmation tone sounds, and the next MWI message will be displayed.

To place a Call Back from a MWI:

1. Retrieve the MWI by pressing the flashing [CALLBK] button, and using the volume up/down buttons.
2. Press [HOLD/SAVE] button. The System will attempt to connect a net call according to MWI data.

Conditions

- The MWI feature is available on Stations with a LCD panel only.
- When the System makes a call back according to MWI data, the CO line is selected within the Network CO group.

Admin Programming

- CLI Message Wait (PGM 114)
- Networking Basic Attribute (PGM 320)
- Networking CO Line Attribute (PGM 322)
- Networking Routing Table - Numbering Plan (PGM 324)

Net Call

Net Call allows a Station User to make a call to a Station in another Networking System by dialing a Station number just as an intercom call would be dialed within the same System.

Note: During a conversation, all telephony features are available (e.g., call transfer, hold, conference).

Operation

To make a Net Call:

1. Lift the handset or press the [SPEAKER] button; the dial tone sounds.
2. Dial a Station number on another Networking System.

   -or-

   Press the [NET DSS] button.
3. The station will access a Network CO Line according to the Net Routing Table; the System will send a digit stream that is modified by the Net Routing Table.
4. The called System will receive a digit stream that is sent by the calling system, and will analyze it using the Net Routing Table to determine the right destination Station.
5. The call will ring at the destination Station.
6. The [COL] flexible button LED representing the CO line for the Net Call, will extinguish when the Net Call is completed.

Conditions

- To make a Net Call, as an Intercom call, the Station User must dial the destination Station number without seizing a CO line.
- After the Station User makes a Net Call, the SBX IP 320 system will access an idle CO line to send the Net Call request to the destination Station. If there is no selectable idle CO line, then the User will receive an error tone.
- A received Net Call will ring in the Terminal Answer Mode, regardless the Intercom answer mode setting (Hands-free/Terminal/Privacy).
- When system detects a fatal error from the Network, the System will send the digit stream to the Network using the alternate speed dial bin. In this case, the Call is not a Networking Call.
- The Net Call is also applied to the CO Call Restriction Timer (PGM 180 - FLEX 17).

Admin Programming

- Networking Basic Attributes (PGM 320)
- Networking CO Line Attributes (PGM 322)
- Network Routing Table - Numbering Plan (PGM 324)

Net Call Forward

A Station User can forward Net Calls over the network to any networked Station, Station Group, or VMIB in the System, by activating feature codes.

There are two kinds of standard net call forward signaling methods in the H.450 protocol specification: Join and Rerouting. The SBX IP 320 System supports both methods. The main difference is how the connecting path is controlled among the forwarding, forwarded, and forwarded-to stations.

Conditions

- To use Net Call Forward, a Networked Station should be activated using ADMIN programming.
- There are several types of Net Call Forwarding: Unconditional, Busy, Busy/No Answer, and No Answer.
- To deactivate Call Forward, press the [DND/FWD] button while it is flashing.
- If both the Call Forwarded and Forwarded-to Stations are located in the same System, the networking path CO line is not needed, that is used for the forward voice path; the Forwarded Call will be initiated as an intercom call.
- At the Forwarding System, it does not check the status of the Forwarded-to Station that is in DND, CFW, or Empty, when it sets the Net Call Forward.

Admin Programming

- Call Forward Attribute (PGM 111)
- Networking Basic Attributes (PGM 320)
- Networking Supplementary Attributes - Transfer Mode (PGM 321)
- Networking CO Line Attribute (PGM 322)
- Networking Routing Table - Numbering Plan (PGM 324)

Net Call Forward - Unconditional

The Forwarded Station refers to the call originator. The Forwarded Station is rerouted to the Forwarded-to Station by a Forwarding Station (the Station forwarding the call).

Operation

To activate Net Call Forward:

1. Press the [SPEAKER] button and the [DND/FWD] button.
2. Dial the Net Call Forward code (1), and dial a Station number on another networking system. The [DND/FWD] button will flash and a confirmation tone will be provided.
3. If there is an incoming call to the net call unconditional Forwarding Station, the call is immediately routed to a Forwarded-to Station, and the Forwarded-to Station will ring.
4. When the Forwarded-to Station is answered, the Call Forwarded and Forwarded-to Station will be connected.

Net Call Forward - Busy

It is possible for a User to forward their Station remotely over the Network when it is busy.

Operation

To activate Net Call Forward, busy:

1. Press the [SPEAKER] button and the [DND/FWD] button.
2. Dial Net Call Forward code (2), and dial a Station number on another networking system.
3. The [DND/FWD] button will flash and a confirmation tone sounds.
4. If an incoming call is received at the net call busy Forwarding Station, the call is routed to the set Forwarded-to Station when the called Station is busy.
5. The Forwarded-to Station will ring, and when answered, the call will be connected.

Net Call Forward - Busy / No Answer

It is possible for a User to forward their Station remotely over the Network when it is busy or there is No Answer within the appropriate timer.

Operation

To activate Net Call Forward, busy/no answer:

1. Press the [SPEAKER] button and the [DND/FWD] button.
2. Dial Net Call Forward code (4), and dial the Station number of another networking system.
3. The [DND/FWD] button will flash and a confirmation tone should be provided.
4. If an incoming call is received at the Net Call Busy/No Answer Forwarding Station, the call is routed to a Forwarded-to Station, when the called Station is busy or does not answer.
5. The Forwarded-to Station will ring.
6. When the forwarded-to station is answered, the Call Forwarded and Forwarded-to Station will be connected.

Net Call Forward - No Answer

It is possible for a User to forward their Station remotely over the Network when there is No Answer.

Operation

To activate Net Call Forward, no answer

1. Press the [SPEAKER] button and the [DND/FWD] button.
2. Dial the Net Call Forward code (3), and dial the Station number of another networking system.
3. The [DND/FWD] button will flash and a confirmation tone should be provided.
4. If an incoming call is received at the Net No Answer Forwarding Station, the call is routed to a Forwarded-to Station, when the called Station is busy or does not answer.
5. The Forwarded-to Station will ring.
6. When the forwarded-to station is answered, the Call Forwarded and Forwarded-to Station will be connected.

Net Conference

Net Conference is generally the same as the Conference feature, with the additional specification that a Networked Station can be assigned as a conference member.

A call to a Station on one node is able to conference in a party on any other networked node. Up to 3 Stations of the network can be members of a conference.

Operation

To conduct a conference from a Station:

1. Press the [CONF] button, during a network conversation. The connected call is put on hold and the ICM dial tone sounds.
2. Make a Net Call to a Station on another Networking System.
3. Press the [CONF] button when the second Called Station is answered; the call is put on hold and an ICM dial tone sounds.
4. Press the [CONF] button again; the conference voice path to all members is connected.

To cancel a Conference:

Any Station on a net conference call can hang up during the conference. The net conference will be ended and the network path will be cleared.

Conditions

- An IP phone cannot be the master station in the Net Conference.
- When an IP phone sends a Net Conference Invite Setup message to a Non-IP phone, the message will be rejected.
- When a Non-IP phone sends a Net Conference invite to an IP phone, the normal Setup message should be sent to the IP phone instead of the Invite Setup message.

Admin Programming

- Networking Basic Attribute (PGM 320)
- Network Routing Table - Numbering Plan (PGM 324)

Net Firewall Routing

SBX IP 320 supports the networking feature regardless of a local network or a different network.

Operation

In the same network configuration:

If the networking system use the same network, then Firewall Routing should be set to OFF.

- The Local IP address will be used for networking feature.

In the different network configuration:

If the networking system use the different network, then Firewall Routing should be set to ON.

- The Firewall IP address will be used for the networking feature.

Conditions

- The Firewall Routing feature can be programmed with each net numbering plan table.
- The VOIBE should be 2.1Ck or a higher version.

Programming

- Firewall Routing (PGM 324 - FLEX 9)

Net Follow-Me Forward

A User from one System can activate a Follow-Me-Forward from a Station on another System within the Network. Once activated, all calls to the Forwarded Station will be forwarded to the designated Station over the Network. The forward can only be cancelled from the Forwarded Station.

Operation

To activate Net Call Forward- Follow-Me:

1. From the Station that the calls will be forwarded to, press the [SPEAKER] + [DND/FWD] button.
2. Dial the Follow-Me-Forward code (0) + the Station number that is to be Call-Forwarded from another System.
3. Dial the Authorization Code + the # key; the [DND/FWD] button will flash at the Forwarding Station and a confirmation tone will be provided.
4. All calls to the Forwarding Station will be routed to the Forwarded Station.

To deactivate Net Call Forward - Follow-Me

1. At the User's own Station, press the flashing [DND/FWD] button.
2. The [DND/FWD] button will extinguish, and the Net Call Forward-Follow Me will be cancelled.

Conditions

- An Authorization code must be registered to use Follow-Me Call Forward.
- Remote deactivation is not supported.
- The confirmation tone will be provided even though Follow-Me Forward may not be allowed by the other system.
- Net Follow-Me Forward is available within a VoIP networking environment only.
- Among networked systems, the same Authorization code mode
  (5 Digit /Variable Authorization code Usage) should be programmed.

Admin Programming

- Authorization Code Table (PGM 227)

Net Transfer

Net Transfer is used to transfer net calls. It can transfer any kind of call to a Station in another System by pressing the [TRANS/PGM] button and dialing the Station number. The operation of net transfer is the same as transferring a call (screened or unscreened) within the same System.

Note: There are two different kinds of standard net transfer signaling method: transfer by join and transfer by rerouting. The main difference is in the connecting path between the transferring, transferred, transferred-to Station controls.

- Join-an additional connecting path is needed to transfer the call to another Station.
- Routing-a new connecting path is used to transfer the call (the old connecting path of transferring station will be cleared).

If both transfer by join and transfer by routing are supported by the System, the appropriate mode can be selected using PGM 321.

Operation

Screened Transfer

1. Press the [TRANS/PGM] button during a conversation. The transferred call will be placed on exclusive hold.
2. Dial the Station number in another Networked System to transfer the call; the transferred-to Station of another System will receive a ring signal.
3. When the transferred-to Station answers, the voice path is connected between the transferring and the transferred-to Stations.
4. Both Stations can talk with one another; the transferred call will remain on hold.
5. Once the transferred-to Station answers, the transferred call will be connected.

Unscreened Transfer

1. Press the [TRANS/PGM] button at a Station during a conversation. The transferred call is placed on exclusive hold.
2. The Transferring Station then dials the Station number of another Networked System to transfer the call, and goes on-hook.
3. The transferring station goes idle, and the call will ring at the transferred-to Station in the other System.
4. The transferred caller and the transferred-to Station will be connected when the transferred-to Station answers the call.

Conditions

- If both transferred Stations and transferred-to Stations are located in the same system, the Networking CO line that is used for the transferring voice path is not needed. That is, the transfer call will be setup as an intercom call.
- The net transfer will be canceled when the transferring Station User presses the flashing [TRANS/PGM] button.
- After the transferring Station goes on-hook, the net transfer call will not recall to the transferring Station though the transferred-to Station.
- If all CO lines are in use, an error tone will be heard when a net transfer is attempted.
- If the call is transferred to a busy station, the busy tone will be heard.

Admin Programming

- Networking Basic Attributes (PGM 320)
- Networking Supplementary Attributes - Transfer Mode (PGM 321)
- Networking CO Line Attributes (PGM 322)
- Networking Routing Table - Numbering Plan (PGM 324)

Security of Transit-Out Code with registered IP

When the system receives a setup packet for the Transit-Out feature from the VOI CO line, the system can check the setup packet with the registered IP as to whether this packet is valid or not. If [TRANSIT-OUT SECURITY] Admin is set, the received IP in all received setup packets for the transit-out feature will be checked with the registered IP, and if this IP is verified, the transit-out feature will be invoked.

Verification is comparison with received IP in setup packet and registered IP: the system compares received IP number with all registered CPN Info IP at the NET Number table (PGM324).

If [TRANSIT-OUT SECURITY] Admin is set, and if received IP in setup packet does not match with chosen IP numbers, this call will be disconnected automatically.

Operation

To set the [Transit-Out Security] Option:

1. Press PGM 161 - FLEX 23.
2. Set ON or OFF.

Operation Step:

1. Setup packet for transit-out arrives from VOI line.
2. If the transit-out security option is set, the system searches for a registered IP in Net Number tables.
3. If received IP does not match with any registered IP in the Net Number Table, the call will be disconnected.
4. When a matched IP is found, the transit-out feature will be invoked.

Conditions

- To set the transit-out security function, set the [Transit-Out Security] Admin and set IP numbers at CPN Info in Net Number Table.
- Transit-Out Security Check is only used at a VOI CO line.

Programming

- PGM 161 - FLEX 23

VOIP Networking

Two SBX IP 320 Systems can be connected through the VOIB. The networking CO line connects the networking Systems and it is used for the network signaling message and voice path. To use the networking CO line, the service type must be set to DID/MSN at PGM 140. And to receive the net call digit information correctly, the appropriate DID conversion type must be set at PGM 143 - FLEX 4. In the following example, DID TYPE 1 satisfies the situation. To configure the net CO line connection, the net CO group and type must be set to '01' and 'NET' at PGM 322 - FLEX 1 & 4.

AT BOTH SYSTEMS A & B

PGM	RANGE	FLEX	ITEM	VALUE
140	VOIB CO range	--	CO service type	DID/MSN (3)
143	VOIB CO range	4	DID Conversion Type	1
322	VOIB CO range	1	Net CO Group	01
		4	Net CO type	NET (1)

AT SYSTEM A

PGM	BIN	FLEX	ITEM	VALUE
324	00	1	System usage	NET (0)
		2	Numbering plan code	1#**
		3	Numbering plan CO group	00
		4	VOIP Called Party Information 1/2/3/4	0.0.0.0
	01	1	System usage	NET (0)
		2	Numbering plan code	2**
		3	Numbering plan CO group	01
		4	VOIP CPN Information 1/2/3/4 (note)	VOIB IP Address of system B

Note: At the PGM - FLEX 4, a maximum 4 VOIP CPN information can be set. The reason for multiple VOIP CPN is to assign a different IP address per each VOIB, when the destination networking system has multiple VOIBs.

AT SYSTEM B

PGM	BIN	FLEX	ITEM	VALUE
324	00	1	System usage	NET (0)
		2	Numbering plan code	2#**
		3	Numbering plan CO group	00
		4	VOIP Called Party Information 1/2/3/4	0.0.0.0
	01	1	System usage	NET (0)
		2	Numbering plan code	1**
		3	Numbering plan CO group	01
		4	VOIP CPN Information 1/2/3/4 (note)	VOIB IP Address of system A

Operation

To enable the networking feature:

> The PGM 320 - FLEX 1 must be set to ON at each System (if the networking lock-key is not installed, this ADMIN program cannot be enabled).

To satisfy the UDP condition:

> Flexible Station Numbering Plan (PGM 105) must be programmed at each System. At this ADMIN program, the Station number range of System A can be changed from 100 to 199 (100-199), and System B also can be changed from 200 to 299 (200-299).

Note: The changed Station numbers must not conflict with the flexible numbering plan code for each System.

If system A has a flexible numbering code 2 as its Pick-Up code, then Station numbers that start with the digit 2, cannot be used.

The destination IP address is needed to route:

PGM 324 - FLEX 4 is used to set the IP address of the destination System's VOIB.

Configure the VOIP Networking between System A & B as listed:

At system A: Enter PGM 105 + [SPEED] + 100 + 199 + [HOLD]

At system B: Enter PGM 105 + [SPEED] + 200 + 299 + [HOLD]

Admin Programming

- CO Service Type (PGM 140)
- ISDN CO Line Attributes - DID Conversion Type (PGM 143)
- DID Receive Digit and DID Digit Mask (PGM 146)
- Flexible DID Table (PGM 231)
- Networking CO Line Attributes - Group and CO Type (PGM 322)

IP Phone Reroute Service

SBX IP 320 provides a rerouting service when IP phones (IP hard-phone and Nomad SP) are not available (line disconnected) for any reason, the call can be routed to his voice mailbox or mobile extension.

Conditions

- A Hunt group call is not allowed this reroute service.
- Only IP phones can use this reroute feature.

Admin Programming

- Auto Forward to VMIB (PGM 113 - FLEX 2 & FLEX 14)
- Forward to VMIB Timer (PGM 181 - FLEX 20)
- Mobile Extension (PGM 236)

ISDN Service

The SBX IP 320 system does not support Basic Rate Interface (BRI) Integrated Services Digital Network (ISDN) circuits. The BRI provides two bearer channels and one data channel (2B+D), but for the Calling Line ID function, the same admin fields are used as ISDN.

Calling Number and Called Number services are supported. Calling Number services will be routed in the same way as ANI (automatic number identification) calls using the DID route table.

The rules and conditions of ANI are the same, and still apply to Calling Number service on ISDN lines. Called Number services will be routed using the DID route table. The rules and conditions of DID and still apply to ISDN Called Number service on ISDN lines.

- The bearer channels (B channels) transport voice information to and from the Central Office.
- The data channel (D channel) controls all signaling information for the bearer channels.

Calling Line Identification Presentation (CLI)

CLI is the telephone number of the caller. By using this, the SBX IP 320 station user can recognize the incoming CO caller's information and send the telephone number of the caller when he makes an outgoing CO call.

Incoming CLI Service

When a station of the SBX IP 320 system receives an incoming CO call that has a telephone number of the caller, the station user can see the telephone number of incoming CO caller on the LCD.

Operation

In the SBX IP 320 system, the various operations about the incoming CLI are provided according to the ADMIN program setting.

- If the CLIP DISPLAY (PGM 114 - FLEX 1) is set to ON, the incoming CLI is displayed on the station LCD.
- If the CLI NAME DISPLAY (PGM 114 - FLEX 11) is set to ON, the incoming CLI digit display can be replaced with the matched station speed dial data name.
- If the CLI print (PGM 200 - FLEX 6) is set to ON, the incoming CLI can be printed to an RS-232C port.
- If the MY AREA CODE (PGM 200 - FLEX 9) is set, and this value matches the start digits of the incoming CLI, then the matched digits are removed automatically.
- If the MY AREA PREFIX CODE (PGM 200 - FLEX 10) is set, it can be used in conjunction with the preceding bullet "MY AREA CODE".
- If the CLI MESSAGE WAIT (PGM 114 - FLEX 4) is set to ON, the unanswered CLI data will be saved in station memory.

Conditions

- Up to 12 digits will be displayed on DKTU as a CLI number.
- Though the power of system is off, the stored CLI messages are not erased.
- The CLI can be shown at SLT which has a CLI display LCD in the SBX IP 320.

Admin Programming

- CLIP LCD Display (PGM 114 - FLEX 1)
- CLI Name Display (PGM 114 - FLEX 11)
- CLI Message Wait (PGM 114 - FLEX 4)
- CLI Print (PGM 200 - FLEX 3)
- My Area Code (PGM 200 - FLEX 9)
- My Area Prefix Code (PGM 200 - FLEX 10)

Outgoing CLI Service

When a station of the SBX IP 320 system makes an outgoing CO call, it can send the telephone number.

Operation

- In the SBX IP 320 system, if the Call Type setting (PGM 143 - FLEX 3) is set to NATIONAL, the outgoing CLI is generated as a National Number, according to the ADMIN program setting.
- If the ISDN CLI of STATION (PGM 114 - FLEX 12) is set, this value is used as the end part of CLI when the outgoing CLI is generated.
- If the MY AREA CODE (PGM 200 - FLEX 9) is set, and this value is used as the first part of CLI when the outgoing CLI is generated.
- If the entry of CLIP/COLP TABLE (PGM 201) is set, and if the CLIP TABLE INDEX (PGM 143 - FLEX 2) is set to the front entry number, then it can be used as the middle part of CLI when the outgoing CLI is generated.
- If the CLI RESTRICTION (PGM 114 - FLEX 14) is set to ON, CLI transmission is restricted.

Condition

- Though the System power is OFF, the stored CLI messages are not erased.

Admin Programming

- CLIP/COLP Table (PGM 201)
- CLIP Table Index (PGM 143 - FLEX 2)
- Call Type (PGM 143 - FLEX 3)
- ISDN CLI of STATION (PGM 114 - FLEX 12)
- ISDN CLIR (PGM 114 - FLEX 14)
- My Area Code (PGM 200 - FLEX 9)
- My Area Prefix Code (PGM 200 - FLEX 10)

CLI Message Wait

When a call exists through an ISDN DID line, calling line identification (CLI) of the incoming call will be displayed on the LCD of the station. If the DID external party hangs up before an Attendant or called station answers, the CLI provided by digital network will be stored in the CO message wait queue of the original called station.

Operation

CLI MESSAGE WAIT FEATURE SUMMARY

BUTTONS	CLI ACTIONS
[VOL UP/DOWN]	See next CLI message
[HOLD]	Make a recall according to CLI message
[CALLBK]	Toggle CLI message and SPEED Name
[SPEED] + Bin # + [CONF]	Store CLI message in SPEED dial bin
[CONF]	Delete current CLI message
[DND]	Delete all CLI messages
At ATD STA: [TRANS/PGM]+055	Delete all CLI messages at ATD station

To activate CLI Message Wait, use the following procedure:

1. CLIP should be programmed as ON (1).
2. The message contents will be shown on LCD.

```
MSG: CLI(#)
```

To retrieve a CLI Message:

1. Press the [CALLBK] button.
2. The message contents (CLI number, date and time, the calling count from the same CLI) will be shown on the LCD. According to admin date LCD display format, the DATE will be display as MM/DD or DD/MM. TIME is always displayed in 24-hour mode.

```
03434507902
DATE TIME CNTxx
```

To delete the current CLI Message and see the next one:

1. Press the [CONF] button.
2. The Station User can see the next CLI message and the current CLI message will be deleted.

To delete all CLI messages:

1. Press the [CALLBK] button.
2. Press the [DND] button and [HOLD] button.

To initiate a call back:

1. Press the [HOLD] button.
2. The System will secure an available CO line in the first accessible CO group and dial it similar to speed dialing.

To see the next or previous CLI message:

Press the [UP/DOWN] keys.

To delete all CLI messages at Attendant:

1. Press the [TRANS/PGM] button.
2. Dial 055.
3. Dial the Station range.
4. Press [HOLD] button.

To program the SPEED BIN number with CLI message:

1. At the retrieve CLI message LCD, press the [SPEED] button. The LCD display message changes as a usual station speed dial program, except that it displays the CLI message using an indication.

```
ENTER SPEED BIN NO(001)
CLI MSG USED
```

2. Press a Station speed bin number.

   -or-

   Press the [HOLD/SAVE] button.

3. After entering the speed bin number, a CLI message using indication is displayed onto speed dial digit position. It means that CLI can be entered pressing [CONF] button. If you want to assign the CO for speed, it must be prior to pressing the [CONF] button.

```
CLI MSG USED
ENTER CO-BTN/DIGIT(001)
```

4. Press the [HOLD/SAVE] button to store the speed dial number.

Conditions

- The CLI display works with about any type of CO service, if CO line is ISDN. But CLI message wait only works with the DID/MSN CO service type of the ISDN line.
- The total number of CLI message wait is 500 (on a System basis).
- A Station without an LCD can not receive CLI message wait even though CLIP is set to ON.
- CLI messages are saved against power failure.
- When the call is routed to a Ring Group, the CLI Message Wait pertains only to the first member of the Ring Group.
- When the call is routed to a member of a UCD/ Circular/ Terminal station group, the CLI Message Wait will be provided to the first ringing station.
- If the external party releases the line during the VMIB announcement, the CLI Message Wait is not saved to any stations. If the external party releases the line during that time that a station is ringing after the VMIB announcement, the CLI Message Wait will remain at the first ringing station.
- Though the call is routed to station (A) that is forwarded to the another station (B), the CLI Message Wait is provided to the original station (A). The basic rule for CLI Message Wait follows the rule of the "Message Wait" feature.
- If there is no remaining buffer, the following warning message will be printed out through RS-232C "WARNING: CLI MESSAGE WAITING BUFFER FULL".
- If the CLI number is programmed in the SPEED BIN No. Table with the "name" and the CLI name display ADMIN programmed, then the name will be displayed in the LCD. The CLI number and CLI user name is toggled by pressing the [CALLBK] button.
- When a user tries to delete all CLI messages at his station and some VMIB voice message wait exist together with CLI message, the "all CLI delete" feature cannot be activated. Because the priority of the message wait is as follows:

  VMIB message wait ->CLI message wait -> VM Group message wait

- If the duplicated CLI messages are left at a station, the LCD of the station will display the CLI message with the CLI duplicated counter (max 15) and the latest message left time.
- If CLI print admin program is set, CLI and Station Number are printed through the RS-232C port.

CLI Transit

When a user tries to make an outgoing call, outgoing CLI can be different in certain situations.

Outgoing CLI can be selected automatically, according to the CLI Transit Option (PGM 143 - FLEX 7) and authorization code.

There are two kinds of options in CLI Transit Type: Original type and Forward type.

Operation

Outgoing Call through DISA Line

When the system receives a DISA incoming call with CLI and then an Outgoing call is made, outgoing CLI is selected depending on the situation.

Type of CLI for DISA call	DISA Account	Type of entered password	CLI provided to PSTN
CFW	ON	Personal Password	CLI of Station which password is attached to
CFW	ON	System Password	CLI of Attendant
CFW	OFF	Without Authorization	CLI of Attendant
ORI	It does not matter	does not matter	Original CLI of external caller (if it is received)
ORI	It does not matter	does not matter	Original CLI of external caller (if it is received)
ORI	It does not matter	does not matter	Original CLI of external caller (if it is received)

Walking COS

- CO Account feature (PGM 141 Btn 9)
- Forced Station Account feature (PGM 112 Btn 20)
- LCR Password feature (PGM 221 Btn 6)

Dialed Password	CLI provided to PSTN
System password	CLI of Attendant
Personal password	CLI of Station which password is attached to

DID Destination is assigned to SPEED Bin

Incoming CO Off-Net CFWD.

Type of CLI	CLI provided to PSTN
CFW	CLI of Attendant
ORI	Original CLI of external caller (if it is received)

Admin Programming

- PGM 143 – FLEX 7

Calling Party Number Service

The system provides an option to display the Calling Party Number (CPN) for an incoming ISDN call which has two CPN information elements in the Setup message. Usually one CPN is real CPN of the line and the other is a transit CPN of the network.

Operation

If PGM 143 - FLEX 15 value is 1, Real CPN is displayed. If the value is 0, Transit Point CPN is displayed.

Admin Programming

- PGM 143 - Flex 15 : ISDN CLI choice.

Linked Stations

Executive/Secretary Pairs

Stations in the system can be assigned as Executive and Secretary pairs. When an Executive station is busy or in DND, intercom calls and transfer calls are automatically routed to the designated secretary. The maximum number of Executive/Secretary pairs is 6.

- If the executive station is idle, the executive station will receive the ring of the CO call.
- If the executive station is busy or in DND, the designated secretary station will receive the ring of the CO call.

Operation

To activate Executive/Secretary Transfer from the Executive's DKTU:

Press the [DND/FWD] button.

To activate the Executive station's DND feature from the Secretary's DKTU:

Press PGM + Flex Button + PGM + 55 + Executive's extension number

Conditions

- A Secretary can pass a call to the Executive when in DND state by using the camp-on feature. One Executive can have multiple Secretaries within the maximum pairs, and one secretary can be assigned to multiple executives within the maximum pairs.
- When the executive is in DND, the secretary can transfer a CO line call or make a camp-on.
- It is possible to make a chain to assign Executive/Secretary pairs. This means that a Secretary may be an Executive of another Executive/Secretary pair. If an Executive and the Secretary which has own Secretary are busy, a call will be forwarded to the second Secretary of Executive/Secretary chain (it cannot be a loop chain).
- If an Executive has multiple secretaries and the first secretary is busy, a call will be forwarded to the next Secretary.
- If an Executive has multiple secretaries, a secretary can forward a call to another secretary. The secretary cannot forward a call to the Executive.
- If an Executive makes a call forward to a non-secretary station, a call to the Executive will be routed to assigned station.

- When both Executive and Secretary are busy, camp-on / transferred calls / messages remain at the last Secretary station in the chain.

Admin Programming

- Do Not Disturb (PGM 111 - FLEX 3)
- Executive/Secretary Table (PGM 229)
- CO Call To SEC (PGM 229 - FLEX 2)
- Call EXEC If SEC DND (PGM 229 - FLEX 3)
- EXEC grade (PGM 229 - FLEX 4)

Linked-Pair Station

Two stations can be linked with each other by programming. Linking with a DKTU and another station, the user can use them alternatively. When two stations are linked, the following functions are supported.

If two linked stations in a Linked-Pair are idle and a CO call arrives, both linked stations ring. If one linked station in a Linked-Pair is busy and a new CO call arrives, the caller will hear the busy tone.

If one of two linked stations receives intercom calls/CO incoming calls (DISA/DID)/recall ring (system/exclusive hold / transfer), then the other linked station will receive ring together. If one station of linked pair goes to DND, call forward, or pre-selected message display state, then the other linked station goes to the same state automatically. Also, if any station in a linked pair comes out of this state, then the other comes out simultaneously.

If one of linked stations is busy, the LCD of the other station will display "IN USE AT LINK STA". When a linked station is busy, the other idle linked station will not receive ring for CO lines, transferred ring, or intercom calls.

Conditions

- The system will support 13 linked station pairs.
- A station can be linked with only one station.
- The intercom number of two linked stations is operated as one number for all features.
- The presented number of a linked pair is the first station number (Master), which is assigned by Admin Programming.

- The station attributes of the second station (Slave) will follow the attributes of the Master's. (Ex. Day/Night COS, CO Warning Tone, CO Auto Hold, CO Call Drop, Alarm)
- The Doorbox, DSS/BLF, or ISDN phone cannot be linked with a station.
- Linked station pairs are operated in Tone mode regardless of intercom Answer mode (STA program 12).
- The Attendant station cannot be linked with another station.
- A linked station can call his pair station by dialing their own number. It is possible to make CO line/Intercom Transfer between two pairs.
- A Port blocked station cannot be linked with any other station.

Admin Programming

- Linked Station Pairs (PGM 179)

Outgoing Call Access

Basic Access

Each station is allowed or denied access based on particular CO Lines or CO Groups. Station users may use Flexible buttons which are assigned as a {CO} button or {CO Group} button, including the {POOL} and {LOOP} buttons. According to the Numbering Plan, station users can access individual CO Lines by dialing CO Access Codes.

FEATURE	DESCRIPTION	OPERATION METHOD	ACCESS CODE
Idle Line Access (88 + CO Line number)	Automatically selects an idle CO Line from the assigned CO Groups	Dial the idle Line Access Number (9), or press a CO Line button.	8801-8816
CO Group Access (8 + CO Group number)	Selects an idle CO Line from the corresponding CO Group.	Dial the CO Group Access number and a CO Group number, or press a CO Group button.	801-808

You can dial 9 to access the first idle line in their CO group.

You can dial 8801 to access CO Line 001 if it is idle.

You can dial 801 to access the first idle CO Line in CO Group 1.

Operation

To access a CO Line from a DKTU:

1. Lift the handset or press the [SPEAKER] button.
2. Press the desired CO Line, {POOL} button, or {LOOP} button.

   -or-

   Dial the individual CO Line Access Code, CO Group Access Code, or the first CO Line Access Code from the accessible group.

To access a CO Line from an SLT:

1. Lift the handset.
2. Dial the individual CO Line, CO Group Access Code, or the first CO Line Access Code from the accessible group.

To access a CO Line Group:

1. Lift the handset.
2. Press 8 for the CO Group.
3. Dial the CO Group number.

To assign the {LOOP} button:

1. Press the [TRANS/PGM] button.
2. Press {FLEX}.
3. Press [TRANS/PGM].
4. Dial 84.
5. Press the [HOLD/SAVE] button to accept changes.

Conditions

- A DKTU should have an idle appearance (CO Line/Pool button/Loop button) to access an incoming/outgoing CO Line.
- When the Override 1st CO Line Group is enabled, the System will search for the next accessible CO group until a CO Line is available, if there is no available CO Line by dialing the CO Line group access code (9 or 0).
- An error tone sounds when a Station is not permitted to access a CO Line, but the Station will still be able to receive a transferred CO Line call as applicable.
- The CO Line choice (Round-robin or Last Choice) is determined by Admin Programming (PGM 160 - FLEX 3)
- If the CO Line is BRI, when a user tries to secure B1 the System can change the CO Line to B2 (if there is a {B2 CO} button or {LOOP} button).
- Unused CO Lines should be assigned to unused CO Group 9 to prevent being accessed by a station.
- The first CO Line group (00) is the directed line group and can be used with the {CO Line} button (Private Line).

Admin Programming

- CO Line Choice (PGM 160 - FLEX 3)
- Inter-digit Timer (PGM 181 - FLEX 8)
- CO Line Group Access (PGM 117)
- CO Line Group (PGM 141 - FLEX 1)
- Override 1st CO Line Group (PGM 161 - FLEX 3)

Call Time Restriction

The Call Time Restriction feature is used to restrict outgoing CO call time. In Station programming, the User can set the Call Cut Off timer, whereas the call will be disconnected automatically when the timer expires. The called and called parties will hear a warning tone 15 seconds before the call is disconnected.

Conditions

- This feature can be assigned on a station-by-station basis, and is applied to just outgoing CO calls.
- If the Call Cut Off timer is enabled on a Station, the timer is still applicable when a call is transferred to another Station.
- On the add-on conference, the Call Cut Off timer-enabled Station will be restricted to the outgoing CO call time.
- The Call Cut Off timer is not released when the call is placed on hold or is transferred.

Admin Programming

- CO Call Time Restriction (PGM 112 - FLEX 3)
- Call Cut-Off Timer (PGM 113 - FLEX 12)

CO Line Queuing

When a Station user receives a busy tone during an attempt to access a CO Line, the user may request a call back (queue call). The station will receive a call back when the busy CO Line becomes available.

Operation

To activate CO Line Queuing while receiving a busy tone:

1. Press and release the hook-switch if the station is an SLT.
2. Dial 556 or press the [CALL BK] button.
3. When the confirmation tone sounds, replace the handset.
4. Once the CO Line becomes idle, the call back ring will be received at the station.
5. Lift the handset. The CO dial tone should be heard to make a call.

Conditions

- A CO Line may have any number of queues at one time.
- When the queued CO Line becomes idle or a CO Line becomes available in the group, the oldest queued station will receive the call back.
- A station can make only one CO Line queuing request at a time. If the station tries to make another CO Line queuing, the previous one is canceled and the newer one is activated.
- If the waiting station is busy and the queued CO Line is available, the available CO Line will be directed to the next queued idle station.
- If the waiting station is idle, the queued CO Line will give a call back signal to the station for 15 seconds. If the signal is not received at the station, the queue is canceled and the next station in the queue will receive the signal.

Admin Programming

- CO Line Queuing (PGM 112 - FLEX 5)

CO Step Call - Analog Only

When an analog Station receives a busy tone after accessing a CO Line, you can dial a CO Line number which has the same first digits as the called busy CO Line without dialing the full number.

Operation

To use CO Step Call when receiving a busy tone, use the following procedure:

1. Press the [SPEED] button and dial the last digit of the previously called number.
2. The previous call is terminated and a new call is established.

Emergency Call Service

You can dial the Emergency Service Code regardless of lower station COS.

Conditions

- An emergency call can be dialed by pressing an available CO Line at the station that is assigned to COS 7.
- If the dialed number for the Emergency Service Code is the same as a station number on the system, or LCR number, the call is operated as an Emergency Call. The preference of the programmed dial number that is sent to external CO Line is:

  Emergency Call Code>LCR Table>Station Number

Admin Programming

- Emergency Service Call (PGM 226)

Hot Line & Warm Line

A station user can instantly make an outgoing call by lifting the handset or pressing the [ICM], if the user has previously stored the destination.

The destination can be a CO Line or CO Line Group; the function can be setup on a Flexible Button, or at another Station.

Hot Line can be activated immediately when the Station is in the off-hook state; Warm Line can be activated after the Warm Line Timer has expired. If the user dials another number prior to the Warm Line Timer expiration, the call will activate as a normal call, not as a Warm Line call.

Operation

To activate a Hot Line:

Lift the handset at a station where Hot Line is assigned. The assigned Hot Line feature is immediately activated.

To activate Warm Line:

Lift the handset at a station where Warm Line is assigned. The assigned Warm Line feature is activated if no dialing has been done while the Warm Line Timer is running.

Conditions

- A station can be assigned Hot Line or Warm Line with Admin Programming (PGM 113 - FLEX 7).
- If there is no Flexible Button at the station, the number is operated as a Speed Dial Number.
- The set value of the Warm Line Timer should be less than that of the Dial Tone Timer.
- A Flexible Button may be assigned as an Idle Line Selection button.
- When lifting the handset or pressing the [SPEAKER] button, the system will be activated as a predefined button if pressed.
- It is possible to activate Hot Line/Warm Line at an SLT station.

Admin Programming

- Warm Line Timer (PGM 182 - FLEX 8)
- Warm Line (PGM 113 - FLEX 7)
- Idle Line Selection (PGM 122)

Least Cost Routing (LCR)

LCR is a system programmable feature that automatically selects the least expensive available route when an outgoing CO call is made. This programming eliminates the necessity for the user to dial the access code of the least expensive carrier. There are three ways to activate LCR:

Internal LCR - If dialed digits match an internal LCR code, the system secures a CO Line from the programmed CO Group and sends the modified digits according to LCR programming.

Loop LCR - When dialing the first accessible CO Group Code (9 or 0), or pressing the [Loop] button, if the dialed digits match a COL LCR code, the system will secure a CO Line from the programmed CO Group and send the modified digits according to LCR programming.

Direct CO LCR - After dialing a CO Line or CO group code (9 or 0, depending on the nation you are calling from), or pressing a CO Line or CO group button, LCR can be activated. If the dialed digits match a COL LCR code, the system will secure a CO Line from the programmed CO group and send the modified digits according to LCR programming.

When a user selects a CO Line and dials a destination number, the system checks the LCR programming and sends the call according to the least cost route according to the ADMIN program.

Operation

To activate Internal LCR:

Dial the internal LCR Code after lifting the handset or pressing the [SPEAKER] button (on-hook dialing can also activate LCR). It is an internal LCR Code if the code is programmed with internal or both in the Leading Digit Table.

To activate Loop LCR:

1. Dial COL LCR code after dialing the first accessible CO Line or CO Group access code (0 or 9), or press the [Loop] button.
2. It is a COL LCR code if the code is programmed with COL or BOTH in the Leading Digit Table.

To activate Direct CO LCR:

Dial the COL LCR code after dialing a CO or CO Group Access code, or press a CO or CO Group button. It is a COL LCR code if the code is programmed with COL or BOTH in the Leading Digit Table.

Ex. 1 Add Prefix Digit - The long distance call access code starts with 0 (e.g., 02, 031, 051). If a cheaper carrier exists, the user can access it with the carrier access code 082 and the long distance access code without 0.

The SBX IP 320 system administrator wants to use this cheaper carrier for all long distance calls (e.g., dial 0314502628, 082314504628).

PGM 220	PGM 221 (LDT)	PGM 222 (DMT)
LCR MODE MO1, M02, M11, M12 (Loop LCR enabled)	Bin 000 LCR TYPE: COL LCR CODE: 0 DMT: 00 00 00	Bin 00 Remove Position: 01 Remove Number: 01 Add Position: 01 Add Digit: 082

Ex. 2 Select CO group - The SBX IP 320 system is connected with two carriers (one carrier is carrier A, the other carrier is carrier B). Carrier B is used for international calls, and Carrier A is used for all other calls. The international call access code is 001.

The SBX IP 320 system administrator wants to program Carrier B to be used for only international calls.

PGM 141	PGM 117	PGM 161-3
Set CO Lines from carrier "A" to CO group 1. Set CO Lines from carrier "B" to CO group 2.	Enable access CO group 01, 02	Override 1st CO Group: OFF

PGM 220	PGM 221 (LDT)	PGM 222 (DMT)
LCR MODE MO1, M02, M11, M12 (Loop LCR enabled)	Bin 000 LCR TYPE: COL LCR CODE: 001 DMT: 00 00 00	Bin 00 CO Group: 02

Ex. 3 Make another CO Access Code - The SBX IP 320 system has VOIP CO Lines and normal CO Lines; system administrator wants to access VOIP CO using code 7, and access normal CO Lines using code 9 (as the default CO Access Code).

PGM 106
Remove/change numbering plan which starts with "7".

PGM 141	PGM 117	PGM 161-3
Set normal CO Lines to CO group 1 Set VOIP CO Lines to CO group 2	Enable access CO group 01, 02	Override 1st CO Group: OFF

PGM 220	PGM 221 (LDT)	PGM 222 (DMT)
LCR MODE M02, M12, M13 (Internal LCR enabled)	Bin 000 LCR TYPE: INT LCR CODE: 7 DMT: 00 00 00	Bin 00 Remove Position: 01 Remove Number: 01 CO Group: 02

Ex. 4 Password for specific dial number - The international access code is 001; system administrator allows international calls by only those users who know the system password.

PGM 220	PGM 221 (LDT)	PGM 222 (DMT)
LCR MODE M12, M13 (LOOP/CO LCR enabled)	Bin 000 LCR TYPE: COL LCR CODE: 001 DMT: 00 00 00 Check Password: ON	Bin 00 CO Group: 01

Conditions

- There are 6 LCR modes. The mode is determined by PGM 220 - FLEX 1.

  - LCR Access Mode 00 (M00): LCR call is disabled.
- The leading digits can be duplicated. FLEX 2 and the DMT index make each entry unique.

- The Leading Digit table is sorted by leading digits, FLEX 2 in LDT (INT, COL, BOTH) and DMT index.
- Internal LCR is applied if the dialed digits are matched with one of leading digits and FLEX 2 is INT or BOTH.
- Loop LCR is applied if the dialed digits are matched with one of leading digits and FLEX 2 is COL or BOTH.
- Direct CO LCR is applied if the dialed digits are matched with one of the leading digits and FLEX 2 is COL or BOTH, and the secured CO Line belongs to the programmed CO Group in DMT.
- To work Loop LCR and Direct CO LCR differently with the same leading digits, there should be a leading digit entry for loop LCR prior to the leading digits for direct CO LCR. It is possible if the DMT index for loop LCR is smaller than the DMT index for direct CO LCR.
- While direct CO LCR is applied to ISDN CO, an ISDN Information message with called party IE, which includes only the numbering plan and numbering type, is sent to the network when a user dials a digit. It is for the network not to disconnect the line.
- For direct CO LCR, leading digits should be programmed in consideration with the dial tone time provided by the Network.
- Direct CO LCR does not use an alternative DMT index if a CO Line is already accessed.
- LCR always has the higher precedence than the flexible numbering plan table.
- LCR can be applied in the following instances:
    - Dialing after accessing a CO Line by dialing a CO Line access code (0 or 0) only.
    - Dialing after accessing a CO Line by pressing the {LOOP} button.
    - Dialing without accessing a CO Line.
    - Speed Dial.
    - Off-net Call Forward
    - Redial (if the previous call is LCR applied).
    - ACNR (If the call is LCR applied when activating ACNR)
- Any leading digit string at the LDT table can be a sub-string of another leading digit string such as 012 and 0123.
- Capacity for LCR Table:
    - 3 Day Zones
    - 3 Time Zones

- Number of 'Dialed Code Bins': 250 bins
- Number of 'Modification Code Bins': 100 bins
- Maximum number of 'Dialed digits': 12 digits
- Maximum number of 'Added digits': 25 digits
- Alternative DMT index: 1 ea

Admin Programming

- LCR Attributes (PGM 220)
- Leading Digit Table (PGM 221)
- Digit Modification Table (PGM 222)
- LCR Table Initialization (PGM 223)

Memory Dialing

Auto Call Number Redial (ANCR)

If Call destination is busy or no answer, redialing is operated repeatedly within the ACNR retry counter. The system will retry the number based on programming with appropriate pauses in between dialing (default = 3 times).

Operation

To use ACNR while receiving a busy/no answer indication on a CO Line, use the following procedure:

1. Press the [REDIAL] button.
2. Replace the handset or go on-hook.
3. The system will automatically retry the call at programmed intervals.
4. When the called party answers, lift handset.

-or-

Press the [MUTE] button to make the call.

To cancel ACNR:

Press the flashing [REDIAL] button.

-or

Lift the handset

or

Press the [MUTE] button while a CO Line is accessed to cancel ACNR.

Conditions

- A DKTU that doesn't have a [REDIAL] button, should be programmed with a [REDIAL] Flexible Button to use ACNR:
- [TRANS/PGM] + FLEX + [TRANS/PGM] + 97 + [HOLD/SAVE] (2 or 8 Button only)
- The analog CO Lines in the system should be equipped with Call Progress Tone detection Units (CPTU).
- When a predefined CO Line is busy in ACNR mode, an available CO Line in the same group will be secured.

Admin Programming

- ACNR Pause Timer (PGM 180 - FLEX 10)
- ACNR Delay Timer (PGM 180 - FLEX 8)
- ACNR Tone Detect Timer (PGM 180 - FLEX 13) - applicable to analog CO Line only
- ACNR No Answer Timer (PGM 180 - FLEX 9)
- ACNR Retry Counter (PGM 180 - FLEX 11)
- ACNR Tone Cadence (PGM 423)

Last Number Redialing

The last dialed number on a CO Line can be stored (up to 32digits) in the station's Last Number Redial buffer. The user may select to redial the last number dialed on the system. Each DKTU with an LCD panel in the system has 10 individual last dialed number directory locations.

Operation

To use Last Number Redial at a DKTU:

1. Lift the handset or press the [SPEAKER] button.
2. Press the [REDIAL] button.

-or

1. Press the [SPEED] button and dial *.
2. Press the [HOLD/SAVE] button to accept.

To use one of the recently dialed numbers in the Last Number Directory by scrolling at a DKTU with an LCD panel:

1. When the last dialed number is displayed, press the [UP] or [DOWN] button to find the desired phone number (up to 10 last dialed numbers can be stored in the directory).
2. To make a call, press the [HOLD/SAVE] button when the phone number is displayed.

To use Last Number Redial at an SLT:

1. Lift the handset.
2. Dial 552.

   -or-

   Press the [REDIAL] button.

Conditions

- When the used CO Line is busy, an idle CO Line in the group is accessed and the last dialed number is dialed.
- The last dialed number directory allows duplicated phone numbers.
- If you use Last Number Redial while the Auto-redial is activated, the auto-redial is canceled.

Save Number Redialing

Any dialed number can be saved temporarily and used at any time. This number is saved until a new number is stored.

Operation

To save a number in the Save Number Redial buffer from a DKTU:

1. Press the [SPEED] button twice, while on a conversation with an external party.
2. Replace the handset or go on-hook.

To dial a number from the Save Number Redial buffer from a DKTU:

1. Lift the handset or press the [SPEAKER] button.
2. Press the [SPEED] button.
3. Press the # button.

Conditions

- When the used CO Line is busy, an idle CO Line in the group is accessed and the saved number is dialed.
- The stored save number is not deleted when the system power is OFF.
- If you press the [SPEED] button twice after accessing a CO Line and dialing, then pause, the save number redial bin will be erased.

Station Speed Dialing

A DKTU user can store up to 100 frequently used station numbers to Station Speed Bins (000-099). Station numbers can consist of up to 24 digits including pauses, Flash commands, pulse-to-tone switchover, and no-display characters (pause is automatically inserted after a flash).

Operation

To make a call using Station Speed Dial from a DKTU:

1. Lift the handset or press the [SPEAKER] button.
2. Press the [SPEED] button.
3. Dial the Station Speed Dial bin (000-099).

To store Station Speed Dial numbers from a DKTU:

1. Press the [TRANS/PGM] button.
2. Press the [SPEED] button.
3. Dial the Station Speed Dial bin (000-099).
4. If desired, press the CO Line or Group button.
5. Dial the desired telephone number, including these special codes:
   - [CALLBK] - Insert Pause
   - key - If stored as the first digit, its function is Display Security. Otherwise, its function is Pulse to DTMF Switchover.
   - [DND/FWD] - If CO Dial Tone Detect (refer to Ref. A) is ON and it is stored as the first or second digit, and the accessed CO Line is behind the PBX mode, its function is Dial Tone Detect. Otherwise, its function is Pause.
   - [FLASH] - Inserts a Flash into the speed number.
   - If the accessed CO Line is analog, its function is Flash to PX (or PBX).
   - If the accessed CO Line is ISDN line (refer to Ref. B) and it is stored as the first digit, it makes the remaining digits sent with envelope information not in the Calling Party Number IE but in keypad facility IE.
6. Press the [HOLD/SAVE] button.
7. If desired, enter the name (Max. 12 characters) using the 2-digit code for each character.
8. Press the [HOLD/SAVE] button.
9. To store continuously, repeat this procedure from Step 3.

To delete a Station Speed Dial bin:

1. Press the [TRANS/PGM] button.
2. Press the [SPEED] button.
3. Dial the Speed Dial bin number to be erased.
4. Press the [HOLD/SAVE] button. The stored Speed Dial number is erased from the speed bin.

To display and enter a Speed Dial bin by scrolling:

1. Press the [TRANS/PGM] button.
2. Press the [SPEED] button.
3. Dial the Speed Dial bin number.
4. Press the [UP] or [DOWN] key to display the next/previous Speed Dial number.

To store Station Speed Dial numbers from an SLT:

1. Lift the handset.
2. Dial the Speed Dial program code 555.
3. Dial the Speed Dial bin number (000-099).
4. Dial the desired phone number (up to 24 digits).
5. Press and release the hook-switch.

To make a call using Station Speed Dial from an SLT:

1. Lift the handset.
2. Dial the Speed Dial access code 558.
3. Dial the Station Speed dial bin (000-099).

To delete a Station Speed Dial from an SLT:

1. Lift the handset.
2. Dial the Speed Dial access code 555.
3. Dial the appropriate Station Speed Dial bin (000-099).
4. Press and release the hook-switch.

Conditions

- A CPTU should be installed to detect a dial tone.
- The Station Speed Dial is secured in data protect mode when the power is OFF.
- There can be a maximum of 24 digits in a station Speed Dial number including special digits and function codes.
- If you dial an empty station Speed dial bin, an error tone will be heard.
- If you select a CO Line before dialing a Speed Dial bin number, the selected CO Line is used even though there is a programmed CO Line in the Speed Dial bin number.

- You can program a station name (max. 12 characters) in the DKTU to be presented instead of a station number. The name is programmed in the Speed Dial bin 000. When the station name is programmed, the speed bin is not used as a Station Speed Dial bin.

Admin Programming

- Speed Dial Access (PGM 112 - FLEX 9)
- CO Dial Tone Detect (PGM 160 - FLEX 6)

System Speed Dialing

The System Speed Dial bins are programmed by the System Attendant. These numbers are available for easy access by all stations allowed in the system. The maximum System Speed Dial capacity is 500.

SYSTEM SPEED DIAL	
SPEED DIAL RANGE	**DESCRIPTION**
2000-2199	Unrestricted
2200-2499	Restricted by Station COS

SYSTEM COS	
COS 1	Unrestricted
COS 2	Monitored by Exception Table A
COS 3	Monitored by Exception Table B
COS 4	Monitored by Exception Table A & B
COS 5	Long distance calls are not allowed; the dialed digits can be longer than 7 digits.
COS 6	Long distance calls are not allowed; only a max of 7 digits may be dialed.
COS 7	Only intercom, paging, and emergency calls are allowed; no dialing allowed on CO Lines.

Operation

To store a number in a System Speed Dial from the System Attendant:

1. Press the [TRANS/PGM] button.
2. Press the [SPEED] button.
3. Dial the System Speed Dial bin.
4. If desired, press the CO Line or Group button.
5. Dial the desired phone number and include these special codes (up to 24 digits).
    - [CALLBK] - Insert Pause
    - key - If stored as the first digit, its function is Display Security. Otherwise, its function is Pulse to DTMF Switchover.
    - [DND/FWD] - If CO Dial Tone Detect (refer to Ref. A) is ON and it is stored as the first or second digit, and the accessed CO Line is behind the PBX mode, its function is Dial Tone Detect. Otherwise, its function is Pause.
    - [FLASH] - Inserts a Flash into the speed number.
    - If the accessed CO Line is analog, its function is Flash to PX (or PBX).
    - If the accessed CO Line is ISDN line (refer to Ref. B) and it is stored as the first digit, it makes the remaining digits sent with envelope information not in the Calling Party Number IE but in keypad facility IE.
6. Press the [HOLD/SAVE] button.
7. If desired, enter the name (up to 12 characters) by dialing 2-digits for each character.
8. Press the [HOLD/SAVE] button.
9. To store continuously, repeat this procedure from Step 3.

To make a call using System Speed Dial from a DKTU:

1. Lift the Handset or press the [SPEAKER] button.
2. Press the [SPEED] button.
3. Dial the System Speed Dial bin.

To make a call using System Speed Dial from an SLT:

1. Lift the handset.
2. Dial the Speed Dial access code 558.
3. Dial the System Speed Dial bin.

Conditions

- The CPTU should be installed for dial tone detect.
- System Speed Dial is restricted by Station COS.
- There can be a maximum of 24 digits in a System Speed Dial number including other applicable digits and special function codes.
- If you dial an empty system speed bin, an error tone will sound.
- If you select a CO Line before dialing a System Speed Dial bin number, the selected CO Line is used regardless of if there is a programmed CO Line in the System Speed Dial bin number.
- If all CO Lines in the group are busy, a busy tone indication sounds when attempting to access a System Speed Dial number.
- System Speed Dial numbers are protected when the system is powered OFF.

Admin Programming

- Speed Dial Access (PGM 112 - FLEX 9)
- System Speed Zone (PGM 232)
- CO Dial Tone Detect (PGM 160 - FLEX 6)

Private Line

CO Lines in the system can be assigned for exclusive use by one or more DKTU users. Private lines are assigned to CO Line Group 00 and an appearance (Flexible CO button) is required at the DKTU (Loop or Pool keys cannot be used).

Operation

A private line will operate as a normal CO Line, except access is limited to assigned stations.

Condition

- A Private line cannot be picked up.

Admin Programming

- CO Line Group (PGM 141 - FLEX 1)

Paging

Internal, External, All-Call, and Meet-Me Page

Stations can individually be allowed or denied the ability to make pages. This applies to all internal zone paging and all external zone paging. A station denied access to paging may still answer a Meet-Me Page announcement.

There is one External Paging Zone available. External paging requires an externally-provided amplifier and paging system. External page can have a relay contact associated to it.

There are ten internal paging zones available. A station can be in any or all zones or in no zone at all. Stations not assigned to a page group can still make page announcements, if allowed in station programming. Stations can be assigned to a page group in order to receive pages but not allowed to make page announcements.

Stations to receive pages for a given zone are assigned to the zone. A page warning tone, if assigned, will be provided to the page zone(s) prior to the audio connection. The user is allowed to continue the page for a specified period. After the time expires, the user is disconnected and the page zone(s) returns to idle.

A user can respond to a page from any station and connect to the paging party for a private conversation. The user should respond to the page in the Page Time-out duration to connect the paging party.

ACCESS CODE	ITEM
501-510	Internal Page Zones 506-510 = Conference page zones
543	Internal All Call Page
544	Meet Me Page
545	External Page Zone
549	All Call Page (Internal & External)

Operation

To initialize a page:

1. Lift the handset or press the [SPEAKER] button.
2. Dial the desired paging code.
3. If assigned, after the page warning tone is presented, make the desired announcement.
4. Replace the handset and go on-hook.

To assign Meet-me Page at a FLEX button:

1. Press the [TRANS/PGM] button.
2. Press the desired FLEX button.
3. Dial 544.
4. Press the [HOLD/SAVE] button to accept changes.

To respond a Meet-me Page:

1. Lift the handset or press the [SPEAKER] button.
2. Dial 544 (Meet-me Page code).

   -or-

   In case of Nomad IP, dial Meet-me Page code and press [Send] button.
3. Press the [HOLD/SAVE] button. The call with the paging party is established and the zone returns to idle.

Conditions

- When external paging is required, appropriate external equipment should be attached to the proper external page connections on the MBU.
- A station which is in DND or busy cannot receive a page.
- A station which is not allowed to page cannot make a page.
- When paging is used in the system, another page is not allowed at the same time.
- A Page warning tone may be controlled by Admin. Programming.
- When the page timer expires, the paging connection is automatically released and ICM busy tone is presented to the paging station.
- You should lift the handset to make a page. When trying to make a page without lifting handset, "LIFT HANDSET TO PAGE" displays on the LCD.
- Paging can be programmed to a Flex. button.
- If an intercom call is received to the paging station, the caller will hear an intercom busy tone.
- If a CO line call is received at the paging station, the station will receive off-hook ring.
- A station may respond to a meet me page regardless of assignment of pick-up/paging group assignment/page access.
- A Page from a CO line can't be answered by pressing the [HOLD/SAVE] button or the code for meet-me answer. If a user tries to answer a meet me page request from CO line, an error tone is heard and an LCD message for error will be displayed.

Admin Programming

- Paging Timeout Timer (PGM 181 - FLEX 10)
- Page Warning Tone (PGM 161 - FLEX 4)
- Page Access (PGM 111 - FLEX 8)
- Internal Page Zone (PGM 118)
- External Control Contact (PGM 168)

Pre-recorded Message

You can record a VMIB message for paging.

Operation

To record a VMIB paging message:

1. Press the [TRANS/PGM] button.
2. Dial 65.
3. The announcement "Press # button to record" should be heard. If there is already a recorded message in the number dialed, the recorded message will be played.
4. Press the # key to start recording.
5. After hearing the announcement "Record your message" and the confirmation tone, record the desired message.
6. Press the [HOLD/SAVE] button.

   -or-

   Press the [SPEAKER] button when finished recording; the confirmation tone will sound.
7. Press the [SPEED] button while the recorded message is playing to deleted the message; a confirmation tone sounds.

To activate a VMIB message for paging:

1. Dial the page code (5xx) and lift handset.
2. The recorded VMIB message is paged.

To delete a VMIB message for paging:

1. Press the [TRANS/PGM] button.
2. Dial 67. Then the recorded message is deleted.

-or-

Press the [SPEED] button while the recorded message is playing, then the message is deleted and confirmation tone sounds.

Conditions

- Lift handset to make a page.
- If there is any recorded message, it is paged and if there is no recorded message, user's voice is paged.

Admin Programming

- Paging Timeout Timer (PGM 181 - FLEX 10)

SOS Paging

The system allows the recording of multiple VMIB messages for pre-recorded paging. A recorded VMIB message can be paged to a page zone in an emergency.

Operation

To assign {VMIB SOS Paging} at a flexible button:

1. Press the [TRANS/PGM] button.
2. Press the FLEX button to assign.
3. Dial paging code (5xx).
4. Dial Message number (001-070).
5. Press the [HOLD/SAVE] button.

To activate VMIB SOS paging:

Press the assigned {VMIB SOS Paging} flexible button.

Conditions

- This feature can be only activated by pressing an assigned flexible button on a DKTU in an idle state.
- The VMIB message for SOS Paging can be recorded only at an Attendant station.
- Paging zones include internal, external, and all call paging area.
- VMIB SOS paging is not restricted by VMIB Paging timer. The whole VMIB SOS paging can be paged even though Paging Timeout timer expires.

Push-to-Talk (PTT)

Using the PTT button of the Nomad IP, the user can talk to members of an internal page zone.

Operation

To register internal page zone for PTT:

1. Press the [TRANS/PGM].
2. Press "1#".
3. Input the internal page zone for PTT. If want to remove the assigned page zone, input "0".
4. Press the [HOLD/SAVE] button.

To use PTT:

1. Press the PTT button of the Nomad IP.
2. The user can talk to members of the internal page zone assigned.
3. If the PTT button is released, paging is ended.

Conditions

- PTT follows the feature operation of internal call page, except the Page Timer isn't applied.
- The range of internal page zone for PTT follows the internal page zone of each system except conference page zone.

Rerouting

Call Forward

A Station User can forward calls to any Station, Station Group, or VMIB in the system by activating feature codes. There are several types of Call Forwarding: Unconditional, Busy, No answer, Busy/No Answer, Unconditional Station Off-Net Call Forward, No Answer Station Off-Net Call Forward, Unconditional Station Off-Net Call Forward with Tel Number, No Answer Station Off-Net Call Forward with Tel Number, Incoming CO Off-Net Call Forward, and Follow Me Call Forward.

Operation

To activate Call Forward, follow the procedures in each sub-heading of this section.

To program Call Forward to a Flexible Button:

1. Press the [DND] button or the FWD Soft Key.
2. Press the flexible button to be assigned.
3. Press the [DND] button or the FWD Soft Key.
4. Assign the Call Forward type:

   1 = Call Forward, Unconditional

   2 = Call Forward, Busy

   3 = Call Forward, No Answer

   4 = Call Forward, Busy/No Answer

   5 = Unconditional Station Off-Net Call Forward

   6 = No Answer Station Off-Net Call Forward

   7 = Incoming CO Off-Net Call Forward

   8 = Unconditional Station Off-Net Call Forward with Tel Number

   9 = No Answer Station Off-Net Call Forward with Tel Number

   0 = Follow Me Call forward
5. Dial the number of the destination that will receive the call.
6. Press the [HOLD/SAVE] button.

To deactivate Call Forward:

1. Lift the handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Press the # key (Call Forward cancel code).

Condition

- To use Call Forward, a Station should be activated in Admin Programming.

Admin Programming

- Allow Off-Net Call Forward (PGM 111 - FLEX 18)
- Authorization Code Table (PGM 227)

Call Forward, Unconditional

A user forwards all calls immediately to another station, Hunt Group, or VMIB.

Operation

To activate Unconditional Call Forward:

1. Lift the Handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial the Call Forward code 1.
4. Dial Station or Group pilot number to receive the calls.
5. Replace the handset and go on-hook.

To assign {CALL FORWARD} button at a flexible button:

1. Press the [TRANS/PGM] button.
2. Press a flexible button.
3. Press the [DND/FWD] button.
4. Assign Call Forward type 1.
5. Dial the destination that will receive the call.
6. Press the [HOLD/SAVE] button to accept changes.

To deactivate Call Forward:

1. Lift the handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial Call Forward cancel code #.

Call Forward, Busy

When the User Station line is busy, incoming calls are forwarded.

Operation

To activate call forward for when the line is busy:

1. Lift the handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial the Call Forward type code 2.
4. Dial Station, Group, or VMIB number that will receive the call.

To assign the {CALL FORWARD} button at a flexible button:

1. Press the [TRANS/PGM] button.
2. Press a flexible button.
3. Press the [DND/FWD] button.
4. Assign the Call Forward type 2.
5. Dial the destination that will receive the call.
6. Press the [HOLD/SAVE] button.

To activate Call Forward:

1. Press the assigned flexible button.
2. The LED of [DND/FWD] button is flashing and the function assigned to the flexible button is activated.

To activate Call Forward to VMIB:

1. Lift the handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial the Call Forward code (1-4).
4. Dial VMIB selection code number that will receive the call.
5. Replace the handset and go on-hook.

To deactivate Call Forward:

1. Lift the handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial Call Forward cancel code #.

Call Forward, No Answer

When the station user does not answer within a predetermined amount time, the call can be forwarded to an alternate location.

Operation

To activate call forward for when there is no answer:

1. Lift the handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial a Call Forward type code 3.
4. Dial Station, Group or VMIB number to receive the call.

To assign {CALL FORWARD} button at a flexible button:

1. Press the [TRANS/PGM] button.
2. Press a flexible button.
3. Press the [DND/FWD] button.
4. Assign the Call Forward type 3.
5. Dial the destination that will receive the call.
6. Press the [HOLD/SAVE] button.

To activate Call Forward:

1. Press the assigned flexible button.
2. The LED of [DND/FWD] button is flashing and the function assigned to the flexible button is activated.

To deactivate Call Forward:

1. Lift the handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial Call Forward cancel code #.

Admin Programming

- Call Forward No Answer Timer (PGM 181 - FLEX 1)

Call Forward, Busy/No Answer

A User can direct the System to re-route calls to another station, group or VMIB when the Station is busy and/or does not answer in a predefined "No Answer" time. Incoming CO Lines, transferred CO Lines, and ringing Intercom calls are forwarded.

Operation

To activate call forward, when the line is busy or not answer:

1. Lift the handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial the Call Forward type code 4.
4. Dial Station, Group or VMIB number that will receive the call.

To assign {CALL FORWARD} flexible button:

1. Press the [TRANS/PGM] button.
2. Press a flexible button.
3. Press the [DND/FWD] button.
4. Assign the Call Forward type 4.
5. Dial the destination that will receive the call.
6. Press the [HOLD/SAVE] button.

To activate Call Forward,

1. Press the assigned flexible button.
2. The LED of the [DND/FWD] button should be flashing and the function assigned to the flexible button will be activated.

To deactivate Call Forward:

1. Lift the handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial Call Forward Cancel Code #.

Call Forward, Station Off-net (Unconditional, No answer)

Stations allowed call forward access can forward intercom and transferred CO Line calls to a directory number (telephone number) outside of the system. When a call is received, the system will access an outgoing CO Line and dial the telephone number entered by the user.

If a station assigned off-net call forward receives a call from an internal and/or external caller, the call will be forwarded to off-net unconditionally (Code 5) or after No Answer Ring timer is expired (Code 6).

Operation

To activate unconditional Off-Net Call Forward:

1. Lift the handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial the Call Forward code 5.
4. Seize a CO Line, if required.
5. Dial the Speed Dial bin number with the desired phone number.
6. Replace the handset and go on-hook.

To activate no answer off-net call forward:

1. Lift handset or press [SPEAKER] button.
2. Press [DND/FWD] button.
3. Dial Call Forward Code 6.
4. Seize a CO Line, if required.
5. Dial a speed bin number with the desired phone number.
6. Replace the handset and go on-hook.

To deactivate Call Forward:

1. Lift the handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial Call Forward cancel code #.

Call Forward, Station Off-net with Tel Num (Unconditional, No answer)

Stations allowed call forward access can forward intercom and transferred CO Line calls to a directory number (telephone number) outside of the system. When a call is received, the system will access an outgoing CO Line and dial the telephone number entered by the user.

If a station assigned off-net call forward receives a call from either an internal caller or an external caller, the call is forwarded to off-net unconditionally (Code 8) or after No Answer Ring timer is expired (Code 9).

Operation

To activate unconditional Off-Net Call Forward,

1. Lift handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial Call Forward Code 8.
4. Seize a CO Line, if required.
5. Dial the desired phone number instead of speed bin.
6. Replace the handset and go on-hook.

To activate no answer off-net call forward:

1. Lift handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial Call Forward Code 9.
4. Seize a CO Line, if required.
5. Dial the desired phone number instead of speed bin.
6. Replace the handset and go on-hook.

To deactivate Call Forward:

1. Lift the handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial Call Forward Cancel Code #.

Conditions

- The entered telephone number will be automatically assigned to a station speed bin, but the telephone number cannot be erased or edited from the normal speed bin programming mode.
- The assigned station speed bin will be deleted automatically when the user cancels this function.

Call Forward, Incoming CO Off-net (ATD only)

The System Attendant can direct the System to re-route (forward) incoming CO Line calls to a directory (telephone) number outside the system. When a call is received, the system will access an outgoing CO Line and dial the telephone number assigned by the attendant. Note that the system will automatically disconnect the call after the Unsupervised Conference Timer has expired.

FIELD	ACCESS CODE
CO Group Access	801-824
Individual CO Access	88XX
Retrieve Held CO Line	8*

Operation

To activate incoming CO Line off-Net at the Attendant:

1. Lift the handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial Call Forward code 7.
4. Dial CO Line/group access code (9, 801-8xx, 8801-88xx, or 8* for all CO Lines) or press {CO Line} button.
5. Dial the Speed Dial bin number with the desired telephone number.
6. Replace the Handset and go on-hook.

To deactivate CO Line Off-Net Call Forward (from the attendant station):

1. Lift the Handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial Call Forward code 7.
4. Dial CO Line/group access code (9, 801-8xx, 8801-88xx, or 8*) or press {CO Line} button.
5. Dial number.

Conditions

- If there is no idle CO Line, Off-net Call Forward is not activated.
- This feature is not applicable for DID lines.
- It is unconditional and forwarded immediately when the CO Line rings in the system.
- If a speed bin is programmed in a Flexible button, you may press the Flexible button instead of dialing the speed bin number.
- Toll restriction will be based on the COS of outgoing CO Line.

Call Forward, Follow Me

Follow-Me Call Forward can be activated at the Station or from any Station in the system with Call Forward access. It must be programmed from the Station that you will be forwarded to and a user password must be entered at the User Station first.

Operation

To activate Follow-me Call Forward from any Station:

1. Lift the Handset or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial Call Forward Code 0.
4. Dial forwarding Station number.
5. Dial forwarding Station authorization code.
6. Replace the Handset and go on-hook.

To deactivate Follow-me Call Forward from the Station (User Station Only):

1. Verify the line is in an idle state.
2. Press the [DND/FWD] button.
3. In off-hook state, press the [DND/FWD] button and dial #.

To register the authorization code in a station:

1. Press [TRANS/PGM] button.
2. Dial 31.
3. Enter the desired Authorization Code (5 digits).
4. Press [HOLD/SAVE] button to Save.

Condition

- An Authorization code must be registered to use Follow-me Call Forward.

Admin Programming

- Allow Off-net FWD (PGM 111 - FLEX 18)
- Authorization Code Table (PGM 227)
- Call Forward (PGM 111 - FLEX 2)
- Call Forward No Answer Timer (PGM 181 - FLEX 1)
- Off-net Call Mode (PGM 112 - FLEX 12)
- Unsupervised Conference Timer (PGM 182 - FLEX 6)

SLT Call Forward

An SLT User can forward calls to other stations, CO Lines, or to the System VMIB.

Operation

To activate call forward from an SLT:

1. Lift the handset.
2. Dial call forward code 554.
3. Dial the call forward type.
4. Dial the Station or Group speed number that will receive call.

   -or-

   Press the # key to forward to System VMIB.
5. Replace the handset and go on-hook.

To deactivate call forward from an SLT:

1. Lift the handset.
2. Dial the call forward code 554 and number.

   -or-

   Dial 559.
3. Confirmation tone should be heard, then replace the handset.

Conditions

- Call forward is maintained until it is deactivated.
- A call cannot be forwarded to a station in DND mode; when trying to forward to the station, an error tone will be heard.

- A call forwarding station cannot leave a VMIB message.
- The call forward feature may be canceled by code 559; the unified cancel code for DND/Call Forward/Message for SLT.
- Dial pulse SLT cannot be forwarded to VMIB.

Admin Programming

Call Forward (PGM 111 - FLEX 2)

VMIB Access (PGM 113 - FLEX 2)

Preset Call Forward

When a station receives incoming CO Calls and the Station is programmed to Preset Call Forward, the call is routed to the Preset Call Forward destination if the station does not answer within the Preset Call Forward Timer.

The destination can be another Station or a Hunt Group.

Conditions

- In Preset Call Forward, a busy station will not receive a CO Line ring and the next assigned station will receive the CO Line ring. If the station is not forwarded to another destination, then the call will not be forwarded and will continue to ring at the station until answered.
- The Preset Call Forward loop feature is not available (A>B>C>A).
- When a CO Line is forwarded with Preset Call Forward, the original station will stop ringing (the LED of {CO} button will flash continuously).
- If there is no direct {CO} button or {LOOP} button at destination station, the station will be bypassed.

Admin Programming

- Preset Call Forward (PGM 121)
- Preset Call Forward Timer (PGM 181 - FLEX 12)

Call Transfer

An Intercom Call or CO Call can be transferred to another station or CO Line during a conversation.

There are 2 kinds of call transfer: Screened and Unscreened Transfer, as described in the following table.

FEATURE	TRANSFERRING METHOD
Screened Transfer (With Announcement)	Transfer is completed after announcing the destination party.
Uncreened Transfer (Without Announcement	Transfer is completed without an announcement. After dialing the destination, and hearing the ring back tone, the originator can replace the handset.

Call Transfer to CO Line

A Station User may transfer a connected call to a new CO call. If an external ISDN party does not answer the transferred call within the designated Transfer Hold Recall time, the transferring Station will receive a recall ring. If the call remains unanswered, the attendant will receive a recall ring for the amount of time on the Attendant Recall timer. After that, the CO Line will be disconnected and returned to an idle state.

Operation

To make an Unscreened CO line call transfer:

1. Press the [TRANS/PGM] button. Intercom dial tone sounds and the previous call is placed on hold.
2. Seize a CO Line and dial the number of the external party that will receive the call.
3. Replace the handset and go on-hook.

To make a Screened CO line call transfer:

1. Press the [TRANS/PGM] button. The Intercom dial tone sounds and the previous call is placed on hold.
2. Seize a CO Line and dial the number of the external party that will receive the call.
3. When the called party answers, announce the call transfer.
4. Replace the handset and go on-hook.

Conditions

- For this feature, CO Lines (transferred CO Line and transferring CO Line) must be able to detect loop lost or disconnection condition.
- If the transferred CO Line does not have answer information (analog CO), recalling will not be presented when it is not answered. Also, the call will be disconnected after the Unsupervised Conference timer expires.
- If the transferred line is an ISDN CO call, a recall ring will be presented to the CO Line after Transfer Hold Recall time while the line is released.
- If you press the original incoming CO Line button while making transfer to an external number (Screened Transfer), the outgoing call is disconnected and the original incoming call is connected.

Admin Programming

- Transfer Recall Timer (PGM 180 - FLEX 7)
- Hold Recall Timer (PGM 180 - FLEX 5)
- Attendant Recall Timer (PGM 180 - FLEX 1)
- Open Loop Detect Timer (PGM 142 - FLEX 13)
- Unsupervised Conference Timer (PGM 182 - FLEX 6)

Call Transfer to Station

A Call can be transferred to another station within the system. The transfer can be screened (announced) or unscreened to an idle/busy station or Hunt Group. The transferred call will ring and provides an Exclusive Hold flashing indication to the receiving party's DKTU.

If the receiving station does not answer the call in the Transfer Recall time, the transferring station and the transferred station will receive recall. If the call still remains unanswered, the attendant will also receive a recall for the Attendant Recall time. After that, the transferred call will be disconnected.

Operation

To transfer to an idle station (unscreened):

1. Press the [TRANS/PGM] button.
2. Intercom dial tone sounds and the previous call is placed on exclusive hold.
3. Dial the station number that will receive the transfer.
4. Replace the handset or go on-hook.

To transfer to an idle station (screened):

1. Press the [TRANS/PGM] button.
2. Intercom dial tone sounds and the previous call is placed on exclusive hold.
3. Dial the station number that will receive the transfer.
4. When the station answers, announce the call being transferred.
5. Replace the handset or go on-hook.

To transfer to a busy station, use the following procedure:

1. Press the [TRANS/PGM] button.
2. The CO Line automatically is placed on exclusive hold and the ICM busy tone sounds.
3. The transferred station will receive a muted transferred CO Line ring.
4. If the call is not answered within the designated Transfer Recall time, the CO Line will be recalled to at transferring station and the attendant will receive a recall ring if the call remains unanswered.

To make an unscreened transfer by SLT:

1. Press and release the hook-switch, the intercom dial tone sounds.
2. The CO Line is placed on exclusive hold, and the Transfer Recall timer is activated.
3. Dial the Station number the call will be transferred to.
4. Replace the handset and go on-hook.

To make a screened transfer by SLT:

1. Press and release the hook-switch, the intercom dial tone sounds. The CO Line is placed on exclusive hold, and the Recall timer is activated.
2. Dial the station number that the call will be transferred to.
3. When the station answers, announce the call being transferred.
4. Replace the handset and go on-hook.

Conditions

- When the attendant has a DSS and a station of programmed in DSS receives transferred call, the LED of DSS button in attendant will flashing.
- When the SLT user is in the screened transfer mode and tries to converse both transferred station and CO Line, the user can activate brokers call with hook-flash.
- It is impossible to transfer a call to another busy SLT from a SLT. When receiving busy tone, SLT user can be connected to the CO Line with hook-flash.

Admin Programming

- Transfer Recall Timer (PGM 180 - FLEX 7)
- Hold Recall Timer (PGM 180 - FLEX 5)
- Attendant Recall Timer (PGM 180 - FLEX 1)
- No Touch Answer (PGM 111 - FLEX 7)

Holding and Parking

Hold

A Station User can place a call on hold. The following features are available depending on the desired result. The result of the holding operation can be determined by ADMIN programming.

FEATURE	DESCRIPTION
System Hold	Any Station can retrieve a held call. Another station in the group can seize the CO Line to answer.
Exclusive Hold	Only the station user who held the call can retrieve it. Another station in the group cannot seize the CO Line.

Operation

To place a CO Line on Exclusive/System Hold from a DKTU:

Press the [HOLD/SAVE] button once or twice (depending on Hold Preference).

To place a CO Line on Exclusive/System Hold from an SLT (depending on Hold Preference):

Hook-flash and dial 560.

To access a CO Line on Exclusive/System Hold from the DKTU where the hold was placed:

1. Lift the handset.
2. Press the {CO} button.

   -or-

   Dial 8 # and the CO Line number.

To access a CO Line on Exclusive/System Hold from the SLT where the hold was placed:

1. Lift the handset.
2. Dial 8*.

Conditions

- The CO Line placed on Exclusive Hold will flash at the Station and the LED of CO Line will light at other stations.
- The CO Line placed on System Hold will flash at all Stations.
- When Exclusive Hold is set in a Station, the Exclusive Hold Recall Timer will be initiated. After the Exclusive Hold Recall Timer is expired, the original Station will receive a recall for the duration of the I-Hold Recall Timer.
- When a System Hold is set at a station, the System Hold Recall Timer will be initiated. After the System Hold Recall Timer expires, the original station will receive a recall for the duration of the I-Recall Timer.
- When the I-Hold Recall Timer is expired, the Attendant will receive a recall for the duration of the Attendant Recall Timer. If the call remains unanswered for after the Attendant Recall time expires, the call will be disconnected.

Admin Programming

- Hold Preference (PGM 160 - FLEX 8)
- Attendant Recall Timer (PGM 180 - FLEX 1)
- Exclusive Hold Recall Timer (PGM 180 - FLEX 4)
- Hold Recall Timer (PGM 180 - FLEX 5)
- System Hold Recall Timer (PGM 180 - FLEX 6)
- Transfer Recall Timer (PGM 180 - FLEX 7)

Hold Preference

Preferred Hold type is set by ADMIN programming. When a User presses the [HOLD] button, the preferred type of Hold is activated. If the user presses the [HOLD] button twice, the other type is activated (toggle).

Operation

If System Hold is set as preferred Hold and the user presses the held the [HOLD] button once, the call is held by System Hold.

If System Hold is set as preferred Hold and the user presses the held the [HOLD] button twice, the call is held by Exclusive Hold.

When Exclusive Hold is assigned, another station in the group cannot seize the held call.

When System Hold is assigned, another station in the group can seize the held call and answer.

Admin Programming

- Hold Preference (PGM 160 - FLEX 8)

Automatic Hold

When a station is connected to a CO Call, the Station User can make another intercom call just by pressing the DSS button. In this case, the previous CO Call is automatically held.

Operation

To use Automatic Hold while on a CO Line call:

1. Press a {CO} button.
2. When the new CO Line is connected, the previous CO Call is placed the Admin Programmed preferred Hold state.

Admin Programming

- Automatic Hold (PGM 112 - FLEX 2)
- Hold Preference (PGM 160 - FLEX 8)

Park

A User can park an ICM or CO call in a virtual location. The User can then make a page announcement for the desired User to pick-up the parked call. The paged user can retrieve the call by dialing the designated location number.

Note: A Station must have a {CO} or {LOOP} button, to retrieve the call.

Operation

To park a call:

1. Press the [TRANS/PGM] button.
2. Dial the parking location 601-608.
3. Replace the handset or go on-hook.
4. Page the desired User to retrieve the call.

To retrieve the parked call from a DKTU:

1. Lift the Handset or press the [SPEAKER] button.
2. Dial the parking location to retrieve the parked call.

Conditions

- To receive the parked call, the Station should have a {CO} or {LOOP} button.
- If the call remains unanswered for the duration of the Call Park Recall time, the original Station that parked the call will receive a recall. If the call is still unanswered, then the attendant will receive the recall. If the Attendant does not answer again in the Attendant Recall time, the CO Line call will be disconnected and the line will be returned to an idle state.
- In case of ICM call parking, the attendant will not receive recall.

Admin Programming

Call Park Recall Timer (PGM 180 - FLEX 2)

Park and Page

This feature allows a caller reaching a mailbox to select an option to be placed into a system park zone and have the mailbox user paged. The mailbox must have a COS with this feature enabled.

Setup

1. Enable the feature in Admin programming.
2. At each station use station programming PGM 6* and record User Name.
3. At each station use station programming PGM 6+1 and record User Greeting (page instrctions).

Operation

1. Caller hears mailbox greeting with page instructions recorded by user.
2. Caller presses 8 to invoke this feature. The system places the caller into a park zone (starts at the first zone goes forward). The system then accesses the all call page code and broadcasts:

   "Call for [subscriber] in location XXX"

   The system then goes on hook.

Admin Programming

- PGM 113 - FLEX 17 Park and Page

Conditions

- Park and Page option is only available to CO callers.
- If the call stays in the park location for the call park recall timer the call will recall to the operator.
- Only one park and page iteration will be made before recall.
- There is a per station enable/disable for Park and Page.
- The default value is disabled.
- If there is no system resources to page, user can retry to request page.
- If 8 is not dialed, then caller can record a voice mail.

Pick-up

A station user can pick up a call received at another Station.

The following pick-up types are available:

FEATURE	DESCRIPTION
Directed Call Pick-up	To pick up a call ringing at another station within the accessible Intercom Tenancy Group
Group Call Pick-up	To pick up a call ringing at another station in the same pick-up Group.

Directed Call Pick Up

A station can pick up a call ringing other station by dialing the direct call pick up code, and the ringing station number.

Operation

To answer a call ringing at another station,

1. Lift the handset or press the [SPEAKER] button.
2. Dial Direct Call Pick-up code 7.

   -or-

   Press the programmed {Direct Call Pick-up} button.
3. Dial the Intercom number of the ringing Station.

To assign {DIRECT CALL PICK-UP} button at a flexible button:

1. Press the [TRANS/PGM] button.
2. Press the FLEX button to be assigned.
3. Dial 7.
4. Press the [HOLD/SAVE] button to accept changes.

Conditions

- A {CO Line}, {POOL} or {LOOP} button is required to pick up a CO Line call.
- When several calls are queued at a station or hunt group, the pick-up depends on the Pick-up Priority (PGM 173).
- Queued callback and private line cannot be picked up.

- An intercom call cannot be picked up at a Station in Hold or Park mode.
- When the same types of CO Lines are queued, the first queued CO Line is picked up.
- Direct call pick-up is allowed within the intercom tenancy group. A Station cannot pick up any call to the Station which belongs to denied Intercom Tenancy Group (PGM 120).
- A Station can answer an intercom call placed to a Doorbox using directed call pick-up.

Admin Programming

- Refer to the SBX IP 320 Installation Guide, Direct Call Pick up Code (PGM 107 - FLEX 6)
- PLA Priority Setting - Pick-up Priority (PGM 173)

Group Call Pick Up

A station can pick up a call ringing at another station in the same pick-up group. Ringing intercom calls, incoming CO Lines, recalling CO Lines and transferred CO Lines can be answered by a station instead of the ringing Station if the Stations belong to the same pick-up group.

Operation

To answer a call ringing at a Station in the same Pick-up group:

1. Lift the handset or press the [SPEAKER] button.
2. Dial the Group Pick-up code 566.

To assign a {GROUP CALL PICK-UP} button at a flexible button:

1. Press [TRANS/PGM] button.
2. Press the FLEX button to be assigned.
3. Type 566.
4. Press the [HOLD/SAVE] button.

Conditions

- An intercom call cannot be picked up at a Station in Hold or Park mode.
- A {CO Line}, {LOOP} or {POOL} button is required to pick-up a CO Line call.
- Queued callback and private line calls cannot be picked up.
- A station can answer an intercom call placed to a Doorbox using group call pick-up.
- When several calls are queued at a Station or Hunt Group, the pick-up order depends on Pick-up Priority (PGM 173).
- When the same type of CO Lines are queued, the first queued CO Line is picked up.
- Group call pick-up is allowed within the intercom tenancy group. A Station cannot pick up any call to the station that belongs to a denied Intercom Tenancy Group.

Admin Programming

- Pick-up Attribute (PGM 190 - FLEX 2)
- Pick-up Group Attributes (PGM 191)
- PLA Priority Setting (PGM 173) - Pick-up Priority

Software Upgrade

The SBX IP 320 MBU software can be upgraded by SBX IP 320 upgrade program using a PC. To upgrade with the SBX IP 320 upgrade program, the PC and SBX IP 320 system must be connected through Serial, LAN, or MODEM interface. Then the software file in the PC is sent to the system and the MBU software is upgraded.

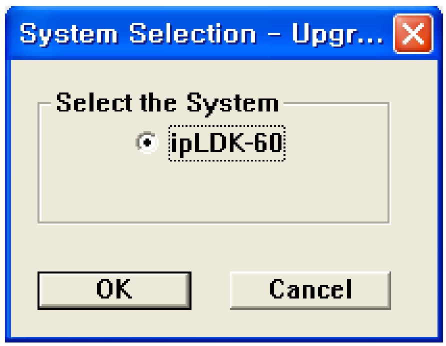
LAN

The MBU software can be upgraded through LAN interface by ipLDK upgrade program in a remote PC.

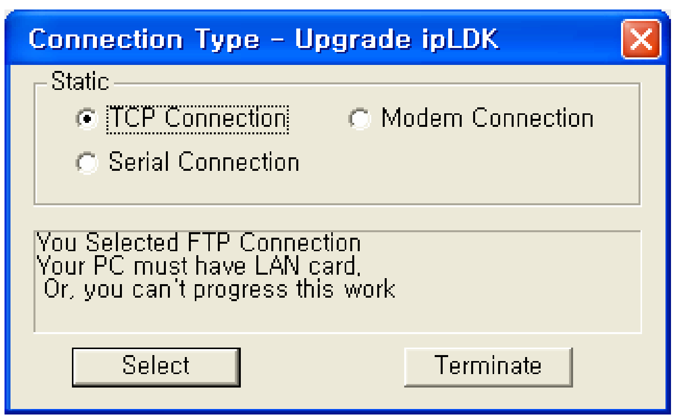
Operation

To upgrade the software via LAN, use the following procedure:

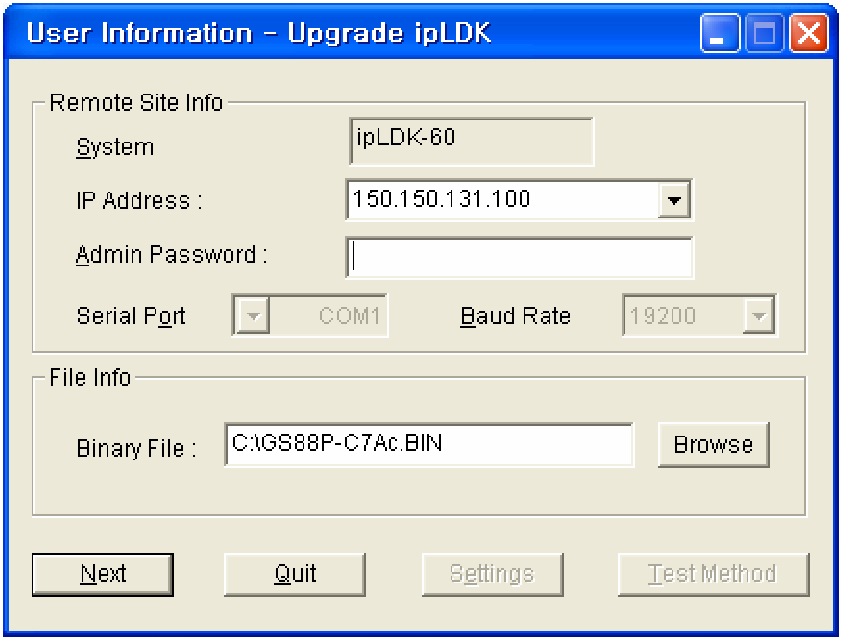
1. Connect the LAN cable to the PC LAN-card.
2. Run the ipLDK PC Upgrade program.



3. Select the ipLDK system to be upgraded.
4. Click on the OK button.



5. Select the port type TCP Connection.
6. Click on the Select button.



7. Enter the SBX IP 320 IP Address, Admin Password, and path of the Binary File.
8. Click the Next button.
9. Click on the Start button; the MBU software upgrade will begin.
10. When the ROM file is finished downloading, the SBX IP 320 System will erase the previous ROM data and fill the SBX IP 320 ROM area with the new ROM file.

Conditions

- When the line is released while upgrading process, just dial the phone number again. If the line was properly disconnected, it will be connected immediately.
- While in the upgrading process, other features do not work in the system.
- When the MBU software upgrade is finished without completing, you can retry to upgrade by doing the whole process again.
- If the ROM files you want to send are invalid, the MBU software upgrade will not be started.

Admin Programming

- Refer to the SBX IP 320 Installation Guide, IP Setting for MBU (PGM 108)

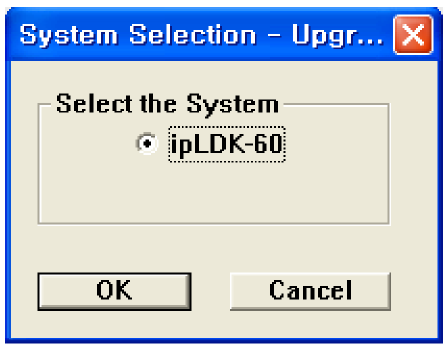
MODEM

The SBX IP 320 MBU software can be upgraded through Modem interface by using the ipLDK upgrade program in a remote PC.

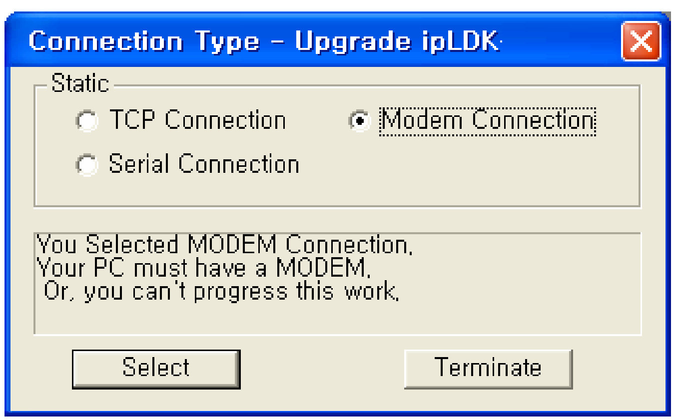
Operation

To upgrade the Modem software

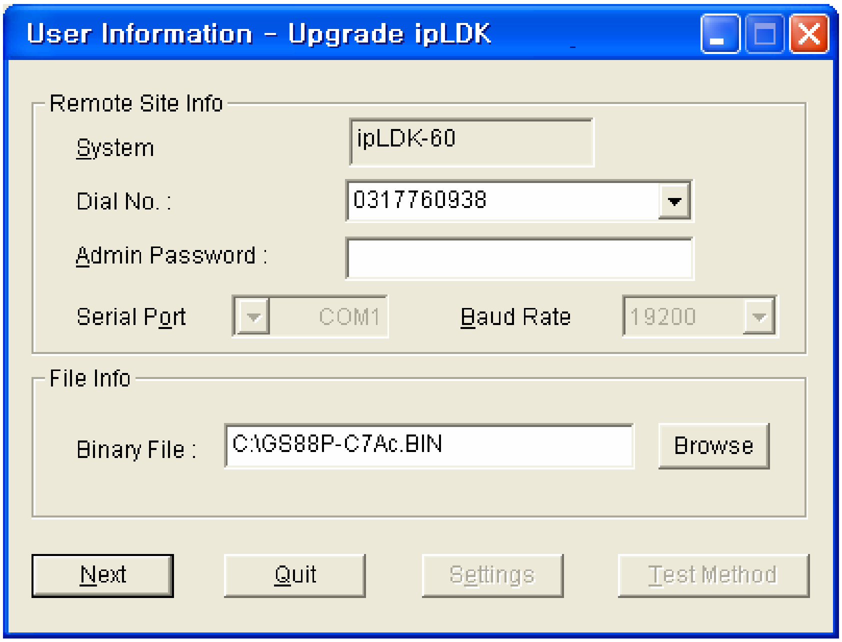
1. Run the ipLDK PC Upgrade program.



2. Select the ipLDK System to be upgraded
3. Click on the OK button.



4. Select the port type Modem Connection.
5. Click on the Select button.



6. Enter the SBX IP 320 Dial No., the Admin Password, and the path of the Binary File.
7. Click on the Next button.
8. Select your Modem Type in the Modem Configuration dialog box.
9. Click on the Start button; the SBX IP 320 MBU software upgrade will begin.
10. When the ROM file is finished downloading, the SBX IP 320 System will erase the previous ROM data and start to fill the SBX IP 320 ROM area with the received ROM file.

Conditions

- When the line is released while upgrading process, retry the ipLDK PC Upgrade program again.
- While in the upgrading process, other features do not work in the system.
- When the MBU software upgrade is finished without completing, you can retry to upgrade by doing the whole process again.

- If the ROM files you want to send are invalid, the SBX IP 320 MBU software upgrade will not be started.

Admin Programming

- Modem Assignment - ASC Device (PGM 170)

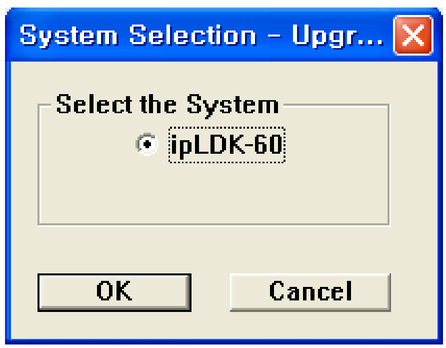
Serial (COM port)

The MBU software can be upgraded through RS-232C interface by using the ipLDK upgrade program in the PC.

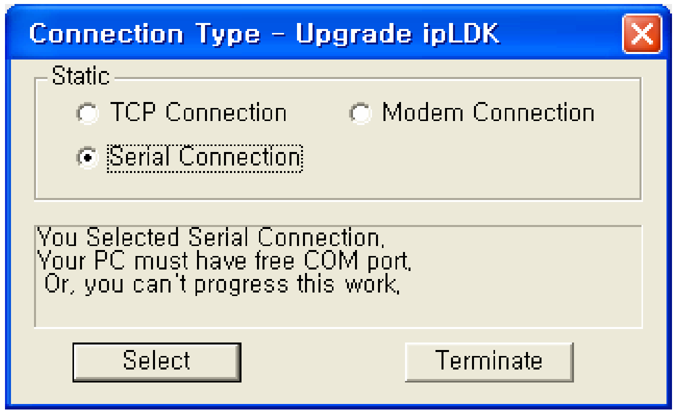
Operation

To upgrade the SERIAL (COM port):

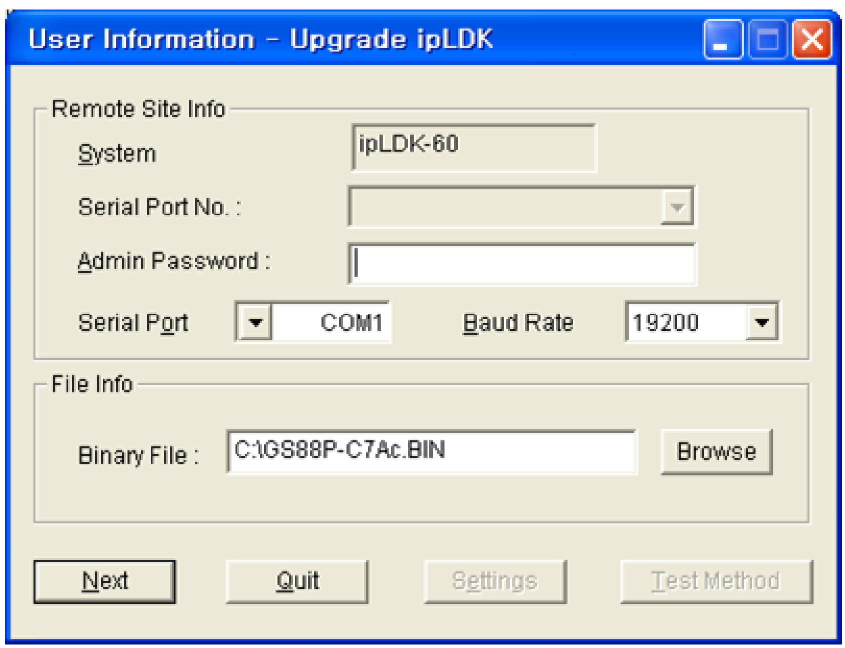
1. Connect the RS-232C cable between the SBX IP 320 system and the PC.
2. Run the ipLDK PC Upgrade program.



3. Select the ipLDK system to be upgraded.
4. Click on the OK button.



5. Select the port type Serial Connection.
6. Click on the Select button.



7. Enter the Serial Port Number, Admin Password, and select the serial port Baud Rate.
8. Click on the Next button.

9. Click on the Start button; the MBU software upgrade will begin.
10. When the ROM file is finished downloading, the SBX IP 320 System will erase the previous ROM data and start to fill SBX IP 320 ROM area with the downloaded ROM file.

Conditions

- While in the upgrade process, other features do not work in the system.
- When the SBX IP 320 MBU software upgrade is finished without completing, you can retry to upgrade by doing the whole process again.
- If the ROM files you want to send are invalid, the MBU software upgrade will not be started.
- The Serial port should be connected to COM port 2 in the MBU.

Admin Programming

- RS-232C Port Setting (PGM 174)

Station Message Detail Recording (SMDR)

The SBX IP 320 System SMDR (station message detail recording) provides detailed information about both incoming calls and outgoing calls. In this feature, it is programmable to record all calls or just outgoing long distance calls. SMDR information includes outgoing CO Line, dialed number, time, date, station that answer the call, and duration of call. Authorization codes may also be entered and recorded.

An SMDR record of ICM Calls can also be printed when ICM SMS is sent by Nomad SP or Nomad IP, and ez Phone

Operation

To print the SMDR:

1. Activate the PC utility program on a networked PC.
2. Connect the serial port of MBU to the serial port of the PC with the RS-232C cable.

At the Attendant Station:

3. Press the [TRANS/PGM] button.
4. Dial 0111 (Station Base) or 0113 (Group Base).
5. Enter the Station or Group range. The SMDR is printed to the PC.

To delete an SMDR:

1. Press the [TRANS/PGM] button.
2. Dial 0112 (Station Base) or 0114 (Group Base).

To abort SMDR printing:

1. Press the [TRANS/PGM] button.
2. Dial 0116.

Conditions

- There is an assignable SMDR record option ADMIN Program (PGM 177 - FLEX 3).
- If SMDR - LONG DISTANCE ONLY is selected, only outgoing toll calls will be printed, except SMDR Local Code (PGM 204).
- If SMDR - ALL CALL is selected, incoming and outgoing local and long distance call are printed.

- If user dials any number with a programmed long distance code as the first and second digit dialed or any number with more than maximum local call digit count, it will be regarded as a long distance call. (Max. local call digit count is programmable and the default value is 7.)
- The SMDR output records contain the following:
  - 5-digit station call originator (terminating for incoming) filed
  - 3-digit used CO line field
  - 8-digit call duration field (HH:MM:SS)
  - 8-digit year, month, and day (YY/MM/DD)
  - 5-digit time of day call originator field
  - 1-digit call identification digit-first digit in digit dial field
  - 18-digit collected dial digit field
  - 2-digit account group number field
  - 5-digit pulse metering count field
  - 10-digit call cost field
  - 12-digit account code field
- When the SMDR storage pools are almost exhausted, the system gives "Buffer full" warning signal to the attendant. The Attendant station LCD will indicate how many SMDR records remain to store at some intervals.
- Some stations can be grouped to count the billing with a SMDR receipt using an SMDR account group.
- The "SLT DTMF RLS TMR" should be adjusted to a reasonable value in order to print all digits that SLT dialed.
- ICM SMDR is not supported for Lost Calls.
- Only Calling Party NO is displayed at STA field.
- Only the Station No which is getting an ICM call is displayed at DIALED field. (ex) - E xxx : E is internal call, S is SM).
- SMS SMDR is also controlled by PGM 177 - FLEX 17, 18 option ON/OFF.
- ICM SMDR record format is same as existing SMDR record format.
- ICM HUNT Call SMDR record is same as just ICM Call. For example, Station 100 call to Hunt 620, and answer 620 member Station 110, then [STA] field Station 100, and [DIALED] field Station 110.

- Paging call SMDR can be printed. At [STA] field, paging station number is displayed, and be paged station number is displayed at [DIALED] field.
- In case of ICM Conference Room SMDR, station number which is entered Conference room is printed at STA field and Conference room no is printed at DIALED field, for example [Conf Rm 1].
- In case of Conference feature, Conference initializing station is always printed at STA field and Conference members are printed at DIALED field.
- In Hands free mode or Private mode, SMDR can be printed.
- ICM call parking feature SMDR is printed as two records. Before parking, and after parking and someone is answered records. Park recalling is printed as another record.
- In the ICM call parking feature, parked time is not calculated SMDR time. Two records are produced. Before parking and after parked call answering time is displayed.
- ICM SMDR is possible when ICM SMDR SAVE is set to ON (PGM 177 - FLEX 17)
- ICM SMDR Print is possible when ICM SMDR PRINT is set to ON (PGM 177 - FLEX 18)

Admin Programming

- SMDR Attributes (PGM 177)
- Metering Unit (PGM 142 - FLEX 3)
- SLT DTMF RLS Timer (PGM 181 - FLEX 13)
- SMDR Local Code Table (PGM 204)

Print-out

Lost call means that the caller gives up and terminates the call before the call is answered. The format of the individual call record is illustrated below, and the contents are focused on each types of lost call.

NO STA CO TIME START DIALED

----- ---- --- ------ ----------------- -----------------------

0001EXT031 00:00:10 24/07/08 11:55 R RING 00:05

Normal incoming call is received at an assigned stations of CO 031 during 5 sec.

0002 10100300:01:20 25/07/08 16:23 R RNG 00:09

DID call is disconnected during it being forwarded to ATD STA 101, because the dialed station does not exist.

0003100001 00:00:20 25/07/08 18:11 R100 RING 00:04

DID call is received at STA 100 during 4sec and disconnected.

0004 102002 00:01:20 26/07/08 18:37 R103 RING 00:04

DID call is received at STA 102 via unconditional call forward to STA 103 during 4 sec and disconnected.

0005 621008 00:00:20 26/07/08 13:02 G620 RING 00:06

DID call is received at Ring Group 621 during 6sec and disconnected.

0006 10000100:00:04 06/07/08 16:04 H100 RING 00:02

DID call is disconnected while STA 100 is being held it.

0007 10200100:00:07 06/07/08 17:04 H100 RING 00:02

DID call is disconnected while it is being transferred from STA 100 to STA 102.

G: Incoming call to hunt group, but the caller hangs up before answer

H: Answered incoming call was transferred to another station, but the caller hangs up before answer. An incoming call placed on hold state and cleared down in hold state.

R: Direct call (DID) to a station, but the call was disconnected before the station answers. Or direct call to station (A), but station (A) does not answer and the call was forwarded to station (B). The call was disconnected before station (B) answers.

Operation

To print the lost call count of record (from the Attendant station):

1. Press the [TRANS/PGM] button.
2. Dial 0117. The lost call count of record is printed in the PC connected to the system.

To clear the lost call count of records (from the Attendant station):

1. Press the [TRANS/PGM] button.
2. Dial 0118; "The lost call count is cleared" is printed and the lost call count time is restarted.

Conditions

- The SMDR record is sent to RS-232C automatically as soon as the event takes place.
- The SMDR record about lost calls is not saved. Only the records are counted.
- To activate the SMDR record, the SMDR field must be set in ADMIN programming.

Admin Programming

- SMDR Print Enable (PGM 177 - FLEX 2)
- Long distance/ All Call Recorded (PGM 177 - FLEX 3)
- Print Lost Call (PGM 177 - FLEX 6)

Supplementary Service

Collect Call Blocking for E1-R2 and LCO in Brazil

When the E1R2 or LCO receives the incoming collect call, it could reject to accept the call by operating collect call blocking procedure.

There are two kinds of collect call service types. One is "Without Indicator" and the other is "With Indicator". For LCO, only "Without Indicator" is available from PSTN network. For E1-R2, "Without Indicator" is operated with E1 line signaling and "With Indicator" is operated with R2MFC signaling.

Operation

To activate (deactivate) collect call blocking feature.

Set Admin to activate collect call blocking enabled (disabled). When the incoming call receives and answers. If activated, as soon as answering, it sends the collect call blocking signal towards the PX.

Conditions

- When the incoming party sends the collect call blocking signal towards the PX, if it is a collect call, it disconnects the call. If it is a normal call, it continues to communicate.
- For LCO, to activate collect call blocking feature, set PGM186-FLEX17 to 1 (Without Indicator) or 2 (With Indicator). For both settings, it works as "Without Indicator" mode.
- For LCO, to block collect calls, loop open is made for "Collect Call Break Timer" after answering during "Collect Call Make Timer".

Admin Programming

- PGM 186 - FLEX 17 - Collect Call Enabled/Disabled (select type of collect call service)

  0 : Disabled, 1:Without Indicator, 2:With Indicator
- PGM 186 - FLEX 18 - Collect Call Make Timer
- PGM 186 - FLEX 19 - Collect Call Break Timer

Message Wait Notification to Mobile Extension

When a VMIB message has been left at a station, the system can send a message wait notification to a mobile phone linked with the station. Mobile Extension is only for SIP trunks.

Operation

To activate (deactivate) message wait notification to a mobile phone:

1. Press [PGM] + 36.
2. Dial "1" to activate ("0" to deactivate).
3. Press the [HOLD] button.

Condition

- This can be activated when the SMS admin PGM turns on.

Admin Programming

- PGM 291 - FLEX 1 - SMS Service Center Number
- PGM 291 - FLEX 2 - SMS CENTER CLI
- PGM 292 - FLEX 1 - SMS Receive Station Assignment
- PGM 292 - FLEX 3 - SMS Usage
- PGM 292 - FLEX 4 - SMS Outgoing CO (CO to use for SMS sending)
- PGM236 - Mobile Extension

PSTN Short Message Service (SMS)

According to the SMS standard specification (ETSI ES201 912 V1.1.1, protocol 1), the SBX IP 320 system can support the PSTN SMS service; however, this feature is not currently supported. The SBX IP 320 system can receive SMS from PSTN and send SMS message to the PSTN using application or terminal.

The feature availability of each extension type is listed below for external SMS exchange.

Extension Type	Model Name	To Send	To Save	To Retrieve	Remark
PC Application	ez-Phone Nomad SP Nomad IP	Yes	Yes	Yes	--
DKTU	7208D 7224D	No	No	Yes	From DKTU, only retrieval is available
SLT	SMS SLT	Yes	Yes	Yes	--
	Other SLT	No	No	No	Non-SMS SLT
Others		No	No	No	--

Feature availability for each extension type

The following summarizes the information on the sending of an SMS message to a PSTN number (not an internal extension number, but a public phone number).

	External PSTN Number (Home, Office, Mobile)	Note
SMS SLT	Yes	Refer to the manual of each telephone for usage
ez-Phone	Yes	
Nomad SP	Yes	
Nomad IP	Yes	
Others	No	

To send an SMS message

To send an external SMS message, users can use the PC applications or extensions listed in the table above.

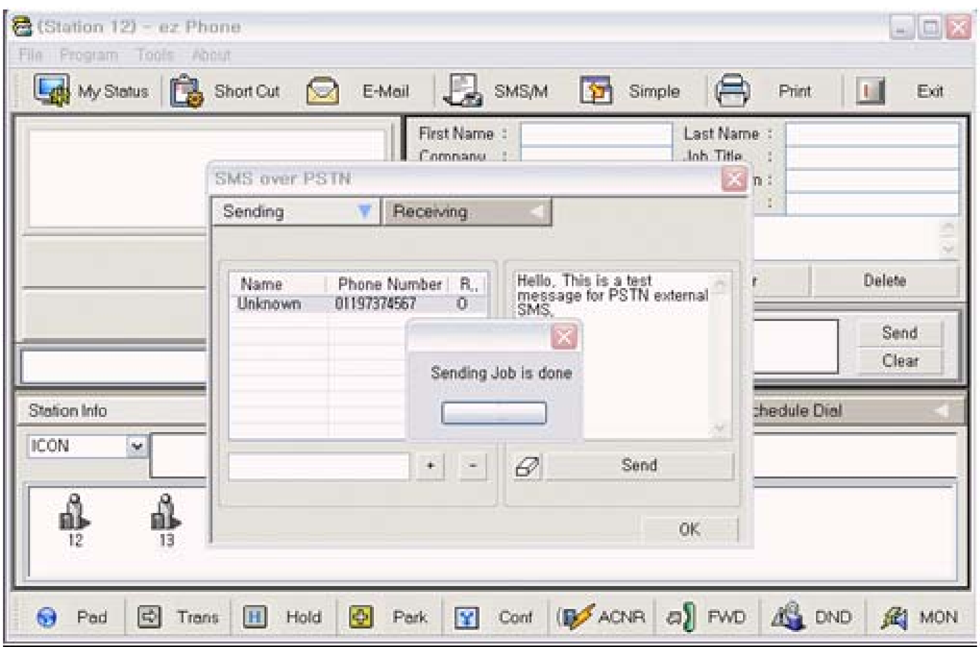
When an SMS message is received from the service center, it is saved in the memory of the SBX IP 320 first. Later, according to the settings in SMS destination (PGM 300), the message is delivered to the extensions. If the extension can store a message, the message is moved from the memory of the SBX IP 320 to the extension so that the message is deleted from the memory of the SBX IP 320. The following table contains the information on the storage place for each SMS message.

Operation

To send an SMS in an application:

1. Enter Called Party Number (Destination Address) in the Phone Number input section of the application and press the "+" button to insert dialed telephone number into SMS receiver list above the input section. Multiple Called Party Number can be added. Then you can send the same message to multiple called parties.
2. Enter your text and press the send button.

Example) SMS Sending in ez Phone



To send an SMS message from an SMS SLT:

1. User composes an SMS message from SMS SLT.
2. User initiates the transmission of the message.

Actually, the procedure is the same even if the SMS SLT is connected to the SBX IP 320 system.

That is, the user edits an SMS message from the SMS SLT and initiates the transmission.

(This is the end of operation from the user side)

Then, the SLT sends the SMS message to the SMS service center.

At the beginning stage of sending the SMS message, the SBX IP 320 system routes the call from the SMS SLT to the SMS service center so that the message can be sent to the destination properly.

To receive an SMS:

1. When an incoming call comes in with the same Caller ID in "SMS Service Center CLI", then the system connects the CO with the SMS channel to receive the SMS from the PSTN.
2. The SMS message delivery is completed and the SMS message is saved in the memory of the SBX IP 320 system.
3. If the assigned station of application is a DKTU, the SMS message is sent to the application. And the system deletes the SMS message from the memory of the SBX IP 320 system.
4. If the assigned station is a DKTU without ez-Phone, the LCD display of the DKTU indicates the SMS arrival.
5. If the assigned station is an SMS SLT, the system regenerates the SMS message to the SLT phone. When the SMS SLT receives a SMS message correctly, the system deletes the SMS message from the memory of the SBX IP 320 system.

To retrieve an SMS in a DKTU:

1. When the new message is received, following display is shown.To retrieve, press [CALLBK].

```
SMS
```

2. Then the SMS message list is shown.

```
*1. This is James........
 2. Hello......
```

3. To retrieve the content, press the number of message (ex. Press Digit 1). Then the content will be shown.

   Lines of a long message can be scrolled by using the volume up/down button.

```
This is James. Please call me first
thing Monday morning on the ....
```

   While you are retrieving SMS messages you can use the [CONF] button to delete the message or the [SPEED] button to move to the upper menu.

Conditions

- In the case of power failure, all SMS messages will not be protected.
- A max of 4 SMS Channels are serviced.
- A max of 100 SMS messages can be stored in the SBX IP 320 system. If the SMS message buffer is full, a warning message is sent to attendant and a new SMS message cannot be received.
- When the SMS destination is an SLT, the SBX IP 320 tries to regenerate the SMS message three times when the destination is in idle state. If the destination does not respond to the call for regeneration, the SMS message is deleted from the SBX IP 320 after retrying three times.
- The feature "Sending SMS to PSTN destination" is available from applications such as ez-Phone, Nomad SP, Nomad IP, and SMS SLT.
- ezPhone, Nomad SP, and Nomad IP can send an SMS message with a max of 160 characters.

Admin Programming

- SMS Service Center Number (PGM 291 - FLEX 1)
- SMS CENTER CLI (PGM 291 - FLEX 2)
- SMS Receive Station Assignment (PGM 292 - FLEX 1)
- SMS Usage (PGM 292 - FLEX 3)
- SMS Outgoing CO (CO to use for SMS sending) (PGM 292 - FLEX 4)

Traffic Analysis

The system can monitor and print various system activities at the request of the main Attendant. The information can be used to:

- Monitor and evaluate system performance
- Observe current usage and take corrective actions, if needed.
- Anticipate possible CO line problems
- Determine system updates and upgrades

The traffic data is output to the RS-232C or LAN. The system supports the following traffic reports.

- Attendant Traffic Report
- Call Summary Report
- Call Hourly Report
- H/W Unit Usage Summary Report
- CO line Traffic Summary Report
- CO line Traffic Hourly Report

Operation

The traffic analysis is only available at the main Attendant (refer to the Attendant Programming Menu Table in the Programming Guide). The measurement time type can be one of Today's peak time, Yesterday's peak time, Last hour, Yesterday's total and Today's total.

To print all summary traffic report:

1. Press the [TRANS/PGM] button.
2. Dial 0121.
3. Select Measurement Time type.
4. Press the [HOLD/SAVE] button.

To print all summary traffic report periodically:

1. Press the [TRANS/PGM] button.
2. Dial 0122.
3. Select the Measurement Time type.
4. Press the [HOLD/SAVE] button.

To cancel periodic printing of all summary traffic report:

1. Press the [TRANS/PGM] button.
2. Dial 0123.
3. Press the [HOLD/SAVE] button.

To print the each traffic report:

1. Press the [TRANS/PGM] button.
2. Dial 0124-0129.
3. Select Measurement Time type or CO group number if required.
4. Press the [HOLD/SAVE] button.

Conditions

- This feature is available at the main Attendant station.
- The Printing of all summary service will generate an attendant traffic report, call summary report, H/W unit usage summary report, and CO traffic summary report.

Admin Programming

- Print Port Selection (PGM 175)

Attendant Reports

The SBX IP 320 system supports the following report to analyze the attendant resource.

- **Attendant Traffic Report** - Choose one of the following: time duration, today's peak time, yesterday's peak time, last hour, yesterday's total and today's total. It provides the following information fields.
- **Analysis Start Hour** - Starting time of hour duration which the data is recorded.
- **Attendant Number** - The station number of the attendant.
- **Total Calls** - The number of total incoming calls except CO ring group call, hold recall ring.
- **Calls Answered** - The number of answered calls by all active attendants during the measuring hour.
- **Calls Abandoned** - The number of calls which ring at attendant and is dropped before answering at attendant.
- **Calls Held-Abandoned** - The number of calls which is dropped while the call is in hold mode. Held calls which is time out and recalled are included in this call count.
- **Calls Held** - The number of calls which answered by attendant and placed on hold state by attendant.
- **Time Available** - The time duration which the attendants don't answer the calls but, are available to handle new calls. Measured with minutes.
- **Time Talk** - The total time during measuring interval (the attendants are active or converse with a CO line). Talk time is not started until the call is answered by attendant. The duration of time between call termination and answering at attendant is not accumulated as Time Available or Time Talk.
- **Time Held** - The total amount of time which attendants have calls on hold.
- **Time No Answer** - The average amount of time that calls in queue and/or ringing at attendant before the caller hangs up.
- **Speed of Answer** - The average elapsed time from when a call is terminated by attendant to when the call is answered by an attendant.
- **Type** - Type of attendant (system or main or intercom tenancy group).

Operation

To print the Attendant Traffic Report at the main attendant station:

1. Press the [TRANS/PGM] button.
2. Dial 0124.
3. Select Measurement Time type.
4. Press the [HOLD/SAVE] button.

Ex.)

```
==========================================================
Site Name   :
Report Type : Attendant Traffic Report - Yesterday Total
Date        : 02/12/08 13:14
==========================================================

 Atd  Meas       --------- Calls ---------    ------- Time ------     Time  Speed   Atd
 No   Hour  Total  Ans  Abnd H-Abd Held  Avail   Talk   HeldNoAns  Ans    Type

 2629  --:--    9     3    6      0      0    02:02   00:00  00:00      00:00  00:00   Sys

 4807  --:--    8     6    2      0      0    04:21   00:13  00:00      00:09  00:04   Main
 3619  --:--    4     4    0      0      0    01:04   00:21  00:00      --:--  00:01   Main
 2618  --:--    0     0    0      0      0    00:05   00:00  00:00      --:--  --:--   Main
 3629  --:--    6     1    5      0      0    02:58   00:23  00:00      00:14  00:03   Main
```

Call Reports

The system supports the following reports to analyze the call status of the system.

Call Summary Report - Monitor all day's call traffic and generate this report that shows call status of last hour, today's peak time, yesterday's peak time, yesterday's total and today's total.

Call Hourly Report - Analysis of call overload by showing the last 24hour's per hour call.

This report provides the following information fields:

- Analysis Start Hour - Starting time of hour duration which the data is recorded
- Number of Calls Completed - The total number of calls completed or answered during the listed hour.

Operation

To print the Call Summary Report at the main Attendant station:

1. Press the [TRANS/PGM] button.
2. Dial 0125.

To print the Call Hourly Report at the main Attendant:

1. Press the [TRANS/PGM] button.
2. Dial 0126.

Ex.)

==

Site Name :

Report Type : Call Summary Report

Date : 02/12/08 13:14

==

	Analysis Start Hour	Number of Calls Completed
Last Hour	**13:00**	**14**
Today Peak	**10:00**	**141**
Yesterday Peak	**10:00**	**119**
Today Total	**--:--**	**413**
Yesterday Total	**--:--**	**970**

CO Reports

The system supports the following reports to analyze the traffics of CO line group.

CO Traffic Summary Report - Analysis of traffic status of the CO group by showing applicable statistics. Choose one of the following: time duration, today's peak time, yesterday's peak time, last hour, yesterday's total and today's total. It provides the following information fields:

- **Peak Hour for All CO Groups** - The time duration (hour) in a day that has the largest total usage when summed over all CO groups.
- **Group Number** - A number that identifies each CO group associated with the displayed data. Group numbers are displayed in numeric order, beginning with the lowest number and continuing to the highest one.
- **Number of CO** - The number of CO line in the group
- **Analysis Start Hour** - The time (24-hour mode) taking the measurement.
- **Total Usage** - Total usage for all CO lines in the CO group. It represents the total time that the CO lines are busy during the measurement period. Total usage measures each time when a CO line is seized for use by an incoming call or an outgoing call.
- **Total Attempt** - The number of incoming and outgoing call attempt in the CO group
- **Incoming Attempt** - The number of incoming call attempt in the CO group.

- **Outgoing Attempt** - The number of outgoing call attempt in the CO group.
- **Group Overflow** - The number of calls offered to a CO group that are not carried. Rejected calls for authorization will not be included.
- **Percentage All CO Busy** - The percentage of time that all CO lines in the CO group are simultaneously in use during the time interval.
- **Percentage Fail to Attempt Outgoing** - The percentage of offered calls that are not carried on the CO group. It will not be included unauthorized calls which are denied on the CO group or uncompleted calls carried on the CO group (unanswered calls).

CO Traffic Hourly Report - Analysis of CO traffic patterns by showing per hour CO traffic for the past 24 hours.

Operation

To print the CO Traffic Summary Report at the main Attendant station:

1. Press the [TRANS/PGM] button.
2. Dial 0128.
3. Select Measurement Time type.
4. Press the [HOLD/SAVE] button.

To print the CO Traffic Hourly Report at the main Attendant station:

1. Press the [TRANS/PGM] button.
2. Dial 0129.
3. Enter the CO group number.
4. Press the [HOLD/SAVE] button.

Ex.)

==

Site Name:

Report Type : CO Group Summary Report - Yesterday Total

Date: 02/12/08 13:15

==

Peak Hour For All CO: 10:00

Grp No	Num COs	Anal Hour	Total Usage	Total Seize	Inc. Seize	Out. Seize	Grp Ovfl	% ACB	% FAO
1	62	--:--	1319	1050	269	781	0	0	---

Hardware Unit Reports

The system supports the following report to analyze the usage of HW unit resources of the system such as Tone Receiver, VMIB.

H/W Usage Summary Report - Analysis of whether the system has enough H/W unit resources such as DTMF Receiver, VMIB, CPTU by showing the statistics. Choose one of the following: time duration, today's peak time, yesterday's peak time, last hour, yesterday's total and today's total. It provides the following information fields:

Type - The type of H/W unit being measured.

Number of Unit - The total number of installed H/W unit.

Analysis Start Hour - The starting time of the last hour or the hour with the highest Peak Req. measurement.

Total Requests - The system-wide total number of requests, by call processing for DTMF, CPTU, VMIB during the listed hour. It is calculated by incrementing a counter for each request.

Total Demand - The system-wide total number of requests that are denied because there is no available H/W unit during the listed hour.

Operation

To print the H/W Unit Usage Summary Report at the main Attendant station:

1. Press the [TRANS/PGM] button.
2. Dial 0127.
3. Select the Measurement Time type.
4. Press the [HOLD/SAVE] button.

Ex.)

```
=====================================================
Site Name:
Report Type : H/W Unit Usage Summary Report - Yesterday Total
Date: 02/12/08 13:15
=====================================================
 Unit   Num   Anal    Total    Total
 Type  Unit   Hour    Req    Denied
 VMIB   4    --:--     27        0
 DTMF  13    --:--     27        0
 CPTU  12    --:--     27        0
```

Voice Service

DID Call to Each Station's Voice Mailbox (Future Feature)

A maximum of 1000 Flexible DID Table entries can be programmed. Each Flexible DID Table entry has five attributes - "DID Name", "Day Destination", "Night Destination", "Weekend Destination", and "Reroute Destination". If the destination of Flexible Conversion DID table (PGM 231) is programmed as the Station 's VMIB, a DID Call is routed to the Station's VMIB and an external DID caller can leave message directly.

Conditions

- If the Day/Night/Weekend DID destination is busy, call is rerouted to Reroute Destination
- VMIB Access option must be ON (PGM 113 - FLEX 2).

Admin Programming

- VMIB Access (PGM 113 - FLEX 2)
- DID Destination (PGM 231)

Direct Transfer to VMIB

A CO/ICM Call can be transferred directly to a station's voice mailbox.

Operation

Below operation is executed at transferring Station:

1. Station is on a CO/ICM call.
2. Press the [TRANS] button to transfer to another station.
3. Press the [CALL BK] button.
4. Dial a destination Station number or press the DSS button of the desired Station.
5. Hang up when transfer is completed. After transfer action is ended, the transferred party hears a recorded message and a confirmation tone; then they can record a voice message.

Conditions

- The VMIB Access option must be ON (PGM 113 - FLEX 2).

- A Direct transfer to a Net number's VM box is not allowed.

Admin Programming

- VMIB Access (PGM 113 - FLEX 2)

Mailbox Buttons

Description

This feature allows station users to assign VM "buttons" to represent specific voice mail boxes in a system. The user can access his own station's voice mail box to check a new or saved message. The user can also program mailbox buttons for other mailboxes, that will be used to check for messages or transfer a caller directly to a mailbox via the use of this button.

Operation

- The user presses TRANS/PGM + FLEX + TRANS/PGM + 47 + XXX (XXX=desired mailbox button) + HOLD/SAVE to make the Mailbox Button.
- When a new message is left for a mailbox, the mailbox button LED flashes at the rate of 1 flash per 1000 ms. The subscriber can press the button to access the mailbox. This applies to all mailbox buttons on the user's keyset.
- While on a call, the user can press the preprogrammed mailbox button to transfer the caller directly to that programmed mailbox.

Conditions

- Each DKTU or IPKTU may have mailbox buttons programmed up to the number of flex buttons on the phone, i.e. 24 or 8 depending on the type of phone.

No answer Call to VMIB

A CO or ICM Call is routed to a station, and this Call is not answered by someone within the predefined [NO ANS TO VMIB] time, the Call is rerouted to the station's voice mailbox.

Operation

1. Call to Station 100.
2. No answer at Station 100.
3. No answer to VMIB timer expired.
4. Recording announcement is played.
5. After confirmation tone is heard, caller can record voice.

Conditions

- If a Destination is forwarded to another destination (another station, hunt group, VMIB…), then this timer is not operated.
- A Hunt Call is not assigned to this timer regardless of the "No Answer to VMIB ON/OFF option", because Hunt uses a call flow.
- To correctly work this feature, the VMIB ACCESS option must be ON (PGM 113 - FLEX 2).

Admin Programming

- NO Answer to VMIB timer (PGM 181 - FLEX 20)
- Auto Forward to VMIB (PGM 113 - FLEX 14)
- VMIB Access (PGM 113 - FLEX 2)

Recording System VMIB Announcement

An Attendant station in the system can record the voice announcements as system greetings and prompts.

System greetings must be recorded before use. System prompts in user's language are contained as default in VMIB. Users can also modify those prompts.

Prompts for date and time are contained in VMIB to be used for date and time stamping. With the help of these prompts, users can understand when the voice message has arrived. Prompts for date and time are also built-in and recorded in the user's language.

Operation

To record system greetings from the Attendant station

1. Press the [TRANS/PGM] button and dial 06.
2. Dial the message number. Then you will hear the announcement "Press the # key to record." If there is already a recorded message in the number dialed, a corresponding message will be played.
3. Dial # to start recording. Start the recording after hearing the announcement "Record your message" and hearing the confirmation tone.

   -or-

   Dial * to record using an external music port on the MBU.
4. Press the [HOLD/SAVE] button to finish recording. Then a confirmation tone sounds and you can record the next one.

   Pressing the [SPEAKER] button while recording, stops the recording and the recorded message is saved.

To delete system greetings from the Attendant station:

1. Press the [TRANS/PGM] button.
2. Dial the code 06.
3. Dial the appropriate message number (when there is a recorded message in the number dialed, the recorded message is played).
4. Press the [SPEED] button while the message is playing to delete it.

To add additional message on VMIB message:

1. Press the [ADD] soft button while listening to the message (on 7208D/7224D).
2. Record additional message.
3. Press the [HOLD/SAVE] button.

To rewind the current message:

Press the [REWIND] soft button, the message is rewound with the VM MSG Rewind Timer (on 7208D/7224D).

SYSTEM PROMPT MESSAGES (FIXED)

NO	MESSAGES
071	Reserved
072	Reserved
073	Invalid Number Prompt
074	Time Out Prompt
075	Retry Prompt
076	Transfer to Attendant Prompt
077	Reserved
078	Leave Message Prompt
079	Record Start Prompt
080	Authorization Code Prompt
081	Busy Prompt
082	Reserved (for Office version) Wakeup Prompt (only for Hotel version)
083	Station Off-net Forward Prompt
084	DND Prompt
085	No Answer Prompt
086	Reserved
087	Reserved
088	Remote VMIB Control Main Menu Prompt
089	Remote VMIB Sub-menu for digit 1 in Main Menu

SYSTEM PROMPT MESSAGES (FIXED)

NO	MESSAGES
090	Reserved
091	Reserved
092	Reserved
093	Remote VMIB Sub-menu for digit 2 in Main Menu
094	Remote VMIB Sub-menu for digit 3 in Main Menu
095	Remote VMIB Sub-menu for digit * in Main Menu
096	Leave Message after Tone Prompt
097	Message Waiting Indication Prompt
098	Default User Greeting Prompt
099	--
100	--

Conditions

- System Greeting messages are 001-070 by default. You can select one of 70 messages.
- System Prompt messages are 071-100 by default. The number is message and user cannot change the numbering plan arbitrarily, but users can modify those prompts by recording their own messages in the number.
- System greetings and prompts can be recorded only at system attendant station.
- There is no time limit to record system greetings and prompts at the attendant station.
- If the VMIB is not installed in the system, it is impossible to record system greetings and prompts. An error tone will sound if an attempt is made to record system greetings and prompts in a system without a VMIB.
- If there is a recorded message in the bin, the previously recorded message is played when user dials the message number.
- If you stop recording by pressing the [SPEAKER] button or go on-hook while recording, the already recorded message is saved. You should delete the recorded message to cancel the recording.
- To record or delete a message at attendant station, all the VMIB ports should be in the idle state.

- When a call is transferred to the attendant, a "Transfer to Attendant" prompt will be provided to the caller and ring-back tone will sound after the announcement.
- If there is no recorded greeting or prompt, the corresponding tone will be heard.
- A max of 800 user messages are available in a VMIB.
- It is possible to use only 100 messages for system greetings (system greetings, system prompt).
- When the memory is full while recording a system greeting, the recorded portion of the message before memory full will be saved.
- It is possible for station groups to have different system greetings.
- When recording system greetings and prompts at an attendant station, they will be saved at all VMIBs in the system.
- The system supports system prompts (072-100) basically. However, users may use their own prompts by recording the prompts at an attendant station.

Admin Programming

- VMIB Access (PGM 113 - FLEX 2)
- VMIB User Record Timer (PGM 181 - FLEX 3)
- VMIB Valid User Message Timer (PGM 181 - FLEX 4)
- Station Group Assignment and Attributes (PGM 190 and 191)

Recording User VMIB Announcement

If access to the VMIB is allowed, a user can record a User Greeting and can forward calls to the VMIB port according to forward condition type. The caller can leave a voice message wait at the station after hearing the user greeting.

Operation

To record a user greeting at a station:

1. Press the [TRANS/PGM] button.
2. Dial 61.
3. Dial the message number; the announcement "Press the # button to record." If there is already a recorded message in the number dialed, the recorded message will be played.

4. Dial #; then start recording after hearing the announcement "Record your message" and a confirmation tone.
5. Press the [HOLD/SAVE] button.

   -or-

   Press the [SPEAKER] button to finish the recording; a confirmation tone sounds.

To delete a user greeting at a station:

1. Press the [TRANS/PGM] button.
2. Dial 66.
3. The User greeting is deleted and Forward is deactivated.

To activate call forward to VMIB from a station:

1. Go off-hook or press the [SPEAKER] button.
2. Press the [DND/FWD] button.
3. Dial the forward type (1-4).
4. Dial the # key; a confirmation tone is heard.

To deactivate call forward to VMIB from a station:

   Press the [DND/FWD] button.

To leave a voice message wait at a Station:

1. The caller will hear the User greeting and the "Record your message" announcement.
2. After a beep tone, record your message.
3. Hang-up to complete recording.

To retrieve a recorded voice message wait at the Station:

1. Press the flashing [CALLBK] button; for SLT and 2/8 BTN DKTU, dial 557.
2. The message number prompt is heard and the voice message (FIFO or LIFO) and Time & Date prompt for the message is played.
3. Pressing the [CONF] button, the current message is deleted and the next message is heard.
    - For SLT and 2/8 BTN DKTU, Dial # 1, and press the [HOLD/SAVE] button; the current message is saved and the next message is heard.
    - For SLT and 2/8 BTN DKTU, dial # 2, and press the [CALLBK] button, the current message is played again.
    - For SLT and 2/8 BTN DKTU, dial # 3.
4. Pressing the [ADD] soft button, you can record an additional message (possible with 7224D which has 3 soft keys).
5. Pressing the [REWIND] soft button, you can rewind the current message with VM MSG Rewind Timer (possible with 7224D which has 3 soft keys).

Conditions

- There is no time limit to record user greeting in a station.
- When a caller leaves a voice message wait, the recording time is controlled by Valid User Message Timer (PGM 181 - FLEX 4) and VMIB User Record Timer (PGM 181 - FLEX 3). When the recorded message is shorter than the Valid User Message time, the message is not saved. Also, if the User Record Timer expires, a confirmation tone is heard and the message is saved in the station.
- If the station has several messages to be retrieved by pressing the [CALLBK] button, the message only with station number will be retrieved first. (Message wait with station number -> VMIB Message wait -> CLI Message wait -> VM group Message wait)
- Pressing the [CALLBK] button at the calling station before the user greeting is played causes the message wait with only station number to be saved in the called station.
- When a user calls a station which is forwarded to VMIB, they will hear the user greeting and a beep tone. After the beep tone, the user can leave a voice message.
- Each station may have up to 800 VMIB message waits.
- If all the VMIB ports are busy, ring-back tone will be provided instead of the user greeting. Also, the VMIB Station Forward Timer is started to retry to answer.
- A User can leave and receive message wait using an SLT with message wait lamp.

- Individual user greetings and VMIB message waits are protected from system reset.
- To retrieve left message wait, the order of playing is changeable. Originally, TIME, DATE, and a left message are played. Depending on how ADMIN is set, DATE, TIME, and a left message are played.
- To retrieve a left message wait, the Message Wait Retrieve Password would be used by ADMIN. If PGM 113 - FLEX 8 is set, a user should enter that stations' AUTHORIZE CODE to retrieve.
- While retrieving messages, the User can rewind messages as allowed by the Rewind Message Timer (PGM 181 - FLEX 17).

Admin Programming

- VMIB Message Type (PGM 111 - FLEX 17)
- VMIB Access (PGM 113 - FLEX 2)
- VMIB MSG Retrieve PASSWORD (PGM 113 - FLEX 8)
- VMIB MSG Retrieve Date/Time (PGM 113 - FLEX 9)
- VMIB Forward No Answer Timer (PGM 181 - FLEX 1)
- VMIB User Record Timer (PGM 181 - FLEX 3)
- VMIB Valid User Message Timer (PGM 181 - FLEX 4)
- VMIB Rewind Message Timer (PGM 181 - FLEX 17)

Remote Control

An outside caller through DID/DISA can access the voice mail system after calling a station which is in a VMIB Forward mode. Entering VMIB controlling mode, the user can retrieve received messages, change user greeting, release Call Forward to VMIB, etc.

An ICM caller also can access VMIB after calling a station which is in VMIB Forward mode. Entering VMIB controlling mode, the user can retrieve received messages, change user greeting, release Call Forward to VMIB, etc.

Operation

To enter Remote VMIB Control mode:

1. Dial the station number forwarded to VMIB from an external party with DID/DISA.
2. The User greeting plays. While the user greeting is playing, press the * key; the "Enter your password" prompt is played.
3. Enter the password (authorization code) and press the # key (if authorization code is longer than 5 digits, the # key is not needed).
4. A message plays that tells you the number of messages present in the inbox.
5. Press the appropriate number (refer to values):
    - 1 = to retrieve voice messages
    - 2 = to listen or change user greeting
    - 3 = to release Call Forward to VMIB mode
    - 4 = to exit remote VMIB Control mode

To listen to the received messages:

1. Dial 1 in the main menu of Remote VMIB control mode.
2. The recorded time & date and recorded message should be heard.
3. Press the appropriate dial number (refer to values).
    - 1 = to listen to the current message again
    - 2 = to listen to the next message
    - 3 = to delete the current message
    - 4 = to delete all received messages

To change the user greeting:

1. Dial 2 in the main menu of Remote VMIB control mode.
2. While the User greeting is being played, press the # key to record a new user greeting.
3. Record the new user greeting.
4. Press the * key when the recording is finished and you are returned to the main menu.

To release Call Forward to VMIB mode:

1. Dial 3 in the main menu of Remote VMIB control mode.
2. The VMIB forward mode of the station is released.

To exit VMIB Control mode:

Dial * in the main menu of Remote VMIB control mode.

Conditions

- If you press the * key while operating in a sub-menu, the system goes to the main control menu.
- If you do not enter any digit for the length of time set for the Inter-digit timer, the connection is dropped automatically.
- If VMIB User Record Timer expires while recording a user greeting, the recording is finished and you are returned to the main menu.

Admin Programming

- VMIB User Record Timer (PGM 181 - FLEX 3)
- Inter-digit Timer (PGM 181 - FLEX 8)
- VMIB Message Rewind Timer (PGM 181 - FLEX 17)

Message Forward Enhancement

This feature allows a mailbox owner to forward a message to another mailbox.

Operation

For 2-hour version of VMIB:

No change to current operation: copying a message to another mailbox will result in the message being deleted from the original mailbox.

For 8-hour version of VMIB:

Forwarding a message to another mailbox will no longer have the effect of deleting it from the current subscriber's mailbox. The subscriber will take independent action to save or delete the original message.

For external VMIB access, the following option will be available:

- When a subscriber is listening to a message, they can press the #6 to forward the message.
- Subscriber will hear the prompt "To forward this message, enter the mailbox number"
- Once the mailbox number is typed by the subscriber, they will hear, "Message has been forwarded to" [mailbox number].

New prompts are:

- To hear the current message, press one,
- To hear the next message, press two,
- To delete the current message, press three,
- To delete all listed messages, press four,
- To reply to the current message, press five,
- To forward the current message, press six,
- For the main menu, press star.
- Please enter the mailbox number
- To forward this message, enter the mailbox number
- Message has been forwarded to [mailbox number]

Reply to Message

Description

This feature allows a caller listening to a message to press a soft key or a DTMF key to reply to the internal station that left the message.

Operation

24-button DKTU or IPKTU:

When a subscriber is listening to a message, they can scroll using the right NAV key to the reply function and press the soft key.

Operation

Any phone:

The reply to message feature will be available to all callers when logged into their mailbox, using any of the following access methods:

- ICM:

  DKTU 8 btn; DKTU 24 btn; SLT (all types); IP phone (all types)
- CO caller:

  Any CO call on any line type including SIP & Analog
- Networked caller:

  Any net call from any phone

When a subscriber is listening to a message, or after the message has been played, they can press (#5) to reply to the message. If the message is from an internal station, the subscriber will be transferred to that station's VMIB mailbox.

Conditions

- The reply feature can be utilized only to reply to voice messages from internal callers.
- If the message is from an external call, the subscriber will be played the prompt: "Not available."

Two-way Recording

Two-way Recording via SMDI

This feature allows a station to record a conversation into the mailbox by pressing a {RECORD} button while the station is talking with a CO party.

Operation

To set a {RECORD} flexible button programmed for the Two-way record feature:

1. Press the [TRANS/PGM] button.
2. Press the FLEX button (to be assigned).
3. Press the [TRANS/PGM] button.
4. Dial 54.
5. Press the [HOLD/SAVE] button.

While a station user is in a conversation with a CO line caller, the user can press the {RECORD} button and the conversation will be recorded in the user's mailbox. The user presses the {RECORD} button again or hangs up to end the recording.

Protocol: When the User presses the {RECORD} button, the System sends a SMDI message to Voice Mail PC through RS-232C cable. The format is the same as follows.

=>"crlfMD0010mmmmH0xxxxxxxxxbbcrlf^Y"

cr : carriage return,

lf : line feed,

mmmm : VM port number,

H : Action code for recording,

xxxxxxxxx : extension which try to record,

b : ascii space.

Conditions

- When the recording feature is enabled, the {RECORD} button will flash at 240 ipm. If it is disabled, the {RECORD} button will not be lit.
- The recording feature is not available to SLT.
- The recording operation is canceled when station goes off-hook, presses the {RECORD} button again, presses the [FLASH] button, or the CO party hangs up.

- This feature is available in the SMDI mode only, not in the DTMF mode.
- Not available for intercom call recording.
- If the system has a VMIB, the conversation will be recorded to the VMIB.
- If Pole 3 of DIP SW1 of MBU is set to OFF (down position) and system has VMIB, the conversation will be recorded to the VMIU.

Admin Programming

- Two Way Recording (PGM 112 - FLEX 10)

Two-way Recording via VMIB

This feature allows a station to record a conversation in the mailbox by pressing a {RECORD} button while the station is talking with a CO party.

Operation

To set a flexible button programmed for 2-way record feature,

1. Press the [TRANS/PGM] button.
2. Press the flexible button to be assigned.
3. Press the [TRANS/PGM] button and dial "54".
4. Press the [HOLD/SAVE] button.

While a station user is in a conversation with a CO line caller, the user presses the {RECORD} button and the conversation will be recorded in the user's mailbox. The user presses the {RECORD} button again or hangs up to end the recording.

Conditions

- During the recording feature is enabled, the {RECORD} button will flash at 240 ipm and if it is disabled, the {RECORD} button will be extinguished.
- Not available to SLT.
- Recording operation is canceled when station goes off-hook, presses the {RECORD} button again, presses the [FLASH] button, or the CO party hangs up.
- Not available for intercom call recording.
- If the system has an external voice mail system, the conversation will be recorded to the external voice mail system.

Admin Programming

- Two Way Recording (PGM 112 - FLEX 10)

VMIB Announcement for Auto Attendant

Incoming CO calls may be answered by the VMIB and rerouted to another station with CCR when the Attendant does not answer the call until No Answer Timer expires or when the Attendant is busy.

Operation

To operate Auto Attendant:

When an incoming call is received at the Attendant station and the call is not answered before the No Answer Timer expires, the call is forwarded to the Auto Attendant. The caller will hear a VMIB message and can reroute to another user using CCR.

Conditions

- Not available for recall and transferred calls.
- CO ringing should be assigned to only the Attendant.

Admin Programming

- Auto Attendant VMIB Announce Number (PGM 165)

VMIB Message Transfer

A message at a station may be transferred to another station.

Operation

To transfer a message to the other station:

While listening to a message, dial the Station number to be transferred to. The message will be transferred to the Station.

Conditions

- An SLT with MSG wait lamp can also transfer VMIB messages.
- The transferred station must have VMIB access.
- A user has a possibility to add an additional voice message when he transfers a voice message to another station. (possible with 7224D which has 3 soft keys).

Admin Programming

- VMIB Access (PGM 113 - FLEX 2)

VMIB Message with CLI

When an outside caller leaves a message, the CLI is saved with message. The CLI is displayed when playing the message, and the station user can make a callback using the CLI.

Operation

Call back when playing a message:

Press the [CALLBACK] soft button. The system dials the displayed CLI automatically.

Conditions

- It is possible only with the 7224D which has 3 soft keys.
- Though a user makes the callback using the CLI, the VMIB message is not deleted.

VoIP Service

Call by IP Address

This feature is established by receiving IP numbers or dialing IP numbers directly.

Operation

To make an IP call by address:

1. Press the desired {CO} line, {POOL}, or {LOOP} button.

   -or-

   Dial the CO line or Group code.
2. Dial tone sounds, then dial the IP address (use the * key in place of where the dots "." of the IP address would be used). Ex., IP Address: 156.147.3.201, Dialed number:156*147*3*201)
3. Press the # button to place the call. If the Called Party is a user in the SBX IP 320 System, ringing will follow the pre-defined Ring Assignment (PGM 144).

Conditions

- When programming Speed Dial for Direct Call, the # key must not be used (ex., to assign 156*147*3*139 to Speed Dial, enter 156*147*3*139, and not 156*147*3*139#).
- In DISA incoming calls, CO access is denied if the line seized by dialing Co access Code is a VOIB line.

Admin Programming

- CO Line Service Type (PGM 140)
- VOIB IP Setting (PGM 340)

Call by Routing Table

This feature is established by dialing Station numbers as programmed in the Network Routing Table (PGM 324).

Operation

To use the Call by Routing Table:

Dial the Station number (included in the range from start range to end range in Network Routing Table).

The System will select one VOIP CO line in the CO Group assigned in Network Routing Table.

The User will hear a Ring Back Tone if this call is possible.

In case of error, busy, or no answer, the call will follow the DID/DISA destination (PGM 167).

Conditions

- For calls using the Network Table, VOIB calls follow the ISDN DID Call Procedure (DID Conversion Type, Digit Conversion Table, Flexible DID Table, etc.).
- VOIP CO calls do not follow the assigned Ring Assignment even if DID/DISA destinations are forwarded to the Attendant. The call will be transferred to the Attendant directly.
- If the Network Routing Table has more than one table entry that is the same routing number but the destination is different, the call will be routed to the default destination of the Network Routing Table.
- To transfer incoming calls to another system via VOIB, the user must not drop the call before hearing the Ring Back Tone.
- DSS, HUNT, VMIB, VMIB # and System Speed can be the destination for the Flexible DID Table (PGM 231).
- CLIP and COLP are not applied to VOIP CO Calls.

Admin Programming

- CO Line Service Type (PGM 140) - ISDN DID/MSN
- VOIB IP Setting (PGM 340)
- DID Conversion Type (PGM 143 - FLEX 4)
- Flexible DID Table (PGM 231) - (used in case of DID Conversion Type 2)

- Networking CO Group (PGM 322 - FLEX 1)
- Networking Routing Table (PGM 324)

Early H.245

The early H.245 option allows the H245 channel to be setup earlier, and hence speeds up the call setup. When early H245 is turned on, the H245 channel address is also supplied in the Setup message. This speeds up the call because the H245 channel negotiations can proceed in parallel to H225.

Admin Programming

- Early H.245 (PGM 340 - FLEX 19)

H.245 Tunneling

Entities in the signalling path such as gatekeepers may perform functions such as divert on no-reply or other advanced call control that results in representing to an endpoint a Q.931 call state that is different from the actual call state at the other endpoint. Such intermediate entities shall ensure that H.245 messages encapsulated in Q.931 messages are forwarded to the other endpoint even if the Q.931 message in which the H.245 message is encapsulated would be consumed and not forwarded to the other endpoint. This is accomplished by transferring the encapsulated H.245 message into a FACILITY message with the h323-message-body set to empty. For example, if a gatekeeper has already sent a CONNECT message to a calling endpoint and later receives a CONNECT message from a called endpoint that contains an encapsulated H.245 message, it must forward the H.245 message using a FACILITY message.

Admin Programming

- H.245 Tunneling (PGM 340 - FLEX 20)

Normal/Fast mode for H.323

H.323 endpoints may establish media channels in a call by using either the procedures defined in Recommendation H.245 or the "Fast Connect" procedure described below. The Fast Connect procedure allows the endpoints to establish a basic point-to-point call with as few as one round-trip message exchange, enabling immediate media stream delivery upon call connection.

The calling endpoint initiates the Fast Connect procedure by sending a SETUP message containing the fastStart element to the called endpoint. The fastStart element consists of a sequence of OpenLogicalChannel structures describing media channels which the calling endpoint proposes to send and receive, including all of the parameters necessary to immediately open and begin transferring media on the channels.

Admin Programming

- Normal/Fast Mode (PGM 340 - F 18)

TOS for H.323

In the IP Datagram, 8 bits are reserved to the Service type. These bits can be further broken into 5 subfields given below.

0	1	2	3	4	5	6	7
PRECEDENCE			D	T	R	UNUSED	

The 3 precedence bits specify the datagram precedence varying from 0 to 7. This allows the senders to indicate the importance of each datagram. The TOS provides a mechanism that can allow control information precedence over data. The other 3 bits represent the following:

D - requests low delay

T - requests high throughput

R - requests high reliability

The precedence bits (bits - 0,1,2) are significant in having an effect on load. So, we vary these bits and try to monitor the changes in the RTT (Round Trip Time) for varying loads.

At low loads, the traffic is not significant enough for the TOS to have any effect on the ping as the packets will get through anyway. Similarly at high loads, the TOS will not have an effect on the ping statistics but in this case there's too much traffic & there will be a significant amount of packet loss & elongated round trip times irrespective of the precedence set. Somewhere in between these two extremes we expect to see a window where the TOS actually has an effect on the RTT/packet loss (high RTT/Packet loss at TOS=0 & low RTT/Packet loss at TOS=224).

Executing "ping2" for different TOS with no load.

The modified NIKHEF ping code was run from nereus.slac.stanford.edutonocdev1-qos.es.net with TOS: 0, 32, 64, 96, 128, 160, 192, 224 & No TOS specification. The code was run for 10,000 packets of each TOS. The TOS numbers represent the precedence bits as follows.

PRECEDENCE	BITS (0,1,2)	TOS
0	000	0
1	001	32
2	010	64
3	011	96
4	100	128
5	101	160
6	110	192
7	111	224

Precedence 0 (TOS 0) is NORMAL PRECEDENCE (low precedence) while precedence 7 (TOS 224) is called network control (High precedence).

Since there is no load on the line, one would expect the TOS to have no effect on the ping statistics. Rightly so, we see from the chart that the obtained results are similar to the predicted results.

Admin Programming

- Precedence value (PGM 340 - FLEX 21)

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