

iPECS -MG

Feature Description & Operation Manual

Revision History

ISSUE	DATE	DESCRIPTION OF CHANGES
1.0	2009.12	Initial Release
1.1	2010.08	Changed the new CI (LG-Ericsson)

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1. INTRODUCTION

1.1 MANUAL APPLICATION

This document provides detailed information covering description and operation of the numerous features available in the iPECS-MG System Software.

1.1.1 Organization

Features are arranged alphabetically in eight different major groupings as follows:

- Section 2 – Directory Number (DN)
- Section 3 – System Features
- Section 4 – Intercom Features
- Section 5 – CO/IP Features
- Section 6 – Digital Phone Features (Multi-function Phone Features)
- Section 7 – Attendant Features
- Section 8 – SLT Features
- Section 9 – SIP Features

1.1.2 Feature Information

Each section is an alphabetical listing of features with the description and operation of each. The structure is divided into 6 parts as below:

- **Description:** explains the nature of the feature.
- **Operation:** gives detailed step-by-step operation of the feature for Digital Phones and SLTs.
- **Conditions:** explains known feature interactions and constraints related to the feature.
- **Programming:** lists database entries that may be required for proper feature operation.
- **Related Feature:** lists related topical information to aid in understanding the feature.
- **Hardware:** lists hardware required for proper feature operation.

iPECS - MG Release 1

Feature Description & Operation

Issue 1.1

1.2 SYSTEM CAPACITIES

The iPECS-MG Series is available in several configurations as listed in Table 2.2.4-1. Total port capacities range from the 50 channel at iPECS-MG 100 to 400 channel at iPECS-MG 300.

Table 2.2.4-1 System Capacity Chart

Items	iPECS-MG 100	iPECS-MG 300
KSU No.	2	3
Slot No. per KSU	6	6
Total Port (Extension + CO line)	200	414 (if IP Phone/DECT not included)
		564 (if IP Phone/DECT included)
Number of extension Port	120	324
Number of extension	180(Ext 120 + DN 60)	648(324 x 2)
Number of CO Line	80	240
Number of Tenant Group	5	9
Numbering Plan	Extension: 8Digits	Extension: 8Digits
	Feature: 8Digits	Feature: 8Digits
	Trunk: 8 Digits	Trunk: 8 Digits
Attendant	5/Tenant	5/Tenant
DSS/BLF Console	5	5
Member of conference	13 party	13 party
Internal Page zone	15	30
System speed dial	1000	2000
	(32 digits)	(32 digits)
Station Speed Dial	50 (32 digits)	50 (32 digits)
Call Log (Outgoing/Incoming/Missed Call)	100 (32 digits) (Not protected)	100 (32 digits) (Not protected)
Save Number Redial(SNR)	1 (32 digits)	1 (32 digits)
Number of SMDR Records	5000	5000
Authorization Code	Max. 12Digits 180: Extension	Max. 12Digits 648: Extension
CO Group No	24	72
Station Group	20 (50 member/Group)	50 (50 member/Group)
Pickup Group	20 (100 member/Group)	50 (100 member/Group)
Command Call Group	10 (12 member + 1 initiator/Group)	10 (12 member + 1 initiator /Group)
Interphone Group	10 (10 member/Group)	10 (10 member/Group)
Page Group	15 (50 member/Group)	30 (50 member/Group)
PTT Group	10 (50 member/Group)	10 (50 member/Group)
Conference Room	9	9

iPECS - MG Release 1

Feature Description & Operation

Issue 1.1

Items	iPECS-MG 100	iPECS-MG 300
Number of Hot Desk Agent	60	324
Station Name Information	16 Characters	16 Characters
Digit Restriction	COS: 16	COS: 16
	Allow/Deny Entry per COS: 100	Allow/Deny Entry per COS: 100
	Max. Digit: 16	Max. Digit: 16
Digit Translation	Table No: 9	Table No: 9
	Number of Digit: 16	Number of Digit: 16
	300 per 1 table	300 per 1 table

2. DIRECTORY NUMBER (DN)

Directory Number (DN) is the telephone number for internal users, which can be used exclusively by only one station or can be shared by multiple stations.

The basic idea of providing the DN feature is not only to assign one telephone number to one physical telephone, but also to allow one telephone number to be shared and used by multiple physical stations.

So, if one DN is contained by a variety of multi-functional terminals or analogue telephones, all those physical stations can be assumed as one logical station as if several analogue telephone sets can be connected to one physical line.

In another way, one physical telephone can have multiple DNs allowing for additional outgoing and incoming calls.

2.1 TERMS

TYPES OF DN

- SADN-NORMAL : Single-Assign Directory Number (SADN) that can be used by only one station.
- SADN-HOTDESK : Single-Assign Directory Number (SADN) for Hot Desk Usage
- MADN : Multi-Assign Directory Number (MADN) that can be used by one or multiple stations.

CATEGORY OF DN

- My-DN (M-DN): each station must have at least one unique number that cannot be used by another station (minimum requirement, automatically assigned by board configuration). Otherwise, it is not possible to make outgoing calls or receive incoming calls.
- Sub-DN (S-DN): station can have more numbers but M-DN. All numbers except M-DN are called S-DN.

NOTE: S-DN can be shared by other stations if it is MADN type.

PRIME DN

- If multiple numbers are used by a station, one DN can be selected to have higher priority over others. When only one number exists, it becomes P-DN, which will be seized first for outgoing calls, answered first if there's are multiple incoming calls, and used for idle status display for DND, Forward, Absent Message and so on that can be set independently for each DN.

Related Programming

System Data 1. Numbering Plan, Station Number (PGM 112)

Station Data 1. Station Number Type (PGM 130 – Index1)
 2. MADN Member (PGM 130 –Index2)
 3. Prime Number Button (PGM123-Index1)

2.2 BASIC FEATURES

To use DN features, or to receive incoming calls or make outgoing calls, DN must be programmed on a Flex button, except in the case of an analog phone (does not have flex buttons). One DN is stored in each station by default, which is M-DN. If there is no DN button at all, it is not possible to call a number or get a call from others.

2.2.1 Making Calls

When making outgoing calls, a Station User can select a DN number either by pressing the appropriate DN flex button, by going off-hook using the handset, or by dialing while the phone is on-hook.

P-DN is seized automatically if the DN button is not explicitly pressed as in the case of going off-hook or on-hook dialing. However, if the P-DN is busy at that time (P-DN can be shared by other stations), the first idle DN button is selected in the order of button number (button 1 first, button 2 second, etc.).

Once a DN is selected for an outgoing call, the tenant group number, calling station number, CLI, COS and other DN-related information are applied for the duration of the call. For example, if a different DN is selected for two outgoing calls, it is possible to have different tenant groups or COS for each call.

2.2.2 Receiving Calls

A physical station can receive additional calls showing on DN buttons, or through other available DN numbers that are stored in that station, even while on a call. However, if the DN is in use, it is not possible to receive a call through that DN number.

NOTE: the status of a physical station and each DN is maintained independently.

If the Station is idle, the normal ring will be provided. Otherwise, off-hook signaling is activated.

The Station User can answer an incoming call by pressing the flashing DN flex button, or by going off-hook without selecting a DN flex button. If there are multiple incoming calls at the same time, going off-hook allows the user to seize P-DN if it is ringing, or to seize first ringing DN in the order of button number (button 1 first, button 2 second, etc.).

Even when there are incoming calls at a station, the User can make an outgoing call by pressing an idle DN button and dialing the called party number.

2.2.3 P-DN feature

P-DN is automatically seized first when a station user goes off-hook or dials while on-hook when receiving or making calls. P-DN can be either M-DN or S-DN. If P-DN is not assigned explicitly, the first DN button becomes P-DN in the order of button number.

The same DN can be used as P-DN for multiple stations. If the state of shared P-DN is changed in this case, the status of P-DN will be updated to all the stations that have the shared P-DN. For example, DND, call forward, and other DN-based status notifications will be displayed at all same DN-programmed stations.

2.2.4 LED of DN button

LED states of DN buttons are as follows:

- Green ON: DN being used by my station
- Red ON: DN being used by another station
- Amber flash: Held DN
- Green Flash: Ringing DN
- Red Flash: DN in DND status or DN in Call Forward

2.2.5 DN Tenant Group/COS

Each DN can be programmed with its own Tenant group or COS information. So, Tenant group and COS can be different for each call depending on the DN used for the calls.

If a station has DN buttons with different tenant groups, the station can make and receive calls using the different tenant groups.

Additionally, if the DN buttons have different COS, a station can have different COS according to the DN button selected.

2.2.6 Branch Line

When a station is using a MADN-type DN, other stations cannot access the same DN. However, if a branch line option is set to the DN, another Station can access the busy DN interrupting its call and establishing a conference call for all users.

2.2.7 Incoming Ring Option

When multiple stations have the same DN button, each station can have a different ring delay option.

- Immediate Ring: Ring signal is sent to station with no delay.
- Delayed Ring: Ring signal is sent to station after the programmed delay.
- No ring: Ring signal is not sent to station, but only LED flashes.

When the DN receives an incoming call, the DN button LED will flash Red regardless of the ring delay option. However, the LCD of a station modified to display the incoming call after it receives a ring signal can be automatically answered just by going off-hook. However, before the station receives a ring signal, the incoming DN call cannot be answered automatically by going off-hook, but the station user should press the flashing DN button manually.

2.2.8 Access Option

When a station has multiple DN buttons, each DN button can have a different access option.

- All Call: No restriction.
- Dial After Seizure: No restriction about incoming ringing, but when making outgoing calls with this button, user should seize the DN by pressing this button even if this button is assigned to prime number button.
- Incoming Only: Outgoing call is not possible with this button.

When there is incoming call to a DN, the DN button LED will flash in red color regardless of the ring delay option. However, the LCD of station is changed to display the incoming call after it receives ring signal and the call can be answered automatically just by going off-hook. However, before the station receives ring signal, the incoming DN call cannot be answered automatically by going off-hook, but the station user should press the flashing DN button manually.

3. SYSTEM

3.1 ACCOUNT CODE

Description

Station users may enter a non-verified variable length (up to 12 digits) identifier for tracking specific calls. The identifier or "Account Code" is output as part of the Station Message Detail Record (SMDR) for the call.

Operation

Digital Phone

To assign a Flex button for {ACCOUNT CODE} operation:

Press [PGM] + {FLEX} + Button Feature Type (1) + {Account Code Feature Code}
+ {Account Code + *} + [SAVE]

To enter an Account Code using an {ACCOUNT CODE} button prior to placing a call when account code is not entered in the button:

1. Lift the handset.
2. Press the {account code} button.
3. Dial the Account Code (1 to 12 digits).
4. Press '*'; Intercom dial tone is heard.
5. Place the CO/IP call as normal.

Using the programmed {ACCOUNT CODE} button prior to making a call:

1. Lift the handset.
2. Press the {account code} button; Intercom dial tone is heard.
3. Place the CO/IP call as normal.

Using an {ACCOUNT CODE} button during a call:

1. Press the {account code} button; CO line is held and the station hears dial tone.
2. Dial the Account Code (1 to 12 digits).
3. Press '*'; Station is re-connected with CO line.

SLT

To enter an Account Code prior to placing a call:

1. Lift the handset.

2. Dial the {Account Code Feature Code}
3. Dial the Account Code (1 to 12 digits).
4. Press '*'. .
5. Place the CO/IP call as normal.

To enter an Account Code during a call:

1. Press for Hook-switch.
2. Dial {Account Code Feature Code}
3. Dial the Account Code (1 to 12 digits).
4. Press '*'. .

Conditions

1. If an Authorization Code is entered as the Account Code, the SMDR record will show the station number or the bin number for a System Authorization Code rather than the user entered Authorization Code.

Programming

Numbering

1. Feature Numbering Plan (PGM 113)

Related Features

Authorization Codes (Password)
Station Message Detail Recording (SMDR)
Station Flexible Buttons

Hardware

3.2 ALARM SIGNAL/DOOR BELL

Description

The system can be configured to recognize the status of an external contact (normally open or closed). The system will signal the assigned station when the contact activates. This capability is commonly employed to provide remote Alarm or Door Bell signals to the user.

A station receives the Alarm Signal, either as a single tone burst repeated at 1-minute intervals or a continuous tone. The Alarm Signal may be terminated at the User's phone by dialing the Alarm Stop code, or pressing the {ALARM STOP} button if assigned. To rearm the Alarm function, the alarm condition must be cleared and the Alarm signal terminated.

When used as a Door Bell, assigned stations receive an Alarm Signal each time the external contact is activated; reset is not required.

Operation

System

At detection of contact operation, the Alarm/Door Bell signal is sent to assigned station.

Digital Phone

To assign a Flex button as an {ALARM STOP} button:

Press [PGM] + {FLEX} + Button Feature Type(1) + {Sys Alarm Reset Feature Code} +
[SAVE]

To terminate an Alarm Signal while idle:

1. Dial the {Sys Alarm Reset Feature Code}; a confirmation tone is received and the Alarm Signal is terminated.

NOTE: If the alarm condition is cleared, the system will automatically rearm the alarm monitoring.

OR

1. Press the programmed {ALARM STOP} Flex button.

Conditions

1. The Alarm contacts must be “dry”, no voltage or current source connected.
2. A station with LCD assigned to receive Alarm/Door Bell signals will show “ALARM” as appropriate.
3. If alarm is active during station busy, mute ring will be served to assigned station, and then after conversation, when station go to idle, the alarm signal will be sent to assigned station again.
4. Assigned stations can be changed using Alarm Assign. (PGM121-Flex12)
5. Only Stations assigned with Alarm ring can terminate the alarm signal.
6. LIP Phone and normal LKD Phone stations can be assigned as alarm stations.
7. In signal mode, station will return alarm ringing again if an assigned station user does not reset the alarm signal prior to the station returning to idle.
8. When the alarm is ringing, the alarm signal must be reset so phone operation will be fully functional (fixed or flex buttons do not operate and the user cannot hear the dial tone during alarm ringing).

Programming

Station Data

1. Alarm (PGM 121-Index12)

System

1. Alarm Enable (PGM 227 - Index1)
2. Alarm Contact Type (PGM 227 - Index2)
3. Alarm Mode (PGM 227 - Index3)
4. Alarm Signal Mode (PGM 227 -Index4)

Related Features

Door Open

Hardware

Digital Phone

External contact connected to Alarm input of MPB, refer to iPECS-MG **Hardware Description & Installation Manual**.

3.3 AUTHORIZATION CODES (PASSWORD)

Description

An Authorization Code is tied to a DN, and provides a means to control access to Walking COS, or DISA and may be required for outgoing CO/IP Lines based on the configuration of the database. When users dial a valid Authorization Code, the system invokes the Station COS.

The Station Authorization Code includes the associated station number and the assigned code. A Station Authorization Code is specifically related to a given station and intended for a single user.

The Administrator and Attendants are permitted to assign any Authorization code including codes for another station. Normal users may only assign the Station Authorization code for the specific station.

Operation

Digital Phone

To assign a Station Authorization Code:

1. Press the **[PGM]** button.
2. Dial '34'{Authorization Code Program}.
3. Dial the Authorization Code (1-12 digits).
4. Dial '*' or Press the **[SAVE]** button to save.

SLT

To assign a Station Authorization code:

1. Lift the handset.
2. Dial {SLT Program Mode Entry code}.
3. Dial Station User Program code 34.
4. Dial Authorization Code (1-12 digits).
5. Dial *

System Attendant

To assign an Authorization Code:

1. Press the **[PGM]** button.
2. Dial Attendant Station Program Code 033.
3. Dial Station number for a Station code.
4. Dial the Authorization Code.
5. Press the **[SAVE]** button.

Conditions

1. A user may enter an Authorization Code from any station to place a CO/IP call using Walking COS.
2. An Authorization code may include any dial pad digit except '*' and '#'.

Programming

- | | |
|---------------------|------------------------------|
| Station Data | 1. Password (PGM 131-Index4) |
|---------------------|------------------------------|

Related Features

Hardware

3.4 AUTO CALL RELEASE

Description

Intercom calls (except Hands-free Ring Back) will be released automatically if the called party does not answer during the pre-set time.

Operation

System

Auto Call Release of Intercom calls:

If a station places an intercom call and the called station does not answer in the Intercom Call Release Time, the call is terminated and the calling user receives an error tone.

Conditions

1. When the handset is used to place a call, the user will receive an error tone for 30 seconds followed by 30 seconds of Howler tone and the station is placed in a fault mode. If on-hook dialing is used, the station will receive an error tone for one (1) second and the phone will return to idle automatically.

Programming

1. **Table Data** System Ring Table: Normal Call Ring (PGM 265-Web admin)

Related Features

Howler Tone

Hardware

3.5 AUTOMATIC PAUSE INSERTION

Description

In addition to a manually entered Pause, the system will automatically pause dialing to allow for potential connection delays. The pause will be inserted when any of the following occur:

- Flash is encountered in a Speed Dial number,
- Pulse to Tone Switchover is encountered in a Speed Dial or Redial number.
- Connect message is received on an ISDN Line.

Operation

System

The system automatically pauses dialing after an appropriate event (as listed above).

Conditions

1. An automatically inserted pause is not counted as a digit in a Speed Dial number.
2. The LCD of the Digital Phone will show a "P" when a pause is encountered.
3. When the System inserts a Pause, "P" indication is not shown.

Programming

Related Features

- Auto Called Number Redial (ACNR)
- Last Number Redial (LNR)
- Dial Pulse to Tone Switchover

Hardware

3.6 AUTOMATIC PRIVACY/BRANCH LINE

Description

Privacy is insured in all communications on the system. If desired, the customer may elect to disable the Automatic Privacy feature, allowing an uninvited station to join in an existing external conversation. In such a case, a conference is established.

The Privacy feature restricts the intrusion/call-wait/camp-on/OHVA at a busy station, while the Branch Line can restrict a conference call by pressing {DN} button in use.

Operation

Digital Phone

To intrude into a call when Privacy is disabled:

1. Make a call to busy station and then press the appropriate code for intrusion/call-wait/camp-on/OHVA when receiving busy tone.

To change privacy mode in conversation:

1. Press the [DND] button during a conversation.

To intrude into a call when Branch Line is enabled:

1. Press a busy (lit steady) {DN} button, the user is connected to the call with the existing internal station user.

Conditions

1. With Automatic Privacy disabled, privacy is still assured on all intercom and conference calls.
2. Only one station can intrude on an active call.
3. An intrusion tone can be provided to the call indicating another station has accessed the line.

Programming

- Station Data**
1. Branch Line (PGM 134-Index10)
 2. Auto Privacy (PGM 134-Index11)

- System Data**
1. INTERCOM BUSY ONE-DIGIT SERVICE (PGM 237)

Related Features

Multi-Party Voice Conference
Station Flexible Buttons

Hardware

3.7 AUTO SERVICE MODE CONTROL

Description

The service mode defines different ring assignments, COS and answering privileges for the system. The service mode can be controlled automatically through definitions in the Auto Ring Mode Selection Table, which defines the time of day for Day, Night and Timed shift modes. The Attendant may change the system mode selection from automatic to manual.

Operation

System

Operation of this feature is automatic.

Conditions

1. If the system has Holiday information and current mode is Holiday, service mode is operated as Night mode

Programming

- Table Data**
1. System Time Table (PGM 253)
 2. Weekly Time Table (PGM 254)
 3. Holiday Time Table (PGM 256)

Related Features

Direct Inward System Access (DISA)
Day/Night/Timed Ring Mode
CO Ring Assignment
LBC (Loud Bell Control)
Dialing Restrictions

Hardware

3.8 AUTOMATIC SYSTEM DAYLIGHT SAVINGS TIME

Description

The system can automatically adjust for Daylight Saving Time (DST). When DST is enabled, the system will adjust the system time forward one hour at the DST Start time and back one hour at the DST End time. The system time is sent for display to all devices and terminals and is the basis of the various system time-based features (Wake-up Alarm, etc.).

Operation

System

Operation of this feature is automatic.

Conditions

1. The DST Start and End times are set by the Web Admin interface only.
2. The interval between the DST Start and End times must be at least 7 days.

Programming

SYSTEM

1. System Time (PGM 233–Index1)
2. System Date (PGM 233–Index2)
3. DST Enable (PGM 233–Index3)
3. DST Start & End Time (Web Admin.)

Related Features

Auto Service Mode Control
Automatic System Time Synchronization
System Clock Set

Hardware

3.9 AUTOMATIC SYSTEM TIME SYNCHRONIZATION

Description

When enabled, the system automatically determines and sets the time of day employing Network Time Protocol (NTP) or ISDN time messages. When using NTP, the system requests and receives GMT time at ten-minute intervals from the specified NTP time server. This feature allows the System Time to synchronize with the NTP time server automatically. If the time deviates more than two seconds, the system clock is adjusted to match the NTP server.

When using ISDN, the system receives the time of day in ISDN messages and automatically adjusts the time if the system time deviates from the ISDN time.

Operation

System

Operation of this feature is automatic.

Conditions

1. NTP packets are expected over UDP port 123; verify the port is open and available.
2. A secondary NTP server address can be defined should the first server not respond.
3. If set, the system adjusts for the local time zone assigned in the system as the Standard System Time as well as Daylight Savings Time (DST).

Programming

System

1. Network Time/Date (PGM 223 – Index5)
2. NTP Active (Web Admin.)
2. NTP Server address (Web Admin.)
3. Std. System Time, Local Time Zone (Web Admin.)

Related Features

Auto Service Mode Control
Automatic System Daylight Savings Time
System Clock Set

Hardware

3.10 BATTERY BACK-UP, MEMORY

Description

The system database is protected from power-loss by a long life (10-year) lithium dry cell battery. Should local power fail, the battery will maintain the system memory and proper operation of the system clock.

Operation

System

When enabled, Operation is automatic.

Conditions

1. The Initialization switch must be in the ON position to enable Memory Battery Back-up. Otherwise, should power fail, the system will initialize the database on power-up. Refer to the ***iPECS-MG Hardware Description and Installation Manual***, Section 3.2.7.
2. The Lithium battery is not field-replaceable.

Programming

Related Features

Hardware

3.11 CALL FORWARD

Description

User may have selected incoming calls re-route to other stations (local or networked), station groups, the VMIB, or over a system CO/IP line (Off-Net).

The user selects the type and condition under which calls will forward by entering a Call Forward code as follows:

- Code 0, Remote Call Forward – forwards all calls to the station, except recalls, activated from a remote station, Call Forward, Follow-me.
- Code 1, Unconditional – all calls to the station, except recalls, are forwarded internally or externally immediately upon receipt.
- Code 2, Busy – if the station is busy, forwards all calls, except recalls, to the selected station.
- Code 3, No Answer – forwards all calls, except recalls, to the selected station when the station does not answer within the No Answer timer.
- Code 4, Busy/No Answer – forwards calls if the selected station is busy or does not answer within the No Answer timer.

Operation

Digital Phone

To activate Call Forward, Unconditional or Busy/No Answer:

1. Lift the handset or press the **[SPEAKER]** button to receive a dial tone.
2. Press the **[FWD]** button.
3. Dial 1-4 {Forward Code} as appropriate.
4. Dial the station or station group to receive calls.
OR
5. Dial CO Group Access code and desired external phone number.
6. Press the **[SAVE]** button to save.
7. Replace the handset, return to idle.

To activate Call Forward, Remote (Follow-me):

1. Lift the handset or press **[SPEAKER]** button to receive Dial tone.
2. Press the **[FWD]** button.
3. Dial 0 {Call Forward code}.
4. Dial the Station's Authorization Code (Station number + password),
5. Dial 1-4 {Forward condition}.
6. Dial the destination station or station group.
OR
Dial CO Group Access code and desired external phone number.
7. Press **[SAVE]** button to save.
8. Replace the handset, return to idle.

To deactivate Call Forward:

1. Press flashing [FWD] button, Call Forward will deactivate; [FWD] LED button is turned OFF.

SLT

To activate Call Forward, Unconditional, Busy/No-Answer:

1. Lift the handset to receive Dial tone.
2. Dial {Call Forward feature code}
3. Dial 1-4 (Call Forward code) as desired.
4. Dial station or station group to receive calls.
OR
Dial the CO Group Access code and the desired external phone number.
5. Press hook-switch to save.
6. Replace the handset, return to idle.

To activate Call Forward, Remote (Follow-me):

1. Lift the handset.
2. Dial {Call Forward code}.
3. Dial 0 {Remote Forward code}.
4. Enter the Station number {Station Authorization Code} +Password.
5. Dial 0 {Remote Forward condition}.
6. Dial the destination station or station group.
OR
Dial CO Group Access code and desired external phone number.
7. Press hook-switch to save.
8. Replace handset return to idle.

To deactivate the Call forward:

1. Lift the handset to receive a stutter Dial tone.
2. Dial {Call Forward feature code}, as desired.
3. Dial '#' to cancel Call Forward.

Conditions

1. A station receiving a forwarded call can transfer the call to the forwarding station.
2. A station, denied the use of Call Forward, will receive an error tone in response to attempts to activate Call Forward.
3. A forwarded intercom call will signal the receiving station in Tone Signaling mode, regardless of the Intercom Signaling Mode at the station.
4. Attempting activation of Call Forward will automatically deactivate any activated Display Text Message (Active Call Back or Queue requests do not cancel).
5. When Call Forward is active, a Station can make outgoing calls (internal or external).
6. For CO/IP calls, when Call Forward is manually activated, it will override any Preset Call Forward assigned for the station or CO/IP line.
7. Call Forward status is maintained in the System's non-volatile memory for protection from power outage.

8. Off-Net Call Forward of incoming CO/IP calls is essentially an automated DISA call, which will establish an Unsupervised Conference; such calls are subject to the conditions of a DISA call and Unsupervised Conference and may require entry of an Authorization Code.
9. Off-Net Forward calls are not answered until the system completes dialing of the external call. The call, internal or external, is then connected to the Off-Premise call.
10. An unlimited number of stations may be set-up in a Call Forward chain, forwarding calls from one station to the next. However, the Call Forward service counter is restricted by the 'Multi-Call Forward Service Count' attribute (tenant based).
11. No Answer Forward employs the Station No Answer Forward Timer.
12. The No Answer Call Forward Timer can be adjusted at the PGM 141-Index4.
13. A Station should have Off-Net Forward access privilege to assign a CO Access code and External Phone number to the forward destination.

Programming

- | | |
|---------------------|--|
| Station Data | 1. Call Forward Access (PGM 132-Index2) |
| | 2. Off-Net Call Forward Access (PGM 132-Index3) |
| | 3. Call Forward Assignment (PGM 143) |
| Tenant Data | 1. Multi-Call Forward Service Counter (PGM 280-Index8) |

Related Features

Station Authorization Code
DND
Dialing Restriction
Station Group
Intercom Signaling Mode
Call Forward, Preset

Hardware

3.12 CALL FORWARD, PILOT HUNT

Description

User may have selected incoming calls in his group to re-route to other stations (local or networked), station groups, or VMIIIB.

The user selects the type and condition under which calls will forward by entering a Call Forward code as follows:

- Code 1, Unconditional – all calls to the station, except recalls, are forwarded internally or externally immediately upon receipt.
- Code 2, Busy – if the station is busy, forwards all calls, except recalls, to the selected station.

- Code 3, No Answer – forwards all calls, except recalls, to the selected station when the station does not answer within the No Answer timer.
- Code 4, Busy/No Answer – forwards calls if the selected station is busy or does not answer within the No Answer timer.

Operation

Digital Phone/SLT

To activate Call Forward, Unconditional or Busy/No Answer:

1. Lift the handset or press the **[SPEAKER]** button to receive dial tone.
2. Dial {Pilot Hunt Call Forward Code}
3. Dial 1-4 {Call Forward code} as desired.
4. Dial the station or station group to receive calls.
5. Press the **[SAVE]** button to save.
6. Replace the handset, return to idle.

To deactivate the Call Forward:

1. Lift the handset or press the **[SPEAKER]** button,
2. Dial {Pilot Hunt Call Forward Cancel Code},

Conditions

1. Station Call Forward has higher priority than Pilot Hunt Call Forward.
2. To assign Pilot Hunt Call forward, the Station should be a member of the Pilot Hunt Group.
3. An external number cannot be assigned to a Pilot Hunt Forward destination.
4. If a station assigns the pilot hunt Call Forward, the pilot hunt call forward in day mode is changed.
5. The Pilot Hunt Ring access privilege can be assigned on a per Station basis; if the Pilot Hunt Ring access of the station is disabled, that station will not receive Pilot Hunt ringing.
6. If a user activates the call forward feature using the {Pilot Hunt Call Forward code}, it is applied to the Day Forward destination.

Programming

- | | |
|---------------------------|--|
| Station Data | 1. Call Forward Access (PGM 132-Index2) |
| | 2. Pilot Hunt Ring Access (PGM 134-Index6) |
| Station Group Data | 1. Pilot Hunt Group (PGM 210-211) |

Related Features

Station Authorization Code
DND
Dialing Restriction
Station Group
Intercom Signaling Mode
Call Forward, Preset

Hardware

3.13 CALL FORWARD, PRESET

Description

Call Forward, Preset calls to a Station are forwarded to a pre-determined destination assigned in the system database. Preset Call Forward can define separate treatment of CO/IP calls and intercom calls. In addition, separate busy and no-answer treatments are defined:

- Internal Unconditional – all intercom calls are immediately forwarded.
- Internal Busy – Intercom calls that encounter a busy, are forwarded immediately.
- Internal No-Answer – Intercom calls, which are not answered in the No-Answer time, or busy, are forwarded.
- External Unconditional – all external calls immediately forward.
- External Busy – external calls that encounter a busy are forwarded immediately.
- External No-Answer – external calls, not answered in the No-Answer time, or busy, are forwarded.

In addition, calls can be directly forwarded to the Users Voice Mail box using Call Forward, Preset.

Preset Call Forward condition and type can be selected as listed:

- Unconditional – all calls to the station, are forwarded internally or externally immediately upon receipt.
- Busy – if the station is busy, all calls are forwarded to the selected station.
- No Answer - forwards all calls to the selected station when the station does not answer within the No Answer timer.

Operation

System

When enabled, Operation of Preset Call Forward is automatic.

Conditions

1. A station receiving a forwarded call can transfer the call to the forwarding station.
2. Calls cannot be forwarded to a station in DND (error tone is returned).
3. Manual forward has a higher priority than Preset Forward and overrides any Preset Forward setting.
4. Preset call forward status is not shown on the Station LCD display.
5. No Answer Forward employs the Station No Answer Forward timer.
6. If Station No Answer Preset Call Forward and CO Preset Forward Ring Table is set the same, the CO Preset Forward Ring Table precedes Station Preset Call Forward.

Programming

Station Data

1. Preset Call Forward (PGM 142)
2. Call Forward No-Answer Timer (PGM 141-Index4)

Related Features

Call Forward
DND
Auto Attendant
Preset Call Forward
VMIB Integrated Auto Attd/Voice Mail

Hardware

3.14 CALL PARK

Description

A User may place (Park) an active intercom or CO/IP call in a special holding location (Park Orbit) for easy access from any station in the system. The system has 50 holding locations (Park Orbits).

Operation

Digital Phone

To park an active external call:

1. Press the **[TRANS]** button.
2. Dial {Call Park Feature Code}.
3. Dial the Call Park No (00-49).
4. Return to idle.

To retrieve a parked call:

1. Lift the handset or press the **[SPEAKER]** button.
2. Dial {Call Park Feature Code}.
3. Dial the Call Park No (00-49).

SLT

To park an active external call

1. Press the hook-switch.
2. Dial {Call Park Feature Code}.
3. Dial the Call Park No (00-49).
4. Return to idle.

To retrieve a parked call

1. Lift the handset.
2. Dial {Call Park Feature Code}
3. Dial the Call Park No (00-49).

Conditions

1. If the selected Park Orbit returns a busy signal, the user may simply dial another Park Orbit without disconnecting.
2. A Parked call will recall to the station that parked the call should the Call Park Timer expire. The normal Hold Recall process is then initiated.
3. A Parked call will indicate busy at all appearances.

Programming

- | | |
|-----------------------|---|
| Numbering Data | 1. Call Park (PGM 113) |
| Tenant Data | 1. Call Park Hold Tone Time (Web Admin. PGM 290-Index58-59) |

Related Features

Hold/Hold Recall

Hardware

3.15 CALL PICK-UP

3.15.1 Directed Call Pick-Up

Description

A station may answer (Pick-Up) incoming and transferred intercom, CO and IP calls ringing at another station. All ringing calls are subject to Directed Call Pick-up except queued Callbacks.

Digital phone users may assign a Flex button as a {DIRECTED CALL PICK-UP} button.

Operation

Digital Phone

To assign a {directed call pick-up} button:

PRESS [PGM] + {FLEX} + Button Feature Type (1) + {Direct Pickup Feature Code} + [SAVE]

To Pick-up a call ringing at another station:

1. Lift the handset or press [SPEAKER].
 2. Dial {Directed Call Pick-up code}.
 3. Dial the intercom number of the ringing station.
- OR
1. Lift the handset or press [SPEAKER].
 2. Press the {DIRECTED CALL PICK-UP} button.
 3. Dial the intercom number of the ringing station.

SLT

To Pick-up a call ringing at another station:

1. Lift the handset
2. Dial {Directed Call Pick-up code}.
3. Dial the number of the ringing station.

Conditions

1. To pick-up a CO/IP call, the station must have an idle appearance button.
2. When several calls are ringing at a station simultaneously, Call Pick-up will connect the first call received.
3. Queue callback calls are not subject to Call Pick-up (receives an error tone).
4. Only ringing intercom calls are subject to Call Pick-up; Intercom calls announced hands free cannot be picked up by another station.

Programming

- | | |
|---------------------------|--|
| Numbering Data | 1. Feature Numbering Plan (PGM 113) |
| Station Group Data | 1. Station Group Attributes, Pick-up Option (PGM 200-Index5)
2. Call Pick-up Group Attributes (PGM 204) |
| Station Data | 1. Pickup by DSS (PGM 124-Index9) |

Related Features

Intercom Signaling Mode
Group Call Pick-up

Hardware

3.15.2 Group Call Pick-Up

Description

A Station can answer (Pick-Up) incoming and transferred intercom, CO and IP calls ringing at another station in the same station group. All ringing calls, except Private Queue Callbacks, are subject to Pick-up by other stations in the same group.

Digital phone users may assign a Flex button as a **{GROUP CALL PICK-UP}** button.

Operation

Digital Phone

To assign a **{GROUP CALL PICK-UP}** button:

PRESS [PGM] + {FLEX} + Button Feature Type(1) + {Group Pickup Feature Code} + [SAVE]

To Pick-up a call ringing at another station:

1. Lift the handset or press [SPEAKER].
 2. Dial {Group Call Pick-up code}.
- OR
3. Press the programmed Group Call Pick-up button.

SLT

To Pick-up a call ringing at another station:

1. Lift the handset.
2. Dial {Group Call Pick-up code}.

Conditions

1. To pick-up a CO/IP call, the station must have an idle appearance button.
2. When several calls are ringing at a station simultaneously, Call Pick-up will connect the first call received.
3. Queue callback calls are not subject to Call Pick-up (receives error tone).
4. Only ringing intercom calls are subject to Call Pick-up; Hands free announced intercom calls cannot be picked up by another station.
5. When a station belongs to multiple groups, calls received are routed to the group with the lowest station number.

Programming

- | | |
|---------------------------|--|
| Numbering Data | 1. Feature Numbering Plan (PGM 113) |
| Station Group Data | 1. Station Group Attributes, Pick-up Option (PGM 200-Index5)
2. Call Pick-up Group Attributes (PGM 204) |

Related Features

Intercom Signaling Mode
Group Call Pick-up
Station Group

Hardware

3.16 CALL TRANSFER

3.16.1 Call Transfer, Station

Description

CO/IP calls can be transferred to other stations on the same System. Calls can be transferred without announcing the call (unscreened), or with announcement (screened).

When a call is transferred, the Transfer Recall Timer is initiated. If the timer expires before the call is answered, the Hold Recall process is initiated.

Operation

Digital Phone

While on a CO/IP call, to perform a Screened Call Transfer:

1. Press **[TRANS]**.
 2. Dial the station to receive the transfer.
 3. At answer or splash tone, announce the call.
 4. Hang-up to complete the transfer.
- OR
5. Press the **{DSS/BLF}** button for the desired station.
 6. When answered or when splash tone is heard, announce the call.
 7. Hang-up to complete the transfer.

While on a CO/IP call, to perform an Unscreened Call Transfer:

1. Press **[TRANS]**.
 2. Dial the station to receive the transfer.
 3. Hang-up to complete the transfer.
- OR
4. Press the **{DSS/BLF}** button for the desired station.
 5. Hang-up to complete the transfer.

SLT

While on a CO/IP call, to perform a Screened Call Transfer:

1. Press for hook-switch.
2. Dial the station to receive the transfer.
3. When answered or when splash tone is heard, announce the call.

4. Hang-up to complete the transfer.

While on a CO/IP call, to perform an Unscreened Call Transfer:

1. Press for hook-switch.
2. Dial the station to receive the transfer.
3. Hang-up to complete the transfer.

Conditions

1. The transferring station may camp on a call on to a busy station.
2. To prevent Toll abuse, CO/IP lines without an active call (either incoming or dialed digits on outgoing) cannot be transferred.
3. For outgoing CO Line calls, the system will monitor the CO Line for dial tone to prevent Toll abuse; when an IP Line is seized, the system does not monitor for dial tone.

Programming

Related Features

Hold Recall
Call Transfer, CO/IP
Station Flexible Buttons

Hardware

3.16.2 Call Transfer, CO/IP

Description

A Station may be permitted to transfer a CO/IP call to another CO/IP line, establishing an Unsupervised Conference between the two external parties.

If the receiving party is called through an ISDN or VoIP path, the Transfer Hold Recall Timer is initiated and if it expires, Hold Recall is initiated.

Operation

Digital Phone

While on a CO/IP call, to perform a Screened Call Transfer:

1. Press **[TRANS]**.
2. Place a CO/IP call in the normal manner.
3. When answered, announce the call.
4. Hang-up to complete the transfer.

While on a CO/IP call, to perform an Unscreened Call Transfer:

1. Press **[TRANS]**.
2. Place CO/IP call in the normal manner.
3. Hang-up to complete the transfer.

SLT

While on a CO/IP call, to perform a Screened Call Transfer:

1. Press for hook-switch.
2. Place CO/IP call in the normal manner.
3. When answered, announce the call.
4. Hang-up to complete the transfer.

While on a CO/IP call, to perform an Unscreened Call Transfer:

1. Press for hook-switch.
2. Place CO/IP call in the normal manner.
3. Hang-up to complete the transfer.

Conditions

1. A call using the service of 2 CO lines and not providing call disconnection detection will be disconnected following the expiration of the Unsupervised Conference timer.
2. The system provides Transfer Recall on ISDN and VoIP calls providing 'Answer Supervision'.
3. If during a transfer to an external party, the user presses the CO/IP line of the original call, the outgoing call is disconnected and the original call is connected to the user.
4. The CO-to-CO transfer can be enabled or disabled using the Transit Option.
5. If the transferred call is not answered by the destination Station, the call is routed to the 'Transfer No Answer Destination' of the CO Alternative Destination.
6. When CO 1 is transferred to CO 2, if the CO 2 does not answer within the CO-to-CO transfer recall timer, both CO lines are disconnected.

Programming

- | | |
|---------------------|--|
| CO Line Data | 1. CO-to-CO Attributes (PGM 179) |
| | 2. Unsupervised Conference Extend (PGM 166, PGM 171) |
| | 3. Unsupervised Conference Timer (PGM 166, PGM 171) |
| | 4. Incoming CO Alternate Destination (PGM 169) |
| | 5. Outgoing CO Alternate Destination (PGM 173) |
| | 6. Co-To-CO Transfer Timer (PGM 220) |

Related Features

Hold Recall
Call Transfer, Station
Unsupervised Conference

Hardware

3.17 CO/IP ACCESS

Description

Stations can access outgoing CO/IP lines based on CO/IP Group Access programming. Digital Phones may use flexible buttons assigned to access a specific {CO} line, using a {CO ACCESS CODE} for outgoing calls.

Operation

Digital Phone

To place an outgoing CO call:

1. Lift the handset or press the [SPEAKER] button.
 2. Press desired {CO} line, and enter {CO ACCESS CODE}.
- OR
3. Dial the CO line or CO Access Code.
 4. Dial the desired number.

To place an outgoing IP call:

1. Lift the handset or press the [SPEAKER] button.
 2. Press desired {CO} line, and enter {CO ACCESS CODE}.
- OR
3. Dial the CO line or CO Access Code.
 4. Dial the desired number registered in H.323 Routing Attribute (PGM 360), then the outgoing call will be made to the assigned IP-Address.

To receive an IP call:

1. If a call is received from the assigned IP-Address in H.323 Incoming Attribute, it is routed to the assigned incoming CO Group.

To answer an incoming CO/IP call:

1. Lift the handset or press the [SPEAKER] button.
- OR
2. Press {DN} button, and lift the handset to speak privately.

SLT

To place an outgoing CO/IP call:

1. Lift the handset.
2. Dial the CO line or CO Access Code.
3. Dial the desired number.

To answer an incoming CO/IP call:

1. Lift handset.

Conditions

1. When a user dials a {CO Access Code}, the system will search the assigned CO group for an idle CO/IP line, if there is no idle CO/IP line then the system will search the 1st CO/IP Group for an idle CO/IP line; the user will receive a busy tone.
2. A telephone user not allowed access to a CO/IP line will receive an error tone when access is attempted; the station may receive transferred calls despite denied access on the line but will not be able to flash or use the CO/IP line for an outgoing call.
3. A station denied access to a CO/IP line but assigned to receive CO/IP line calls may answer incoming calls. The user may transfer calls but cannot make an outgoing call on the CO/IP line.
4. CO/IP lines placed on hold may be retrieved by dialing the {Retrieve Held CO/IP code} and the CO/IP line number.
5. The Tx path to a station will be muted until the system has verified the Toll Restriction for the CO/IP line.
6. When a CO line is seized, the system will monitor the line for dial tone.
7. The System selects lines from a group using the Round Robin, First-Choice or Last-Choice method based on Admin Programming.

Programming

CO Line Data	1. CO Group Access Code (PGM 180) 2. CO Line Group (PGM 160-Index3-4)
H.323 Data	1. H.323 Routing Attribute (PGM 360)
Station Data	1. CO Group Access (PGM 150)

Related Features

Hardware

3.18 CO/IP CALL TIME RESTRICTION

Description

The System can be programmed to limit the length of calls at specified stations. When a specified Station places a call, the system initiates the Call Restrict timer, and 15 seconds prior to timer expiration, a warning tone is delivered. At expiration, the system terminates the call returning the external CO/IP line to idle.

Call time restriction can be applied differently according to call types (Local call, Long Distance call or international call).

Operation

System

Operation of this feature is automatic when assigned:

Conditions

1. The warning tone can be provided periodically or once as programmed.
2. Once activated, the Call Warning Tone timer continues timing while the call is connected to the system even if the call is transferred or picked up at another station.

Programming

Station Data

1. Call Duration Restrict Access (PGM 134-Index4)

Tenant Data

1. Call Duration Restriction (PGM 284-285)
2. Local Call Prefix Table (PGM 286)
3. Long Distance Call Prefix Table (PGM 287)
4. International Call Prefix Table (PGM 288)

Related Features

Hardware

3.19 CO/IP CALL WARNING TONE TIMER

Description

Stations can receive a tone indicating the elapsed time of a CO/IP call has reached the CO Warning Tone time (timer expiration). A warning tone is presented to the call parties notifying that the Warning Tone Timer is about to expire.

Operation

System

If enabled, operation of this feature is automatic:

Conditions

1. Warning tone is received 15 seconds prior to expiration of the timer and can be repeated every tone repeat time.

Programming

Station Data

1. Call Duration Restrict Access (PGM 134-Index4)

Tenant Data

1. Call Duration Restriction (PGM 284-285)
2. Local Call Prefix Table (286)
3. Long Distance Call Prefix Table (PGM 287)
4. International Call Prefix Table (PGM 288)

Related Features

CO/IP Call Time Restriction

Hardware

3.20 CO/IP QUEUING

Description

When CO/IP lines are busy, permitted users can request to be placed in queue awaiting availability of the CO/IP line or a CO/IP line in the same group. When an appropriate CO/IP line becomes available, the system calls the waiting station on a First-In, First-Out (FIFO) basis.

Operation

Digital Phone

To request to be placed in queue for a busy CO/IP line:

1. Press the desired **{CO GROUP ACCESS CODE}** button or dial the CO Group Access Code.
2. Press the **[MSG/CALLBK]** button, a confirmation tone is received.
3. Hang-up; the **[MSG/CALLBK]** LED will flash.

To cancel the queue from the queued station:

1. Press the **[MSG/CALLBK]** button, the **[MSG/CALLBK]** LED extinguishes.

SLT

To request to be placed in queue while receiving the "All Lines Busy" signal:

1. Press the hook-switch.
2. Enter the {Call Back Feature code}.

To cancel the queue from the queued station:

1. Lift the handset.
2. Enter the {Call Back Cancel Feature code}.

System

When a CO/IP line becomes available:

1. A distinctive Queue Recall is sent to the station with the oldest queued call, the appropriate {CO/IP} line button LED will flash; the CO/IP line and station will appear busy to all other users.

Conditions

1. A CO/IP line can have any number of simultaneous queue requests.
2. A Station may only have a single active CO/IP queue request; activating a new queue request will replace (cancel), an existing queue.
3. A Queue recall will always notify the station with a tone ring, ignoring the station's assigned Intercom Signaling mode.
4. Queue recall will signal a station for 15 seconds, if unanswered, the station is removed from the queue.
5. If a station requests CO Queuing on a busy CO line, the requesting station checks the busy CO line's status every 5 seconds and receives CO Queue Recall Ring when the status check timer expires after a busy CO line returns to idle. Therefore, Queue Recall ring may be delayed after a busy CO line returns to idle. In addition, when several

stations request CO Queuing to a busy CO line, the Queue Recall Ring may not be provided sequentially.

Programming

Station Data 1) CO Queuing Access (PGM 133-Index1)
Numbering Data 1) CO Group Access Code (PGM 114)

Related Features

CO/IP Access

Hardware

3.21 CONFERENCE

Description

Conference supports communication between multiple parties (up to 13 per conference). The following table lists conference capacities for the iPECS-MG 100/300 systems.

NOTE: There are 133 extra TDM channels for the Conference feature.

Conference Type	Total number of conferences
3-way conference	No limit
4-way conference	33
5-way conference	19
6-way conference	13
7-way conference	12
8-way conference	9
9-way conference	5
10-way conference	4
11-way conference	4
12-way conference	3
13-way conference	3

3.21.1 Conference Room

Description

In addition to ad-hoc conferencing, users may establish a Conference Room. Both internal and external parties can be invited to a conference room and can join a conference room without further action by the user that established the Conference Room. A user can transfer an active call to a Conference Room. A Conference Room can be password protected restricting parties allowed to enter.

Up to 9 Conference Rooms can be set-up and each can support up to maximum 13-party.

Operation

Digital Phone

To set-up a Conference Room:

1. Enter the {Create-Conference-Room Feature} Code.
2. Dial the desired Conference Room number (571-579).
3. If desired, enter a password for the Conference Room (Max. 6 digits).
4. Press **[SAVE]** to establish the Room.

To join a Conference Room:

1. Dial the Conference Room Number.
2. Dial the Conference Room password and '*' for end mark if password is less than 6-digits (if the password is 6-digits, dialing '*' is not needed).

To delete a Conference Room:

1. Enter the {Delete-Conference-Room Feature} Code.
2. Dial the Conference Room number (571-579).
3. Dial the Conference Room password and '*' for end mark if password is less than 6-digits (if the password is 6-digits, dialing '*' is not needed).
4. Press **[SAVE]** to delete the Conference Room.

To transfer a call to a Conference Room:

1. Press the **[TRANS]** button.
2. Dial the Conference Room Number.
3. Dial the Conference Room password and '*' for end mark if password is less than 6-digits (if password is 6-digits, dialing '*' is not needed).
4. Hang-up to complete the transfer.

SLT

To set-up a Conference Room:

1. Lift the handset.
2. Dial {Conference Room Create Code}.
3. Dial the desired Conference Room number (1-9).
4. Dial the Conference Room password.
5. Press the hook-switch.

To join a Conference Room:

1. Lift the handset.
2. Dial the Conference Room Number.
3. Dial the Conference Room password and '*' for end mark if the password is less than 6-digits (if password is 6-digits, dialing '*' is not needed).

To delete a Conference Room:

1. Lift the handset.
2. Dial {Conference Room Delete Code}.
3. Dial the Conference Room number (1-9).

4. Dial the Conference Room password and '*' for end mark if the password is less than 6-digits (if password is 6-digits, dialing '*' is not needed).
5. Press the hook-switch.

Conditions

1. Once established, a Conference Room will remain opened until the Room is deleted.
2. Phontage and UCS Client may also create, delete and join a Conference Room; for operation instructions, refer to the **Phontage** or **UCS Client User Guide**.

Programming

Station Data	1) Conference Access (PGM 133-Index2)
Numbering Data	1) Conference Room Create/Delete Code (PGM 113)

Related Features

Automatic Speaker Select
Hold Recall
Unsupervised Conference

Hardware

3.21.2 Multi-Party Voice Conference

Description

The system allows multiple internal and external parties to be connected on a call, in conference mode.

Operation

Digital Phone

To establish an ad-hoc conference:

1. Establish the first call.
2. Press the **[CONF]** button; the LED will light, the connected party is placed on exclusive hold and the user receives a dial tone.
3. Place the second call.
4. When connected, press **[CONF]**; the new call is placed on exclusive hold.
5. Repeat steps 3 and 4 above to add additional conference parties (up to 13 total per conference).
6. Press the **[CONF]** button to establish conference.

To place a conference on hold:

1. Press the **[HOLD]** button, the **[CONF]** button LED will flash.

To retrieve a held conference call:

1. Press the **[CONF]** button, all parties will be reconnected.

SLT

To establish an ad-hoc a conference:

1. Establish the first call.
2. Press the hook-switch, the connected party is placed on exclusive hold and the user receives a dial tone.
3. Dial the {Conference Member Add Code}.
4. Place the second call.
5. When connected, repeat steps 2-4 above to add additional parties.
6. When completed adding parties, quickly press the hook-switch twice (within 2 seconds), all parties will be connected.

Conditions

1. The [CONF] button will remain illuminated at the initiators phone for the duration of the conference.
2. If the system receives a disconnect signal and no internal parties remain in the conference, the conference will be terminated and all remaining parties will be disconnected.
3. The normal Hold Recall process is applied to a conference on hold using the Unsupervised Conference Recall timer for recall timing.
4. If while setting up a conference, a system error tone is received, the initiator must press the [CONF] button (SLT must hook-flash) to regain the Intercom dial tone.
5. A station that is busy, in DND or other non-idle state cannot be added to a conference.

Programming

- | | |
|---------------------|---------------------------------------|
| Station Data | 1. Conference Access (PGM 133-Index2) |
|---------------------|---------------------------------------|

Related Features

Automatic Speaker Select
Hold Recall
Unsupervised Conference
Broker Call
Conference Room

Hardware

3.21.3 Unsupervised Conference

Description

A Digital phone user may establish a conference with external parties and exit the conference while allowing the external parties to converse privately without supervision from the user.

The system will disconnect the Unsupervised conference if disconnect is detected with only two parties connected or at expiration of the Unsupervised Conference timer. A Disconnect Warning tone is provided fifteen seconds prior to expiration of the timer.

If enabled, either party in an Unsupervised Conference can request the Unsupervised Conference timer be extended. The party enters the Timer Extension feature code and a digit 1 to 9 indicating the Timer extension multiplier; the system will then extend the timer based on the dialed digit multiple of the Timer. For example, if the Unsupervised Conference timer is 5 minutes and the user dials the digit 4, the timer will extend to 20 minutes (4 multiplied by 5 minutes).

Operation

Digital Phone

To set up an Unsupervised Conference:

1. Establish a conference by the normal procedure.
2. Goes On-Hook then remained External users can communicate without Supervisor..

To set up a Supervised Conference:

1. Establish a conference by the normal procedure.
2. Press the **[CONF]** button; the button LED will flash to indicate the Supervised Conference activation (once set up, the conference supervisor must re-enter the conference).

To reenter the Supervised Conference:

1. Press the flashing **[CONF]** button.

Conf Party

To extend the Unsupervised Conference from a connected party:

1. Dial the Timer extension multiplier (1-9).

Conditions

1. The Unsupervised Conference Timer will also apply to an external call placed by a DISA user.
2. An Unsupervised conference will be terminated if the system receives a disconnect signal or the Unsupervised Conference Timer expires.
3. An Unsupervised Conference will not recall at the user Station.

Programming

Station Data	1) Conference Access (PGM 133-Index2)
CO Line Data	1) Unsupervised Conference Timer (PGM 166-Index9/PGM 171-Index6) 2) Unsupervised Conference Extend (PGM 171-Index2)
Numbering Data	1) Unsupervised Conference Extend Code (PGM 113)

Related Features

Multi-Party Voice Conference

Hardware

Digital Phone to establish Unsupervised Conference

3.22 CUSTOMER SITE NAME

Description

A Customer Name, up to 24 characters, may be entered into the system database. The name will be displayed on the SMDR and database outputs as well as during Admin. sessions.

Operation

System

When a name is assigned, operation of this feature is automatic.

Conditions

Programming

System Info. 1) Site Name (PGM 100)

Related Features

Hardware

3.23 DATA LINE SECURITY

Description

Data transmitted over CO lines is subject to distortion and errors if system tones such as Camp-On, Call Wait and Override are applied during transmission. To eliminate such errors, stations that use analog data (modems or Fax) can be assigned to block incoming system tones.

Operation

System

When Data Line Security is assigned, System tones are automatically blocked.

Conditions

1. Stations or an Attendant attempting to Camp-On or Override a station with Data Line Security will receive an error tone.
2. When Data Line Security is enabled, the system will not apply audio gain to the call.

Programming

Station Data 1) Data Line Security (PGM 123–Index6)

Related Features

3.24 DELAYED CO/IP RING

Description

Determines if the CO/IP Ringing will be sent immediately on receipt, or delayed. The delay can be up to 9 system ring cycles, allowing other stations to answer the call.

Operation

System

When assigned, Delay Ring operation is automatic.

Conditions

1. Delay Ring can be assigned for a station.
2. The station will receive immediate ringing, if no delay is entered when programming Ring assignments.
3. If stations are assigned for immediate ring, the call will be routed using the incoming CO Alternate programming.
4. The delay is applied only when ring service type is 'ALL-RING'.

Programming

- | | |
|--------------|---|
| CO Line Data | 1. CO Station Ring Assignment (PGM 167) |
|--------------|---|

Related Features

Hardware

3.25 DELAYED AUTO ATTENDANT

Description

An incoming CO/IP call can be routed to the VMIB Auto Attendant either immediately upon detection or after a delay time (up to 90 seconds). This allows other stations assigned for immediate ringing the opportunity to be answered before the call is routed to the Auto Attendant.

Operation

System

When assigned, operation of this feature is automatic.

Conditions

1. CO ring Assignment must be set for service type 'All_Ring', and the Feature Code Delay Counter (PGM 167) must be enabled including the desired VM announcement.
2. When Delayed Auto Attendant Ring is assigned, following expiration of the delay, the call will no longer ring at the assigned stations and will be routed to the VMIB Auto Attendant.
3. If no delay is entered, the call will ring to the VMIB Auto Attendant immediately.

4. To assign a Delayed Attendant ring, at least one station must be assigned for immediate ringing.

Programming

CO Line Data	1. CO Ring Assignment (PGM 167)
Table Data	1. Announcement Table (PGM 259)

Related Features

Hardware

3.26 DIAGNOSTIC/MAINTENANCE

Description

The System software incorporates various diagnostic and maintenance routines that may be “called” remotely or locally through the System RS-232 Serial ports, a TCP/IP connection using a Web browser established over IP networks or a PPP connection over ISDN. Routines that can be accessed include trace functions at the device level, commands for diagnostics and maintenance, and tools for manipulation at the OS level.

An optional Network Management System (NMS) application is available providing remote access to the system for maintenance and diagnostics (refer to the *iPECS NMS Manual*).

Operation

Conditions

Programming

Related Features

Hardware

3.27 DIAL-BY-NAME

Description

A name, up to 16 characters, may be assigned to each Station and System Speed dial. In addition, each station may be assigned a 16-character name. When assigned, a user may place an intercom call to another station or select a Station or System Speed dial using the name.

The user selects from one of three Dial-by-Name directories and enters characters employing 2 dial pad buttons for each character. The system finds and displays the nearest match to the user entries. The user may continue entering characters or scroll the directory at any point using the

[VOL UP]/[VOL DWN] button and select a name to call. The number associated with a selected name is displayed by using the [TRANS] button.

Operation

Digital Phone

To use Dial by Name on a two-line phone:

1. Press the [SPEED] button twice.
2. Dial the desired directory,
 - 1 – Station Speed
 - 2 – System Speed
 - 3 – Station Name
3. Search the directory using the [VOL UP]/[VOL DOWN] button or by entering characters.
4. Press the [SAVE] button to place the call.

To use Dial by Name on a three-line phone:

1. Press the {DIR} Soft button.
2. Dial the desired directory,
 - 1 – Station Speed
 - 2 – System Speed
 - 3 – Station Name.
3. Search the directory using the [VOL UP]/[VOL DOWN] button or by entering characters.
4. Press the [SAVE] button to place the call.

To program the station user name:

1. Press the [PGM] button.
2. Dial 12 {User Name Program code}.
3. Enter the name (up to 16 characters); refer to Station Speed Dial for character entry.
4. Press [SAVE].

SLT

To program the station user name:

1. Lift the handset
2. Dial {Name Register Code}.
3. Enter the name (up to 16 characters); refer to Station Speed Dial for character entry.
4. Press the hook-switch, confirmation tone is received.

Conditions

1. Available characters are A to Z, space, and period; refer to Station Speed Dial for character entry.
2. The LCD will display multiple names (one per LCD line up to 16 characters).
3. If a user selects a directory with no entries or there is no match to the user entry, an "Empty List" message is displayed and the error tone is provided.
4. Dial-by-Name is only available to Digital Phones with a display; other users will receive an error tone if an attempt is made to access Dial-by-Name.
5. A user may both scroll and enter characters to search a directory.

Programming

- Station Data** 1. Speed Access (PGM 134-Index1)

Related Features

Station Speed Dial

Hardware

Digital Phone w/Display

3.28 DIAL PULSE TO TONE SWITCHOVER

Description

On a pulse dial CO line, the user can request the system to change the signaling mode from pulse to DTMF, allowing the user to access outside facilities that require DTMF signals such as banking services, voice mail, etc.

Operation

To switch from pulse to DTMF while on a pulse CO line:

1. Dial '*' (signaling change to DTMF).

Conditions

1. In a Speed Dial, the '*' will automatically insert a pause before dialing the remaining digits.
2. This command is only recognized for analog pulse dial CO lines.
3. Dial pulse to tone switchover is not available in the Redial features.
4. For VoIP calls, pulse dialing is not available; switchover is not required or supported.

Programming

- CO Line Data** 1. CO Dialing Type (PGM 160-Index10)

Related Features

Speed Dial

Hardware

3.29 DIALING RESTRICTIONS

3.29.1 Class of Service

Description

Dialing privileges can be assigned for each DN at a Station and CO line (up to 16 privileges). The Class of Service (COS) feature is applied in the following cases:

- 1) When an internal station dials out through a CO line,
- 2) When an external caller tries to make another external call using DISA or DID.

The dialing privileges are the result of the interaction of the Station and CO Class of Service (COS) assignments as shown in the following tables. Users placing an outgoing call or dialing after answering a call will be allowed the dialing privileges assigned.

Station/CO COS	Dialing Restriction
0	Intercom and Emergency number calls are allowed; incoming and transferred calls are allowed.
1	No restrictions are placed on dialing.
2 - 15	Assignments in each toll table are monitored for Allow and Deny numbers.

- Toll Tables – Each Toll Table permits entry of 100 Allow codes and 100 Deny codes. Each code can contain up to 16 digits including digits 0-9.
- Toll Table process – As digits are dialed, they are compared to entries in the appropriate Toll Table. Based on the Allow and Deny entries, the system applies the following rules to allow or deny the call.

Rule 1 – If a table has no entries, no restrictions are applied.

Rule 2 – If there are only Deny entries, restrictions are provided as Deny only.

Rule 3 – If there are only Allow entries, restrictions are provided as Allow only.

Rule 4 – If there are both Allow and Deny entries, the Deny entries are searched. If the dialed number matches a Deny entry, the call is restricted; if no match is found the call is allowed.

Operation

System

The assigned COS is applied automatically by the system.

Conditions

1. There are 16 different COS; Stations and CO Lines can each have a different COS according to the Station grade and CO Line type.
2. Toll Exception can be programmed differently according to the Day/Night/Timed Mode.
3. If COS is set to 0, the DN can make intercom calls only; CO line access is disabled.
4. If COS is set to 1, the DN can make any call (no exception).

Programming

Station

1. Station COS (PGM 137)

CO/IP

1. CO COS (PGM 177)
2. Incoming CO Group Access, Outgoing CO Group (IC Call Transfer Directly (PGM 179-Index5)

CO Line Data

1. Toll Exception Table (PGM 250)

Related Features

Temporary Station COS/LockWalking COS

Hardware

3.29.2 Day/Timed & Night Station COS

Description

CO Line Toll Exception can be applied differently in Day/Night/Timed mode at each Station. The service mode is generally controlled by the Attendant group member, and based on the mode, appropriate dialing privileges are established.

Operation

System

Dialing restrictions are automatically applied based on COS assignments:

Condition

1. If COS is set to 0, only intercom calls can be placed; CO line access is disabled.
2. If COS is set to 1, there are no restrictions.

Programming

Station Data

1) Station COS (PGM 137)

Table Data

- 1) System Time Table (PGM 253)
- 2) Weekly time table (PGM 254)
- 3) Holiday time table (PGM 256)
- 4) Toll Exception table (PGM 250)

Related Features

Authorization Codes (Password)
Class of Service
Temporary Station COS/LockWalking COS
Auto Service Mode ControlDay/Night/Timed Ring Mode

Hardware

3.29.3 Temporary Station COS/Lock

Description

A User or an Attendant can temporarily change the Station COS preventing unauthorized toll dialing from the station (ex., "locking the station"). When locked, the station will still be allowed to place internal calls and Emergency number calls.

Operation

Digital Phone

To activate Temporary COS:

1. Press the **[PGM]** button.
2. Dial 31, {Temporary COS code}.
3. Dial the applicable Authorization Code.
4. Press the **[SAVE]** button.

To restore the assigned COS:

1. Press the **[PGM]** button.
2. Dial 32 {Restore COS code}.
3. Dial the applicable Authorization Code.
4. Press the **[SAVE]** button.

SLT

To activate Temporary COS:

1. Press the hook-switch.
2. Dial the {SLT Programming code}.
3. Dial 31, {Temp COS code}.
4. Dial the applicable Authorization Code.
5. Press the hook-switch.

To restore the assigned COS:

1. Press the hook-switch.
2. Dial the {SLT Programming code}.
3. Dial 32, {Restore COS code}.
4. Dial Authorization Code.
5. Press the hook-switch.

System Attendant

To activate Temporary COS:

1. Press the **[PGM]** button.
2. Dial 031, {Temp COS code}.
3. Enter the Station range.
4. Press the **[SAVE]** button.

To restore the assigned COS:

1. Press the **[PGM]** button.
2. Dial 032, {Restore COS code}.
3. Enter the Station range.
4. Press the **[SAVE]** button.

Conditions

1. The Station is restored to the Station COS as appropriate for the active service mode (Day, Night, or Timed).

Programming

- | | |
|----------------|-----------------------------------|
| Station | 1. Station COS (PGM 137) |
| CO/IP | 1. CO COS (PGM 177) |
| System | 1. Toll Exception Table (PGM 250) |

Related Features

Hardware

3.29.4 Walking COS

Description

A User may temporarily override Toll Restriction at a Station to make Toll Calls from a normally Toll Restricted station. An Authorization Code is required in order to activate Walking COS.

Operation

Digital Phone

To activate Walking COS:

1. Press the **[PGM]** button.
2. Dial 33, {Walking COS code}.
3. Enter the Station number.
4. Dial the Station Authorization code (password).
5. Dial '*' (end mark).
6. Place a call in the normal manner.

SLT

To activate Walking COS:

1. Dial the {SLT Programming code}.
2. Dial '33', the Walking COS code.
3. Enter the Station number.
4. Dial the Station Authorization code (password).
5. Dial '*' (end mark).
6. Place call as normal.

Conditions

1. The Station COS applied for Walking COS is the COS of the station.
2. Walking COS applies the temporary COS for only one call; terminating the call returns the station to the assigned Station COS. The user may reactivate Walking COS to place another call or press the Flash key (instead of hook-switch) at the end of previous call to maintain Walking COS.

Programming

Station	1. Station COS (PGM 137)
CO/IP	1. CO COS (PGM 177)
System	1. Toll Exception Table (PGM 250)

Related Features

Class of Service
Auto Service Mode Control
Day/Night/Timed Ring Mode
Authorization Codes (Password)

Hardware

3.30 DIFFERENTIAL RING

Description

Differential Ring provides one of 4 different audible Ring signals to be assigned to a Digital Phone, allowing users to determine which phone is ringing and the type of call (Intercom or CO/IP). When the phone receives an incoming call, the designated ring signal is provided over the speaker. Different selections are assigned for Intercom and CO/IP calls.

Operation

Digital Phone

To select the desired ring tone:

1. Press the [PGM] button.
2. Dial 2 (Ring Selection).
3. Dial 1 (Intercom) or 2 (CO/IP) ring.
4. Dial 1 (Ring Source).
5. Dial Ring Tone selection –
1-4: Digit Phone
1-8: IP Phone
6. Ring Tone is presented.
7. Press the [SAVE] button.

Conditions

Programming

- | | |
|---------|---|
| Station | 1. Differential Ring (PGM 124-Index3-4) |
|---------|---|

Related Features

Hardware

3.31 DIGIT CONVERSION

Description

When a User dials digits, the dialed digits are converted according to the Digit Conversion Table before the Numbering Plan is checked. Digit conversion is performed on outside incoming CO calls, in addition to converting User dialed digits.

- Time Zone for Digit Conversion – *The pressed digits can be converted into a different digit stream according to the time zone: Always, Day/Night/Timed zone, and LCR Day/Time zone. There can be up to 9 conversion matrices in the LCR Day/Time zone, while 3 conversion rules are possible in the Day/Night/Timed zone. Digit conversion is performed only when there is a conversion rule that applies based on the specific time the digits are pressed.*
- Dummy CO Dial Tone – *The CO line is seized following digit conversion. Therefore, it is impossible to receive a CO dial tone when digit conversion is programmed. To remove any*

inconveniences of users, the system can be configured to provide a dummy CO dial tone after one of the dialed digits is pressed.

- Digit information Display Before or After Conversion - *Each Station can be programmed to display either the dialed digits or the digits after conversion. SMDR also can print either dialed digits or the digits after conversion.*

Operation

Digit Conversion is applied automatically according to ADM programming.

Conditions

1. Each DN and CO line can be set to have one of 9 digit conversion Tables.
2. There are 300 entries for each digit conversion table.
3. The Digit Conversion Table allows up to 16 digits to be programmed as 'dialed digit' and 'converted digit'.
4. The following features have higher priority over digit conversion;
 - AND (Automatic Network Dialing)
 - Automatic CO seize
5. In Call log, dialed digit is displayed on the station's LCD.

Programming

Station Data	1. Digit Conversion Table Index (PGM 131-Index3)
CO Data	1. Digit Conversion Table Index (PGM 160-Index6)
Table Data	1. Digit Conversion Table (PGM 251, PGM 252)

Related features

Hardware

3.32 DO NOT DISTURB (DND)

Description

A Station enabled with the Do Not Disturb (DND) feature, can be placed in DND to block incoming ringing for CO/IP and Intercom calls, transfers, and paging announcements.

Operation

Digital Phone

To activate DND for a P-DN (Prime Directory Number):

1. Press the [DND] button; the [DND] button LED illuminates.

To remove DND for a P-DN (Prime Directory Number):

1. Press the [DND] button; the [DND] button LED extinguishes.

To activate DND for a S-DN (Sub Directory Number):

1. Press {S-DN} button.
2. Dial the {DND Feature Code}; the {S-DN} button LED illuminates.

To remove DND for a S-DN (Sub Directory Number):

1. Press {S-Line} button.
2. Dial the {DND Feature Code}; the {S-DN} button LED extinguishes.

SLT

To activate DND:

1. Dial the {DND feature code}; a confirmation tone is received.

To remove DND:

1. Dial the {DND feature code}; a confirmation tone is received.

Conditions

1. A station will receive an error tone if unable access DND.
2. If DND is enabled, pressing the **[DND]** button while ringing will activate One-Time DND.
3. An Attendant may cancel DND for Stations on the System.
4. DND service is available for use by Attendants.
5. Recalls for CO/IP calls will override the DND feature.
6. A station in DND is out-of-service for all incoming calls including Station Group calls.
7. A station in DND ignores calls forwarded to the station; if the last station in a Call Forward chain is in DND, the call will ring at the previous Station in the chain.
8. When calling a Station in DND, the Digital Phone display will indicate the DND status.

Programming

Station Data	1. DND Access (PGM 132-Index4)
System Data	1. LED Color/Flash (PGM 234)

Related Features

Hardware

3.33 DOOR OPEN

Description

The hardware is equipped with a relay that activates an External Control Contact. The contact can be assigned to one of several functions including a Door Open Contact; the contact is connected to a door-lock release mechanism. When a Station receives the Door Bell signal, the Station User may dial the Door Open code to activate the contact.

Operation

Digital Phone

To assign a {DOOR OPEN} button:

[PGM] + {FLEX} + Button Feature Type (1) + {Door Open Feature Code} + [SAVE]

To activate the relay contact:

1. Lift handset or press **[SPEAKER]** button.

2. Dial the {Door Open code}.
3. Hang-up to return the phone to idle.

OR

1. Lift the handset or press [**SPEAKER**].
2. Press the programmed {**DOOR OPEN**} button.
3. Hang-up to return the phone to idle.

Conditions

1. One relay contact is available.
2. The contacts are rated at 1 amp, 24 VDC.

Programming

- | | |
|-----------------------|--|
| Station Data | 1. Door Open Access (PGM 121-Index13) |
| System | 1. External Contact Control (PGM 228)
2. Door Open Timer (PGM 222-Index1) |
| Numbering Data | 1. Door Open Code (PGM 113) |

Related Features

Loud Bell Control (LBC)

Hardware

3.34 DOOR PHONE

Description

An intercom box can be connected to the System and located in a convenient place at your facility for receiving page announcements and intercom calls. Additionally, the intercom box can signal assigned Stations using the Auto Dial feature in the System.

Operation

To call an intercom box, perform the following Steps:

1. Lift the handset or press the [**MON**] button.
2. Dial the Station number of intercom box, or press the programmed flexible button for the Intercom box.
3. After answering the call from the Intercom box, announce the call.

To place a call from an intercom box:

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

Conditions

1. An Intercom box can be a member of the Page Zone group.
2. To receive Intercom box calls, set the Auto Dial Digit and Pause timers (PGM 138).

3. An Intercom box can be answered automatically using Hands free mode when in ICM Answer Mode (PGM124)
4. The new Intercom box (LDP-DPB) has the **[CALL]** button.

Programming

System

1. Station Auto Dial Attribute (PGM 138-Index1-2)
2. ICM Answer Mode (PGM 123-Index5)

Related Features

Door Open

Hardware

3.35 EMERGENCY CALL

Description

Regardless of Station dialing restrictions (COS), the user may dial assigned Emergency numbers as needed.

Operation

System

The system will automatically override any toll restrictions and process an assigned Emergency number call.

To dial an Emergency Call at the station:

1. Lift handset or press **[SPEAKER]** button.
2. Dial the {Emergency Code} (ex., 911) without accessing CO Line.

Conditions

1. The CO Line Group Access Code and digits to be dialed should be assigned to the emergency changed digit.
2. If the dialed number for the Emergency code is the same as the Numbering code including station in the system, the Emergency code has the preference.

Programming

Table Data

1. Emergency Code Table (PGM 258)

Related Features

Hardware

3.36 EXECUTIVE/SECRETARY BY DN(DIRECTORY NUMBER)

Description

Executive/Secretary feature can be achieved by utilizing the DN (Directory Number) feature of the system (refer to DN).

For example, when a DN is set as MADN, calls for an Executive can be routed to that DN. The DN can be programmed on a flex button at the Executive station with a “no ring” option. At the Secretary station, the DN is stored on a flex button with an “immediate ring” option. In this case, the calls for the executive will ring the Secretary’s station immediately. When the secretary answers the call, the call can be put on Hold. The secretary will be able to inform the executive that there’s a held call on a button (requiring an additional button programmed for hands-free access to the Executive DN). When the secretary presses the button, it would be possible to tell the executive to answer the held call.

The Executive’s DN button may have a “delayed ring” option instead of “no ring” option, the executive will hear the ring signal after programmed delay. Since two stations (executive and secretary) ring after delay time, either Executive or Secretary can receive the call. This setting helps when the Secretary is not at their desk temporarily.

Operation

To program Executive/Secretary Forward:

1. Assign a DN as MADN to be used as Executive’s Secretary for calls.
2. Register a DN flex button at the Secretary Station and Executive Station.
3. Set the DN flex button Ring option as ‘No Ring’ or ‘Delayed Ring’ at the Executive Station.
4. Enable the ‘Forced Hands free Access’ option at the Executive and Secretary Stations.
5. Assign the Executive station flex button as telephone number {Forced Hands free code} + {Secretary Station number} for use when Executive intercom calls the Secretary.
6. Assign Secretary station flex button as Telephone number {Forced Hands free code} + {Executive station number} to use when Secretary Intercom calls the Executive.

Conditions

1. An Executive may have multiple Secretaries and a Secretary may have multiple Executives; each is considered a separate Executive/Secretary pair.
2. If the Secretary is busy when a call is received for the Executive, the caller will receive a busy tone.
3. If an Executive has multiple Secretaries, calls will automatically route to the Executive’s first idle Secretary.
4. The Executive may use Call Forward to send calls to stations other than the Secretary.

Programming

- Numbering Plan** 1. Forced Handsfree code, Feature Numbering Plan (PGM 113)

- Station**
1. Station Number Type (PGM 130–Index1)
 2. Flex Button Assign (PGM 126)
 3. DN Button Ring Option (PGM 126-Index2)
 4. Forced Hands free Access (PGM 132-Index1)

Related Features

3.37 EXECUTIVE/SECRETARY BY EXEC/SEC ASSIGNMENT

Description

Phones can be assigned as Executive/Secretary groups. By activating DND, the Executive also activates Unconditional Call Forward to the Secretary, which will forward Executive calls to the Secretary. With the “CO Call to Secretary” option enabled, all CO calls to the Executive forward to the Secretary regardless of the Executive’s station status. In addition, if the Secretary is in DND (or, all Secretaries are DND), Executive calls sent to the Secretary route back to the Executive if the “Call Exec If First Sec in DND” or “Call Exec if All Sec in DND” option is enabled.

Each Executive access privilege can be assigned. If executive access is enabled, the call is routed to the executive directly.

If the “ICM Call to Secretary” option is enabled, all internal calls to the Executive (except for calls from the executive having executive access privilege) forward to the Secretary regardless of the Executive’s station status.

Callers to an Executive can leave a Message Wait indication. The message waiting indication is given to the Executive or the first Secretary station assigned as the message wait destination.

Operation

iPECS Phone

To activate/deactivate Executive/Secretary forward from the Executive Phone

1. Press the [DND] button to toggle Executive/Secretary Forward.

Conditions

1. An Executive can have up to 3 Secretaries.
2. If the Secretary is busy when a call is received for the Executive, the caller will receive busy tone.
3. If an Executive has multiple Secretaries, a Secretary can be selected by ‘Secretary Choice’ option. There are three options, 1) First Idle 2) Longest Idle.
4. An Executive cannot be a Secretary to another Executive. And a Secretary cannot be an Executive to another Secretary.
5. The Executive may use Call Forward to send calls to stations other than the Secretary.
6. Message wait station can be the Executive or the first Secretary
7. A Secretary can call his executive.
8. If a Secretary (Station B) assigns unconditional call forward to another station (Station C), the forward destination station can make a call to the Executive (Station A).
9. When calls are forwarded to subsequent stations (ex., Station C assigns call forward to Station D, E, or F), Executive Call Service is not supported.

Programming

- | | | |
|----------------------|----|---------------------------------------|
| Station | 1. | DND Access (PGM 132-Index4) |
| Station Group | 1. | Executive/Secretary Assign (PGM 241) |
| | 2. | Executive/ Executive Access (PGM 242) |

Related Features

Hardware

Digital Phone

3.38 EXTERNAL AUTO ATTENDANT/VOICE MAIL

3.38.1 AA/VM Group

Description

The system provides support for an adjunct Auto Attendant/Voice Mail system via connection to SLT ports. When a call arrives for the External AA/VM Group, the system will search the group for an idle port and deliver the call.

Signaling information between the system and AA/VM system may be assigned for in-band DTMF signaling or the Simplified Message Desk Interface (SMDI) signaling protocol over the assigned system RS-232 port.

Operation

System

The system will interface with the External AA/VM based on database assignments.

Conditions

1. Selection of SMDI or in-band signaling can be selected in Admin. Programming (refer to **Admin Programming Manual**).
2. Only one AA/VM Group can be defined in the system; multiple definitions may cause erroneous system operation.
3. SLT connected to SLIM cannot be assigned as member of VM Group.

Programming

- | | | |
|-----------------------|----|------------------------------------|
| Station Groups | 1. | VM Group Assignment (PGM 200) |
| | 2. | VM Group Attribute (PGM 203) |
| | 3. | Voice Mail Dialing Table (PGM 269) |

Related Features

In-band (DTMF) Signaling
SMDI (Simplified Msg Desk Interface)

Hardware

External AA/VM system

3.38.2 In-band (DTMF) Signaling

Description

The system may employ in-band signaling to communicate with an External AA/VM system. When a call is routed to the AA/VM SLT port, the system will send DTMF signals informing the AA/VM of the characteristics of the call. DTMF digit strings are assigned to various functions allowing the AA/VM to respond appropriately to the call. These definitions are entered in the "Voice Mail Dialing Table."

Operation

System

The system will interface with the External AA/VM based on database assignments:

Conditions

1. Selection of SMDI or in-band signaling can be selected in Admin. Programming.
2. Only one AA/VM Group can be defined in the system; multiple definitions may cause erroneous system operation.

Programming

Station Groups	1. VM Group Assignment (PGM 200) 2. VM Group Attribute Assignment (PGM 203)
Table	1. Voice Mail Dialing Table (PGM 269)
System	1. Voice Mail Interface Select (PGM 223-Index3)

Related Features

AA/VM Group
SMDI (Simplified Msg Desk Interface)

Hardware

External AA/VM system

3.38.3 SMDI (Simplified Msg Desk Interface)

Description

The system may employ SMDI protocol to communicate with an adjunct AA/VM system. When a call is routed to an AA/VM SLT port, the system will send SMDI messages over the assigned SMDI RS-232 port, informing the AA/VM of the characteristics of the call.

- SMDI Protocol – There are three types of SMDI messages (listed below). Within each message is an "Action Code", which defines the function or required action of the AA/VM system. Fields within the messages also define the called/calling station and station status. The various message types and definition of the fields are shown in the chart below.
 - Type I message: cr lf **MD** ggg mmmm a xxxxxxxx sp yyyyyyyy sp cr lf^Y
 - Type II message: cr lf **MD** ggg mmmm a xxxxxxxx sp sp cr lf^Y
 - Type III message: cr lf **MD** ggg mmmm a sp yyyyyyyy sp cr lf^Y

Table 3.38.3-1 Message Field Definitions

Field	Description	Values
cr	Carriage Return	
lf	Line Feed	
MD	Message Desk	
ggg	Message Desk Number, AA/VN system	Default=001
Mmm	Message Desk terminal	Range=0001-9999 VM port
A	Action Code	
xxx...x	Called Station Number or Station Calling the VM Group	
yy...y	Calling Station Number	
Sp	ASCII Space Character	
^Y	End of SMDI Message	Control + Y (0x19)

The following table provides detailed information on the meaning and function of the various SMDI messages used.

Table 3.38.3-2 SMDI Messages

Action Code	Reason	Purpose	In-band code	Message Type	SMDI Message MD 001 0001-
A	Unconditional forward to VM	Put Mail	P#	II	A xxxxx yyyy
B	Called Station busy	Busy Mail	P#3P	II	B xxxxx yyyy
C	Disconnect, connected party	disconnect	*****	II	C xxxxx yyyy
D	Direct Fwd to VM group	Get Mail	P##	II	D xxxxx yyyy
E	Error, invalid number	Error	P#*5P	II	E xxxxx yyyy
H	Two-way Record	record	None	II	H xxxxx yyyy
I	DND	DND	P#*6P	II	I xxxxx yyyy
N	No Answer	No Answer	P#*4P	II	N xxxxx yyyy
R	Direct CO/IP ring to VM group	AA	None	III	R xxxxx yyyy

Operation System

The system will interface with the External AA/VM based on database assignments:

Conditions

1. Selection of SMDI or in-band signaling can be modified using Admin. Programming (refer to **Admin. Programming Manual**).
2. Only one AA/VM Group can be defined in the system; multiple definitions may cause erroneous system operation.
3. The calling number will be displayed with SMDI CLI INFO attribute (PGM203-Index7)

Programming

- | | |
|-----------------------|--|
| Station Groups | 1. VM Group Assignment (PGM 200)
2. VM Group Attribute Assignment (PGM 203) |
| System | 1. Voice Mail Interface Select (PGM 223-Index3) |

Related Features

- AA/VM GroupIn-band (DTMF) Signaling
- VMIB Integrated Auto Attd/Voice Mail

Hardware

- External AA/VM system

3.39 FLEXIBLE NUMBERING PLAN

Description

User access to System resources and features is accomplished using Feature codes or Flexible buttons. The Administrator can select from one of seven different standard Numbering Plans, and if desired, can assign codes for individual functions in the Flexible Numbering Plan. The feature codes are defined in the System's Flexible Numbering Plan (refer to [Appendix B](#) of the *iPECS-MG Admin. Programming Manual*).

Operation

System

The System implements Feature activation based on the selected Flexible Numbering Plan.

Conditions

1. The System can support up to 8-digit numbering for Station numbers or Feature codes.
2. To assign a Numbering Plan code, it should be matched (type) with a Prefix Numbering Plan consisting of a prefix and additional digits.
3. The selected Prefix Numbering Plan cannot conflict (ex., if a prefix consists of 1 digit and 4 additional digits, then there cannot be another prefix of 10 digits with 4 additional digits).
4. The additional digits of a Prefix Numbering Plan cannot be more than 4.
5. When a Prefix Numbering Plan consists of more than 4 digits, the preceding digits of the prefix code are placed at more than 4 digits from end digit (called Master Prefix Digits, can be up to 3 in the MG-100 system and 5 in MG-300 system).
4. When a conflicting Prefix is identified, the existing non-conflicting Numbering Plan is used until correctly updated.
5. If Numbering Plan type 7 is selected, all numbering codes are deleted; the first user should assign the Prefix Numbering plan. After configuring the prefix, the user can assign Station Number, CO Group Access Code, Extra Numbering and Feature code (for use when user wants to reconfigure all numbering codes).

Programming

- Numbering Plan**
1. Numbering Plan (PGM 110)
 2. Prefix Numbering Plan (PGM 111)
 3. Flexible Station Number (PGM 112)
 4. Feature Numbering Plan (PGM 113)
 5. CO Group Access Code (PGM 114)
 6. Station Group Number (PGM 115)

Related Features

Hardware

3.40 GREEN POWER SAVE

Description

The system can disable the power of a Digital Phone or SLT (Single Line Telephone) installed in the DTIB/SLIB/DSIU at night or during holiday mode. The power On/Off can be controlled by Web Admin manually or automatically according to the assigned power On/Off time.

Operation

System

Operation of this feature is either automatic (when programmed) or by Web Admin.

Conditions

1. SLT (Single Line Telephone) is supported on DSIU, while Digital Phone is not supported.
2. If phone power is disabled, calls cannot be placed and received.
3. In the event of system reset, power is enabled.

Programming

- System**
1. Green Power Save Time (Web Admin.).
 2. Green Power Save Enable (Web Admin.).

Related Features

Hardware

3.41 HEADSET COMPATIBILITY

Description

An industry standard headset can be connected to a Digital Phone in place of or in addition to the handset. The Station must be set for Headset operation.

In Headset mode, pressing the **[SPEAKER]** button will send audio to the Headset instead of the speakerphone. Additionally when in the Headset mode, ring signals can be delivered to the speaker or the headset as defined in the System database.

Operation

Digital Phone

To change operation from Speakerphone to Headset:

1. Press the **[PGM]** button.
2. Dial 61 {Headset select code}.
3. Dial the appropriate selection,
 - 0: Headset,
 - 1: Speakerphone,
 - 2: Ear-Microphone.
4. Press the **[SAVE]** button.

To change the device to receive ring signals:

1. Press the **[PGM]** button.
2. Dial 62 {Ring select code}.
3. Dial the appropriate selection,
 - 1: Speakerphone,
 - 2: Headset,
 - 3: Both.
4. Press the **[SAVE]** button.

To place/answer calls using the headset:

1. Press the **[SPEAKER]** with the phone in Headset mode.

Conditions

1. The Intercom Signaling Mode can be set in the Headset mode as with the Speakerphone mode.
2. Although the phone is in the Headset mode, the system will monitor hook-switch status; if the user lifts the handset to go off-hook, audio automatically is delivered to the handset.

Programming

Station

1. Speaker/Headset Ring Mode (PGM 121-Index3)
2. Speakerphone/Headset (PGM 121-Index2)

Related Features

Speakerphone

Paging

Hardware

3.42 HOLD

3.42.1 Hold

Description

The user can place a CO/IP line or Station on Hold during a phone conversation.

Operation

Digital Phone

To place an active CO Call on Hold:

1. Press the [HOLD] button.

To retrieve the Held CO Call:

1. Press the {CO} OR {LOOP-KEY} button associated with the held CO, and then the CO call is connected again.

SLT

To place an active CO Call on Hold:

1. Press the Hook-switch during a conversation.
2. Dial {System Hold Code}; a confirmation tone is heard.
3. Place the handset.

To retrieve the Held CO call:

1. Lift the handset; the Station is connected with the held party.

Conditions

1. After placing a call on Hold, the station returns to an idle state and the user can make another call.
2. If the Station is in the off-hook state when making a call on hold, the dial tone is heard.

Programming

- Numbering Plan** 1. Feature Numbering Plan (PGM 113)

3.42.2 Hold Recall

Description

When a user places a CO/IP call on hold, a hold timer is activated. If the timer expires, the held call will recall at the station for the duration of the I-Hold Recall timer. If the call remains unanswered at timer expiration, the Attendant will then receive recall for the Attendant Recall timer. If still unanswered after timer expiration, the CO/IP call is disconnected and the circuits in use are returned to idle.

Operation

Hold Recall operation is automatic.

Conditions

1. Separate Timers are assigned for the various types of hold: System, Transfer, etc.
3. Hold Timer can be assigned by Tone time in the Tone Table.
4. The route destination after a Hold recall ring is programmed in Incoming/Outgoing CO Alternate. The following destinations can be assigned:
 - 1) Disconnect
 - 2) Attendant
 - 3) CO Ring Assign
 - 4) Alternative Ring Table
 - 5) Tone
 - 6) Pilot Hunt
 - 7) Transfer Station (Transfer Call Only)

Programming

- | | |
|---------------------|---|
| CO Line Data | <ol style="list-style-type: none">1. Incoming CO Alternate (PGM 169)2. Outgoing CO Alternate (PGM 173) |
| Table Data | <ol style="list-style-type: none">1. CO Hold Tone Timer (Web Admin. PGM 290–Index55) |

Related Features

Call Transfer, CO/IP

Hardware

3.42.3 Automatic Hold

Description

While on an active CO/IP call, the system will place the call on hold automatically.

The station can be programmed to support CO/IP to CO/IP Automatic Hold. In this case, pressing a CO/IP button while on a CO/IP call will place the active call on hold and access the selected CO/IP line.

Operation

Digital

To use Automatic Hold:

1. Press **{CO}** OR **{LOOP-KEY}** button, while on an active Station or CO/IP call; the call is placed on Hold.

Conditions

1. CO/IP lines placed on hold with Automatic Hold are placed in the assigned Hold Timer.
2. Hold Timer can be assigned by Tone time in the Tone Table.
3. There is no limit on the number of calls that can be placed on hold using Automatic Hold.

Programming

- | | |
|---------------------|---|
| Station Data | <ol style="list-style-type: none">1. Automatic Hold Access (PGM 123-Index3) |
|---------------------|---|

Related Features

Hold Recall

Hardware

3.43 HOT DESK

Description

Digital Phones can be assigned as Hot Desk (Dummy Terminal) phones allowing Users (Agents) to login to the System. The Hot Desk will become active and will take on the attributes defined for the Agent's Station number. When the Agent logs off, the Hot Desk phone becomes inactive and the Agent's calls can be forwarded to the User-entered destination. A different Agent may then login using the inactive Hot Desk phone.

Operation

To program a Hot Desk phone:

1. In Admin. Programming, assign digital phone as Hot Desk (Dummy Terminal).
2. Assign the DN number type as SADN – Hot Desk Agent; the Hot Desk Agent Number option will automatically be set to ON.
3. Assign a password for the Hot Desk agent station (if needed).

To login to the System through an inactive Hot Desk Station:

1. Lift the handset or press the [SPEAKER] button.
2. Dial the {Hot Desk Feature Code}.
3. Dial the Agent's Station number and password.
4. Dial '*' or press [SAVE] button; the agent will be logged in.

To logout through the active Hot Desk:

1. Dial the {Hot Desk Feature Code}.
2. Dial the call forward destination for Agent calls (Dial '#' to not assign a call forward destination)
3. Dial '*' or press [SAVE] button; the Hot Desk Station will return to inactive.

Conditions

1. The Hot Desk station can be programmed to log-out automatically if no action has occurred by the Agent for the duration of the Hot Desk Log-out timer.
2. An active (logged in) Agent can login to another inactive Hot Desk phone, however this will log-off Agent activity from the previous Hot Desk location.
3. An Agent may only logout from an activated Hot Desk phone.
4. The Flex button map of the Hot Desk station is fixed and will not take on the configuration associated with the Agent's station.

5. The number of Hot Desk phones is limited by the physical station port number, Hot Desk users are limited by the additional station number of the System capacity.
6. Each Hot Desk phone and Hot Desk user (Agent) requires a separate station number(DN) in the system.
7. When a Dummy Terminal seizes a SADN-type number (Sub-DN), the Hot Desk feature cannot be supported.

Programming

- | | |
|-----------------------|--|
| Numbering Plan | 1. Hot Desk Feature Code (PGM 113) |
| Station | 1. Dummy Terminal (PGM 121–Index8)
2. Station Number Type (PGM 130–Index1)
3. Station Password (PGM 131-Index4)
4. Hot Desk Agent Number (PGM 131–Index8) |
| System Timer | 1.Hot Desk Logout Timer (PGM 220-Index2) |

Related Features

Call Forward

Hardware

Digital Phone

3.44 IN-ROOM INDICATION

Description

When an Executive is in the office, their Secretary can press the programmed LED In-Room Indication button signaling other stations of the Executive's status.

Operation

To program In-Room Indication:

1. Using Admin. Programming, set a MADN-type DN number to be used as an In-Room Indication button.
2. Enter {DND Status Change code} to register auto-dialing digits for the DN.
3. Assign a flex button for the DN at the Secretary station and to other Stations needing to know the Executive's status.

To Active or Deactivate In-Room Indication:

1. Press the {DN} button; the DND status of the DN will toggle and the LED signal will be changed at the same time.

Conditions

1. Set other Station's MADN Flex button Access Option to Disabled (Incoming only) to make the secretary station the only station able to control the DN state.

Programming

- | | |
|-----------------------|---|
| Numbering Plan | 1. DND Status Change Code (PGM 113) |
| Station | 1. Station Number Type (PGM 130-Index1)
2. Flex Button Assign (PGM 126)
3. Station Auto Dial Attributes (PGM 138) |

Related Features

Hardware

3.45 IP TRANS-CODING

Description

The system employs either the IEEE g.711, g.729 or g.723 codec to digitize and compress voice signals for RTP packets between devices. IP Phone or terminals on DTIM/SLTM incorporate DSP functions to support codec conversion. Available VOIBs include DSP circuitry used to support trans-coding (converting) codecs for incoming VoIP calls to devices. The VOIBs will trans-code the incoming voice codec (g.711, g.723, g.729) to the System codec and reverse the process for outgoing packets. When the external VoIP connection can only support g.729 and the system codec is g.723, the DSP must implement a complex trans-coding operation, which requires 2 DSP channels. In all other cases, trans-coding only requires a single channel per call.

Operation

System

IP Trans-coding is automatic:

Conditions

1. The system codec for the VOIB can be changed anytime within an IP call.
2. The VOIB DSP can generate and detect in-band DTMF and Call Progress tones in support of DISA functionality.
3. For complex trans-coding (g.723/g.729), the VOIB DSP will require 2 channels.

Programming

Related Features

Hardware

VOIB8 or VOIB24

3.46 LAST NUMBER REDIAL (LNR)

Description

The last number dialed is stored (up to 32 digits) in the station's Last Number Redial (LNR) buffer. The user may request the system redial the last dialed number without the need to dial the number.

Digital Phone users can display stored LNR numbers on the phone LCD using the [REDIAL] or [SPEED] button and [VOL UP]/[VOL DOWN] buttons, to select the number to dial from the list and place a call.

Operation

Digital Phone

To use LNR using [REDIAL] button:

1. Lift the handset or press the [SPEAKER] button.
2. Press the [REDIAL] button.
3. Press the [VOL UP]/[VOL DOWN] button to highlight the desired number.
4. Press [SAVE] or [REDIAL] to dial the number highlighted.

To use LNR with [SPEED] button:

1. Lift the handset or press the [SPEED] button.
2. Dial '**'

SLT

To use LNR:

1. Lift the handset.
2. Dial the {Last Number Redial code}.

Conditions

1. For Digital Phones with LCD display, the LNR redial buffer will store duplicate numbers unless dialed consecutively.
2. When the CO/IP line used for the original call is busy, the System will select an idle line from the same CO/IP line Group to place the call.
3. Using LNR will cancel Automatic Called Number Redial if active.
4. The LNR is stored in volatile memory, so it is not protected in case of a power failure
5. Manually dialing a Flash during an outgoing call will cause only those digits dialed after the Flash to be stored in the LNR buffer.
6. LNR applies to both CO and VoIP calls.

Programming

- | | |
|-------------|-----------------------------------|
| Tenant Data | 1. Redial Method (PGM 281-Index6) |
|-------------|-----------------------------------|

Related Features

- Saved Number Redial (SNR)
- Station Speed Dial
- System Speed Dial

Hardware

3.47 LEAST COST ROUTING (LCR)

Description

The LCR feature is supported using digit-conversion; the rule can be set differently according to the Day/Night/Timed mode or LCR Time. If digit conversion is enabled, the System will seize the CO/IP line after digit-conversion is completed. Meanwhile, the Dummy Dial tone can be provided if programmed.

Operation

Digits are converted automatically based on the Digit Conversion table.

Conditions

1. The digit conversion apply rule is applied to bin base of digit conversion table.

Programming

- | | |
|-------------------|--------------------------------------|
| Table Data | 1) Digit Conversion Table (PGM 251) |
| | 2) Digit Conversion option (PGM 252) |
| | 3) LCR Time Table (PGM 255) |

Related Features

- Digit Conversion
- CO/IP Access
- Station Flexible Buttons

Hardware

3.48 LINKED STATION PAIRS/GROUP

Description

One MADN can be assigned to 10 stations at max so 10 stations can make a linked group, not only paired. If all of the stations set the DN to their Prime-DN, all of 10 stations act like the same station. If one of the stations uses the number, other stations cannot use the same number. Only different Sub-DN can be used at the same time.

Operation

This is automatically supported by the system database.

Conditions

1. If a member of MADN presses the [DND] button while ringing, only the member station's ringing is stopped.

Programming

- | | |
|----------------|---|
| Station | 1. Prime Number Button (PGM 123-Index1) |
| | 2. Flex Button Assign (PGM 126) |
| | 3. Station Number Type (PGM 130-Index1) |

Related Features

Intercom Caller Controlled ICM Signaling

Hardware

3.49 LOUD BELL CONTROL (LBC)

Description

The hardware is equipped with a relay that activates an External Control Contact. The contact is assigned to one of several functions including a Loud Bell Control. If used as a Loud Bell Control, the contact will activate when:

- External Page is accessed
- Assigned Station receives a call (LBC)

Operation

System

When set, relay Operation is automatic.

Conditions

1. A Single-Assign Directory Number (SADN) can be assigned for the LBC feature.
2. A SIP Station cannot be assigned for LBC feature.
3. One relay contact is available (rated at 1 amp, 24 VDC), shown in Table 2.2.4-1.
4. When assigned to activate as LBC, CO Incoming ring and Intercom calls to the assigned station will activate the contacts.
5. Dummy Station for Hot Desk cannot be used as LBC Station.

Programming

- | | |
|---------------|---------------------------------------|
| System | 1. External Contact Control (PGM 228) |
|---------------|---------------------------------------|

Related Features

Door Open

Hardware

External Control Contact connected to an external loud bell.

3.50 MOBILE EXTENSION

Description

A mobile phone may be registered to a station allowing the mobile phone to place and receive calls through the system. DID calls are sent to the user's Phone and the active registered mobile phone simultaneously.

Mobile phone users can access the facilities of the system to place internal and external calls as well as activate/access features. To access system facilities and resources, the mobile user calls the DID number of the corresponding Phone. When the call is received, the system matches the CLI to the mobile phone and provides the mobile user with a system dial tone.

One station can have up to 2 external numbers for their mobile extension. If a mobile extension is being used, the station is in busy state, and the LED is flash steady ON.

Operation

Digital Phone

To activate a registered mobile extension from the user's station:

1. Press the **[PGM]** button.
2. Dial 51 {Mobile Extension code}
3. Dial mobile phone index (1 or 2).
4. Dial digit '1' to activate, '0' to deactivate.
5. Press the **[SAVE]** button.

To register a mobile phone number:

1. Press the **[PGM]** button.
2. Dial 52 {Mobile Extension Registration code}.
3. Dial mobile phone index (1 or 2).
4. Dial the mobile phone number with CO access code.
5. Press the **[SAVE]** button.

To place a call from the mobile extension using the system:

1. Dial the DID number of the station, the system will check the CLI information, answer the call and the user will receive intercom dial tone.
2. Place internal or external call as normal.

To Transfer a call from the mobile extension using the system:

1. Dial {Mobile Flash code}.
 3. Dial the desired extension, the call is transferred and the mobile phone returns to idle.
- NOTE:** The mobile phone may reconnect by dialing the {Mobile Flash Code}.

System

Incoming DID calls are sent to active mobile phones automatically.

Conditions

1. When the mobile phone places an external call through the system, the CLI of the corresponding station is used.
2. The Mobile Extension features are supported via system digital lines only.
3. Message Wait and Callback cannot be activated for use with a mobile phone.
4. The Mobile Extension feature is not supported over a distributed network environment.

5. When an incoming ISDN DID call is received, the system will access an ISDN line and place a call to the mobile phone; an ISDN line must be available for the system to notify the mobile user of the incoming call.
6. Hold and Transfer Recalls to the mobile phone are sent to mobile phone and Mobile Extension and Mobile phone simultaneously.
7. Station Group calls can be routed to the active Mobile Extension.

Programming

System Data

1. Mobile Attributes (PGM 236)

Station Data

1. Mobile Extension Access (PGM132-Index6)
2. Mobile Extension Number Attributes (PGM 146)

Related Features

Do Not Disturb (DND)
Station Message Wait/Call Back
Attendant Recall
Distributed Control Network

Hardware

3.51 MULTIPLE LANGUAGE SUPPORT

Description

With the VMIB, the system can support three (3) languages simultaneously. Prompts in the desired languages are loaded into the VMIB memory along with the Language Selection prompts. To assure the proper language is employed, the Language Selection prompt is played when an incoming call is assigned to be answered by a DID, DISA, Auto Attendant or Station Hunt group announcement. The Language Selection prompts are played in multiple phrases, one in each of the equipped languages, with a request for the caller to input a digit to select the appropriate language. The system then employs the defined announcement (DID, DISA, etc.) recorded for the selected language.

Operation

System

System automatically plays the Language Selection prompts and plays prompts in the selected language.

To record a VMIB Multi-Language Selection prompt at the Attendant:

1. Press the [PGM] button.
2. Dial 062 {Record VM Announcement code}.
3. Dial the VMIB Slot number.
4. Dial the VMIB Multi Language selection Announcement number (01-70).
5. Dial the Language Type number (1-3).
6. Press the '#' key.

7. After the beep-tone, record the desired message.
8. Press the **[SAVE]** button to stop recording and save the message.

Conditions

1. Multi-language support is available with the VMIB/AAIB.
2. Separate announcements must be recorded by the Attendant for each language supported.

Programming

- | | |
|---------------------|---|
| Station Data | 1. Station VMIB Prompt Language Index (PGM 145) |
| CO Data | 1. CO VMIB Prompt Language Index (PGM 161-Index8) |

Related Features

VMIB Integrated Auto Attd/Voice Mail

Hardware

VMIB

3.52 MULTIPLE VOICE MAILBOX SUPPORT

Description

A station can access any Voice Mailbox by dialing the {VMIB Access} code, the mailbox number and password. Phone users may assign one or more Flex buttons to access a specific mailbox.

Operation

Digital Phone

To access a Voice Mailbox:

1. Lift handset or press the **[SPEAKER]**.
2. Dial the {VMIB Access} Code
3. Dial mailbox (station) number.
4. Dial the mailbox(station) password
5. Press the '*' key (password end mark)

To assign a {VMAILBOX} Flex button:

[PGM] + {FLEX} + Button Feature Type (1) + {VMIB Access} Code + Mailbox(station) number + Mailbox(station) password + **[SAVE]**

To access a Voice Mailbox using the {VMAILBOX} Flex button:

1. Lift the handset or press the **[SPEAKER]**.
2. Press the {VMAILBOX} Flex button.
3. Dial the Mailbox password.

Conditions

Programming

Related Features

VMIB Voice Mail

Hardware

Phone

VMIB

3.53 MUSIC-ON-HOLD (MOH)

Description

When a call is placed on Hold, the System will deliver audio from the defined MOH source. In this way, the connected user can determine that the connection is still active.

The system has connections for one music source. The MOH can be either an internal or external source connected to either of the MOH inputs.

Additionally, a message recorded in the VMIB can be employed as MOH along with Background Music (BGM). The Attendant records the VMIB announcement for MOH and VMIB MOH is assigned as the MOH source. Separate messages can be recorded for each of the 3 languages supported by the system.

Operation

System

When set, Operation of MOH is automatic:

To record a VMIB announcement for MOH:

1. Press the **[PGM]** button.
2. Dial the {VMIB Record Feature Code}.
3. Dial the VMIB Slot number.
4. Dial the VMIB Multi-Language selection Announcement number.
5. Dial the Language Type number, only required with multi-language support; the current announcement is played followed by the "Press # to record" prompt.
6. Press the '#' key.
7. After the beep-tone, record the desired message.
8. Press the **[SAVE]** button to stop recording and save the message.

Conditions

1. There are 5 kinds of MOH,
 - Normal Tone
 - Prompt
 - Announcement
 - VMIB MOH

- SLT MOH
2. MOH can be activated by programming the Hold Tone for MOH within the Tone Table.
 3. Plugging-in the BGM RCA jack on the front panel of the MPB and the BGM input on the rear panel RJ-11 jack are common; however, the external music source should only be connected to one of the inputs.

Programming

- | | |
|--------------------|---|
| Table Data | 1. Tone Table (Web Admin. PGM 290-Index49-61) |
| System Data | 1. Music Source (PGM 229) |

Related Features

- Hold
- Multiple Language Support

Hardware

External Music source is connected to MPB music source input.

3.54 NETWORK MANAGEMENT SYSTEM (NMS)

Description

The Network Management System (NMS) is a Web-based application for monitoring and managing multiple systems using standard Simple Network Management Protocol (SNMP). NMS is an efficient and convenient tool employing standards based protocols and a Web-based architecture to permit Administrators remote access to systems using any common Web browser. NMS monitors the multiple systems displaying real-time detailed status information for the system devices and channels. NMS maintains a log of alarm and fault events defined by the administrator and can alert administrators of potential service-affecting faults. In addition, call statistics are maintained and can be reported with various tables and graphs.

Operation

Once configured, Operation of NMS is automatic.
Administrative operations are covered in the NMS Manual.

Conditions

1. NMS is subject to the conditions outlined in the NMS Manual.

Programming

- | | |
|---------------|---------------------------------|
| System | 1. SNMP Attributes (Web Admin.) |
|---------------|---------------------------------|

Related Features

- Diagnostic/Maintenance

Hardware

3.55 NETWORK SECURITY & PRIORITY

Description

The System supports several security and priority protocols. Characteristics that can be established are:

- IEEE 802.1p/Q, VLAN – sets Virtual LAN tag and priority for Ethernet frame
- Diffserv – sets Diffserv Code Point priority for IP packet
- IPSec – enables IPSec to establish IPSec tunnel and encryption of IP packet
- SRTP – enables Secure RTP for RTP packet payload using AES (Advanced Encryption Service).

Operation

System

Once configured, Operation of Security and Priority is automatic.

Conditions

1. For Web Admin, the password is encrypted using the LG-Ericsson Java Virtual Encryption plug-in. A Java Virtual Machine (MS or Sun) must be installed in the User's PC to support password encryption.
2. Security and priority characteristics can be set for all devices, local or remote.
3. The implementation of IPSec employs a proprietary Key exchange protocol from the MP to the System device.

Programming

System Data

1. Web Password Encryption (PGM223-Index1)

Related Features

3.56 ONE DIGIT SERVICE

When a User calls a Station and receives a Busy signal, the User can access the following features by dialing one digit:

- Camp-on
- Call Wait
- Voice Over
- Intrusion
- Pilot Hunt Call

3.56.1 Camp-On

Description

Camp-On is used to notify a Busy Station that a call is waiting to be answered. The busy station is notified of the waiting call by a Camp-On tone. The camp-on station is placed in the Hold state while waiting.

Operation

Digital Phone/SLT

To activate a Camp-On while receiving an Intercom busy tone:

1. Press the {Camp-On} button, both the called and calling stations will receive Camp-On tone.

Conditions

1. The user may only Camp-On to a station in the busy mode; a user may not Camp-On to a station in DND, conference, or receiving a Page, etc.
2. A Camp-On tone is sent each time the calling user presses the {Camp-On} button.

Programming

System Data

1. Camp-On Access (PGM 133-Index8)

Tenant Data

1. Intercom Busy One-Digit Attributes (PGM 237-Index1)

Related Features

Do Not Disturb (DND)

Hardware

3.56.2 Call Wait

Description

Call Wait is used to notify a busy station that a call is waiting to be answered. The busy station is notified of the waiting call by a Call Wait tone. For Digital Phone users, the **[HOLD]** button LED will flash.

The called station can respond by either:

- Answering the waiting call (which places the active call on Hold first),
- Activating One-Time DND
- Ignoring the Camp-On tone.

Operation

Digital Phone

To activate a Call Wait while receiving Intercom busy tone:

1. Press the {Call Wait} button, called and calling stations will receive the Call Wait tone.

To answer a Call Wait after receiving the Call Wait indication:

1. Press the [HOLD] button; the first active call is placed on hold and the station is connected with the Call Waiting station.

Digital Phone/SLT

To activate a Call Wait while receiving Intercom busy tone:

1. Press the {Call Wait} button, called and calling stations will receive the Call Wait tone.

To answer a Call Wait after receiving the Call Wait indication:

1. Press the hook-switch; the first active call is placed on Hold, and the station is connected with the Call Waiting station.

Conditions

1. The user may only Call Wait to a station in the busy mode; a user may not Call Wait at a station in DND, conference, or receiving a Page, etc.
2. A Call Wait tone is sent each time the calling user presses the programmed {Call Wait} button.

Programming

System Data

1. Call Wait Access (PGM 133-Index7)

Tenant Data

1. Intercom Busy One-digit Attributes (PGM 237)

Related Features

Do Not Disturb (DND)
Intercom Call (ICM Call)
Voice Over

Hardware

3.57 PRE-DEFINED & CUSTOM TEXT DISPLAY MESSAGES

Description

When not available, a user can pre-select a text message to be shown on the LCD of an incoming caller's Digital Phone display. When a user activates Text Display Messages, incoming intercom calls will signal the user with normal ringing, and the LCD of the calling station will display the selected message. There are ten Pre-defined messages (01-10), ten System-wide Custom messages and one User-defined Custom message. Several of the ten Pre-defined messages allow for auxiliary information such as a time, date or number.

System-level Custom Messages may be entered from the Attendant or Administrator's phone or via the Web Admin. The User's Custom Message can be assigned from their own Station phone as well as at the Attendant or the Administrator.

Digital Phone users may assign a Flex button as a **{Preselected Message PGM}** button.

The Pre-defined messages are:

Message Number	Display	Comment
01	LUNCH RETURN AT hh:mm	hh:mm can be set with return time
02	ON VACATION RETURN AT DATE mm:dd	mm:dd can be set with return date

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Feature Description & Operation

Issue 1.1

Message Number	Display	Comment
03	OUT OF OFFICE RETURN AT TIME hh:mm	hh:mm can be set with return time
04	OUT OF OFFICE RETURN AT DATE mm:dd	mm:dd can be set with return date
05	OUT OF OFFICE RETURN UNKNOWN	
06	CALL (enter up to 24 digits)	Can be set with destination to call.
07	IN OFFICE STA xxxx	xxxx is set with station number
08	IN MEETING RETURN AT TIME hh:mm	hh:mm can be set with return time
09	AT HOME	
10	AT BRANCH OFFICE	

Operation

Digital Phone

To assign a Flex button for Display Messages:

PRESS **[PGM]** + **{FLEX}** + Button Feature Type(1) + {Preselected Message PGM Code} +
[SAVE]

To activate a Display Message that will be presented to incoming callers:

1. Press the **[PGM]** button.
2. Dial 41 {Display Message code}.
3. To scroll through the available messages, press **[VOL UP]/[VOL DOWN]** button.
4. Dial the Message number (0-9, or * for User's Custom Message).
5. Enter auxiliary input (hh:mm, mm:dd, etc. as needed).
6. Press the **[SAVE]** button.

To cancel an active Display Message:

1. Press the flashing **[DND]** button.
- OR
2. Press the **[PGM]** button.
 3. Dial 41 {Display Message code}.

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Feature Description & Operation

Issue 1.1

4. Press the # key.
5. Press the [SAVE] button.

To define the User Custom Text Message (*):

1. Press the [PGM] button.
2. Dial 42 {Custom Message program code}.
3. Enter the Message contents (up to 16 characters, refer to Alphanumeric Entry Chart).
4. Press the [SAVE] button; confirmation tone is heard and the new User Custom Text Display Message is stored.

NOTE: Alphanumeric characters may be entered using the following as a guide.

Alphanumeric Entry Chart

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

Graphical LCD phones (ex., LKD-30DH) may use the following table for entering alphanumeric characters.

Alternate Alphanumeric Entry Chart

Dial Pad button	Letter Type									
	Uppercase (ABC)				Lowercase (ABC)				Num	
	Button depressions									
	1	2	3	4	1	2	3	4	1	
1	@	:	/	<	@	:	/	<	1	
2	A	B	C		a	b	c		2	
3	D	E	F		d	e	f		3	
4	G	H	I		g	h	i		4	
5	J	K	L		j	k	l		5	
6	M	N	O		m	n	o		6	

Dial Pad button	Letter Type									
	Uppercase (ABC)				Lowercase (ABC)				Num	
	Button depressions									
	1	2	3	4	1	2	3	4	1	
7	P	Q	R	S	p	q	r	s	7	
8	T	U	V		t	u	v		8	
9	W	X	Y	Z	w	x	y	z	9	
0	.	,	?	!	.	,	?	!	0	
*	*				*				*	
#	#				#				#	

SLT

To activate a Display Message:

1. Lift the handset.
2. Dial {SLT Programming code}.
3. Dial 41 {Display Message code}.
4. Dial the Message number (0-9, or * for User's Custom Message).
5. Enter auxiliary input (hh:mm, mm:dd, etc. as needed).
6. Press the hook-switch; confirmation tone is heard.

To cancel an active Display Message:

1. Lift the handset.
2. Dial {SLT Feature Cancel code}.

To enter the User Custom Message (*):

1. Lift the handset.
2. Dial {SLT Programming code}.
3. Dial 41 {Display Message code}.
4. Enter the Message contents (up to 16 characters, refer to Alphanumeric Entry Chart).
5. Press the hook-switch, confirmation tone is received.

System Attendant

To activate Display Messages for other stations:

1. Press the **[PGM]** button.
2. Dial 051 {Attendant Display Message code}.
3. Dial the desired Station range.
4. To scroll through the available messages, press **[VOL UP]/[VOL DOWN]** button.
5. Dial the Message number (0-9, or * for User's Custom Message).
6. Dial auxiliary input (hh:mm, mm:dd, etc. as needed).
7. Press the **[SAVE]** button.

To cancel active Display Messages for other stations:

1. Press the **[PGM]** button.
2. Dial 052 {Attendant Display Message Cancel code}.
3. Dial the desired Station range.
4. Press the **[SAVE]** button.

Conditions

1. Alphanumeric characters are displayed as they are entered.
2. Display Message is cancelled if the User activates DND or Call Forward.
3. Custom Text Display Messages and Display Message status are stored in non-volatile memory to protect against loss during power failure.
4. Incoming Caller Station will display the message.
5. SLTs are notified of an active Display Message with a stutter dial tone, while Digital Phones will have a flashing **[DND]** button when there is an active Text Display Message.
6. Activating a Text Display Message does not affect normal operation of the station.
7. Pre-defined Messages 01-04, and 06-08 permit the user to input auxiliary information such as time, date or number, as applicable.
8. The Attendant station can activate a Text Display Message for other stations, however this feature is not available to an Attendant.

Programming

Related Features

Do Not Disturb (DND)
Call Forward
Speed Dial

Hardware

Digital Phone required receiving Display Messages

3.58 REGISTERING IP DEVICES & FRACTIONAL MODULE TABLES

The System can register IP Phones, DTIM, and SLTM using the MAC address or Phontage, and SIP Phone with ID/password.

3.58.1 Registration with MAC Address

Description

Using the defined MAC address registration, the system allows IP Phones, DTIM and SLTM devices with matching MAC addresses to register regardless of the Database Protection Switch (DPS) position.

Operation

Registration is automatic.

Conditions

Programming

System Info

1. IP Phone/Phontage Registration Table (PGM 106)
2. DTIM/SLTM Registration Table (PGM 107)
3. Logical Slot Assignment (PGM 104-Index2)
4. Slot Assignment & Logical SLOT Assignment (PGM 101, PGM 103)

Related Features

Hardware

3.58.2 Registration with ID/Password

Description

The System can be programmed to register a Phontage or SIP Phone using an ID & Password. Devices with matching ID & Password can be registered regardless of the Database Protection Switch position.

Operation

Registration is automatic.

Conditions

Programming

System Info

1. IP Phone/Phontage Registration Table (PGM 106)
2. Logical Slot Assignment (PGM 104-Index2)

Related Features

Hardware

3.58.3 Registration with Station Number

Description

System allows IP Phone registration if the IP Phone Station number matches the Station number designated regardless of the Database Protection Switch position.

Operation

Registration is automatic.

Conditions

Programming

System Info

1. Logical Slot Assignment (PGM 104-Index2)
2. IP Address Plan (PGM 101)
3. Slot Assignment (PGM 103)

Related Features

Hardware

3.59 REMOTE DEVICE ZONE MANAGEMENT

Description

Remote devices, in particular those not reachable by the System, are managed by grouping devices by various characteristics in a Zone. Placing devices into Zones simplifies management allowing definition of common characteristics to the devices within the zone. Zone attributes include:

- Nation Code
- Language
- RTP Relay
- RTP Relay group

Operation

When set, Zone operation is automatic.

Conditions

1. It is recommended to assign CO/IP lines and Stations of a Tenant group in the same Device Zone.
2. Wake-up time is based on the time displayed in the Station LCD.

Programming

Zone Data

1. Zone Attributes (Web Admin PGM 395)
2. Zone RTP Relay Group (Web Admin PGM 396)
3. Inter-Zone Attributes (Web Admin PGM 397)
4. Station Zone Attributes (Web Admin PGM 399)

Related features

Hardware

3.60 REMOTE SERVICES, MANAGED NET

Description

IP Phones, DTIM and SLTM can run on a System located on a different LAN segment or WAN without the need for additional equipment.

The System can be assigned with the IP address of the default gateway (router) running the remote device. The system will register the device through the router. Using this configuration, the device can establish a connection with the system and then establish peer-to-peer communication with other devices as needed.

Operation

Operation of this feature is automatic.

Conditions

1. The managed network must provide appropriate security, bandwidth and QoS.

Programming

System Info

1. IP Phone/Phontage Registration Table (PGM 106)
2. DTIM/SLTM Registration Table (PGM 107)
3. Logical Slot Assignment (PGM 104-INDEX2)

Related Features

Hardware

3.61 REVERTIBLE RING

Description

This feature can be used to identify the extension DN especially when a SLT is in use. Additionally, the User can verify the incoming ring signal is working correctly.

Operation

To listen to Revertible Ring:

1. Lift the Handset or press **[SPEAKER]**.
2. Dial the station DN; confirmation tone is heard
3. Replace Handset (go on-hook); incoming ring will be presented.
4. Lift Handset or press **[SPEAKER]**, confirmation tone is heard.

Conditions

Programming

Table Data

1. Ring Table, Revertible Ring (Web Admin PGM 265-11)

Related Features

Hardware

3.62 SPEED DIAL

3.62.1 Speed Dial Pause Insertion

Description

A pause dialing command may be inserted in a Station or System Speed Dial number. When encountered, the System will stop dialing the Speed Dial number for the assigned “pause” duration. Multiple pauses ([HOLD] button depressions) may be inserted into a Speed Dial number.

Operation

System

When set, Pause operation is automatic.

Conditions

1. Timed pause is used only with analog CO lines.

Programming

Station Data	1. Speed Access (PGM 134-Index1)
Table Data	1. System Speed Dial Table (PGM 257)

Related Features

Station Speed Dial System Speed Dial

Hardware

3.62.2 Station Speed Dial

Description

Each User can store commonly dialed numbers for easy access using Station Speed Dial bins. Each Station has access to 50 Speed Dial numbers. Each Speed Dial number can be up to 24 digits in length and may include special instruction codes.

Special instruction codes are:

- Flash as 1st digit: Activates dial tone detect.
- Pause [CALLBACK]: Inserts a pause dialing command.
- ‘*’ not 1st digit: Switches from Pulse to DTMF dialing.

Digital Phone users may assign a Flex button for One-Touch access to a specific Speed Dial bin.

Operation

Digital Phone

To dial using a Station Speed Dial:

1. Lift handset or press the [SPEAKER] button.
2. Press the [SPEED] button.
3. Dial the desired bin number.

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Feature Description & Operation

Issue 1.1

To program a Station Speed Dial number:

1. Press the [PGM] button.
2. Press the [SPEED] button.
3. Dial the Speed Dial bin number.
4. Dial {CO/IP Line/Group Access code}.
5. Enter the number to be stored.
6. Press the [SAVE] button.
7. If desired, enter a name (refer to Alphanumeric Entry Chart).
8. Press the [SAVE] button.

Alphanumeric characters may be entered to name the Speed Dial number using the following as a guide.

Alphanumeric Entry Chart

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

Graphical LCD phones (ex., LKD-30DH) may use the following table for entering alphanumeric characters.

Alternate Alphanumeric Entry Chart

Dial Pad button	Letter Type									
	Uppercase (ABC)				Lowercase (ABC)				Num	
	Button depressions									
	1	2	3	4	1	2	3	4	1	
1	@	:	/	<	@	:	/	<	1	
2	A	B	C		a	b	c		2	
3	D	E	F		d	e	f		3	
4	G	H	I		g	h	i		4	
5	J	K	L		j	k	l		5	
6	M	N	O		m	n	o		6	
7	P	Q	R	S	p	q	r	s	7	

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Feature Description & Operation

Issue 1.1

Dial Pad button	Letter Type									
	Uppercase (ABC)				Lowercase (ABC)				Num	
	Button depressions									
	1	2	3	4	1	2	3	4	1	
8	T	U	V		t	u	v			8
9	W	X	Y	Z	w	x	y	z		9
0	.	,	?	!	.	,	?	!		0
*	*				*					*
#	#				#					#

To program a Station Speed Dial number using the 3-soft-key (LCD display phones):

1. Press the **{DIR}** Soft key.
2. Dial 1 or Press the **{OK}** Soft key with Station Speed selected.

STATION 2000 (T)		
14 MAR 09		11:45 am
LOG	DIR	REDIAL->

3. Press the **{ADD}** Soft key.

1>STATION SPEED	
2 SYSTEM SPEED	
OK	EXIT

4. Dial the Speed Dial bin number or press **{OK}** button.

SPD_NO	LAST(*)	SAVE(#)
BACK	SEARCH	ADD

5. Dial the {CO/IP Group Access code}.
6. Dial the number to be stored.
7. Press the **[SAVE]** button.
8. If desired, enter a name (refer to Alphanumeric Entry Chart).
9. Press the **[SAVE]** button.

ENTER SPEED BIN NO(000)		
BACK	OK	EXIT

SLT

To dial using Station Speed Dial:

1. Lift handset.
2. Dial {SLT Speed Dial access code}.
3. Dial the desired bin number.

To program a Station Speed Dial number:

1. Dial {SLT Programming code}.
2. Dial {SLT Speed Dial access code}.
3. Dial the Speed Dial bin number.
4. Dial the {CO/IP Group Access code}.
5. Dial the number to be stored.

6. Press for hook-switch.
7. If desired, enter a name (refer to Alphanumeric Entry Chart).
8. Press for hook-switch.

Conditions

1. Accessing an empty Speed Dial bin will return an error tone.
2. Speed Dial numbers can reference a specific CO/IP Group entered by the user. If the assigned line is busy, a line from the same group will be selected. If all lines in the group are busy, the user may queue for the next available line.
3. All Speed Dial numbers are stored in protected memory in case of power loss.
4. A name can be entered for a Speed Dial number to permit access from the Dial-by-Name directory.

Programming

Station Data	1. Speed Access (PGM 134-BTN 1)
Table Data	1. System Speed Dial Table (PGM 257)

Related Features

Dial-by-Name
Last Number Redial (LNR)
Dial Pulse to Tone Switchover
Saved Number Redial (SNR)
Speed Dial Pause Insertion
System Speed Dial

Hardware

3.62.3 System Speed Dial

Description

Commonly dialed numbers can be stored by the System Attendant or by the Administrator using Web Admin. for easy access to Stations allowed use of System Speed Dial bins. Each Speed Dial number can be up to 24 characters in length and may include special instruction codes.

Special instruction codes are:

- Flash as 1st digit: Activates dial tone detect.
- Pause [CALLBACK]: Inserts a pause dialing command.
- '*' not 1st digit: Switches from Pulse to DTMF dialing.

Digital Phone users may assign a Flex button for One-Touch access to a specific System Speed Dial bin.

Operation

iPECS - MG Release 1

Feature Description & Operation

Issue 1.1

Digital Phone

To dial using a System Speed Dial:

1. Lift handset or press the **[SPEAKER]** button.
2. Press the **[SPEED]** button.
3. Dial the desired bin number

SLT

To dial using a System Speed Dial:

1. Lift handset.
2. Dial {SLT Speed Dial access code}.
3. Dial the desired bin number

Attendant

To program a System Speed Dial number:

1. Press the **[PGM]** button.
2. Press the **[SPEED]** button.
3. Dial the Speed Dial bin number
4. Dial the {CO/IP Group Access code}.
5. Dial the number to be stored.
6. Press the **[SAVE]** button.
7. If desired, enter a name (refer to Alphanumeric Entry Chart).
8. Press the **[SAVE]** button.

Alphanumeric characters may be entered using the following as a guide.

Alphanumeric Entry Chart

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

Graphical LCD phones (ex., LKD-30DH) may use the following table for entering alphanumeric characters.

iPECS - MG Release 1

Feature Description & Operation

Issue 1.1

Alternate Alphanumeric Entry Chart

Dial Pad button	Letter Type									
	Uppercase (ABC)				Lowercase (ABC)				Num	
	Button depressions									
	1	2	3	4	1	2	3	4	1	
1	@	:	/	<	@	:	/	<	1	
2	A	B	C		a	b	c		2	
3	D	E	F		d	e	f		3	
4	G	H	I		g	h	i		4	
5	J	K	L		j	k	l		5	
6	M	N	O		m	n	o		6	
7	P	Q	R	S	p	q	r	s	7	
8	T	U	V		t	u	v		8	
9	W	X	Y	Z	w	x	y	z	9	
0	.	,	?	!	.	,	?	!	0	
*	*				*				*	
#	#				#				#	

To program a System Speed Dial number with soft-key

1. Press the **{DIR}** Soft key.
2. Dial 1 or Press the **{OK}** Soft key with System Speed selected.
3. Press the **{ADD}** Soft key.
4. Dial the Speed Dial bin number or press **{OK}** button.

```

                STATION 2000 (T)
14 MAR 09          11:45 am
LOG      DIR      REDIAL->
    
```

```

1 STATION SPEED
2>SYSTEM SPEED
                OK      EXIT
    
```

3. Press the **{ADD}** Soft key.

```

SPD_NO  LAST(*)  SAVE(#)
BACK    SEARCH  ADD
    
```

4. Dial the Speed Dial bin number or press **{OK}** button.

```

ENTER SPEED BIN NO(2000)
BACK    OK      EXIT
    
```

5. Dial the **{CO/IP Group Access code}**.
6. Dial the number to be stored.
7. Press the **[SAVE]** button.
8. If desired, enter a name (refer to Alphanumeric Entry Chart).
9. Press the **[SAVE]** button.

Conditions

1. Accessing an empty Speed Dial bin will return an error tone.

2. A Speed Dial number can use a specific CO/IP Group entered by the user. If the assigned line is busy, a line from the same group will be selected. If all lines in the group are busy, the user may queue for the next available line.
3. All Speed Dial numbers are stored in protected memory in case of power loss.
4. A name can be entered for a Speed Dial number to permit access from the Dial-by-Name directory.

Programming

- | | |
|---------------------|--------------------------------------|
| Station Data | 1. Speed Access (PGM 134-Index1) |
| Table Data | 1. System Speed Dial Table (PGM 257) |

Related Features

Dial-by-Name
Last Number Redial (LNR)
Dial Pulse to Tone Switchover
Saved Number Redial (SNR)
Speed Dial Pause Insertion
System Speed Dial

Hardware

3.63 STATION CALL COVERAGE

Description

The DN button at a Station can be set for incoming calls only by disabling outgoing calls. If the MADN-type DN button has a delayed ring option, the button will operate as a call coverage button. This feature must be programmed by the Administrator; individual users cannot set this feature.

Operation

If Programmed, Station Call Coverage is automatic.

Conditions

Programming

- | | |
|----------------|--|
| Station | 1. Station Number Type (PGM 130-Index1) |
| | 2. Flex Button Assign (PGM 126) |
| | 3. DN Flex button Ring Option (PGM 126-Index2) |
| | 4. DN Flex button Access (PGM 126-Index3) |

Related Features

Hardware

3.64 SYSTEM GROUPS

Description

Stations can be grouped for call routing, dialing, call pick-up, or other various purposes. The following groups can be defined:

- Station Group: Terminal / Circular / Ring / Longest Idle / VM
- Pilot Hunt Group
- Pick-Up Group
- PTT Group
- Command Conference Group
- Interphone Group
- Paging Group

3.64.1 Station Group

Description

Stations can be grouped so incoming calls may be routed to an idle station in the group. The different types of Station Groups are described:

- Terminal Group – Calls to a station in a Terminal Station Group that encounter an unavailable or go unanswered will be routed through the hunt process. The call will proceed to the next listed station in the group until reaching the last listed station. The queued call may be taken out of the group if set to forward to an overflow destination.
- Circular Group – Calls to a station in the Circular Group will go to the station, if unavailable or unanswered in the hunt no answer time; the call will be directed to the next station defined in the group. The call will continue to hunt until each station in the group has been tried. The queued call may be taken out of the group if set to forward to an overflow destination.
- Ring Group – An incoming call to any station in the Group will cause all stations in the group to ring, and any station in the group may answer the call. Multiple calls can be received by a Station Ring Group and can be serviced in any order according to Station availability. The queued call may be taken out of the group if set to forward to an overflow destination.
- Longest Idle (UCD) Group – Calls are sent to the group by dialing the Hunt group Number or assigning CO lines to directly terminate at 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 a station in the group to become available. The queued call may be taken out of the group if set to forward to an overflow destination.
- Station VM Group – Can be enabled to support an external Auto Attendant/Voice Mail (AA/VM) system that employs SLT ports to interface with the system. An External AA/VM group is assigned for either Circular, Terminal, or Longest Idle hunt. The External AA/VM may employ either in-band signaling over the audio channel or SMDI protocol with a signaling connection to the System RS-232 channel.

Operation

When programmed, Station Group operation is automatic.

To pick-up a call that comes in to the Station Group:

1. When a call is ringing in to the Group, lift handset or press the **[SPEAKER]** button.
2. Stations can be set to be Disabled, receive All calls, receive Intercom calls, or to receive External calls.

Conditions

1. Station Group calls are not routed to member stations that are in DND.
2. A call transferred to a Station Group will follow the routing for the group and will not initiate the Transfer Recall process.
3. Calls to a Station Group receive either a ring-back tone or MOH while queued to the group.
4. Calls can be routed to the defined Overflow destination according to Forward type, Station, Group, etc. If Forward type is set to NOT USED, the call is dropped about each Forward case.
5. Stations can be a member of one or more Station Groups (Multiple Member Assignment).
6. A User can program Queue Count for each member of the Station group (0-99).
7. The System can provide a call to a group member during the Greeting according to the Call in Greeting option.
8. A User can assign a Group Name for each Hunt Group.

Programming

Station	1. DND Access (PGM 132-Index4)
Station Group	1. Executive/Secretary Assign (PGM 241) 2. Executive/ Executive Access (PGM 242) 3. Station Group (PGM 200) 4. Station Group Greeting/Queuing Attribute (PGM 201) 5. Station Group Attributes (PGM 202) 6. VM Group Attributes (PGM 203)
Table Data	1. Announcement Table (PGM 259)
Tenant Data	1. Tone Table (Web Admin. PGM 290)
Numbering Plan	1. Station Group Number (PGM 115)

Related Features

Executive/Secretary
Music On Hold
Call Forward
VMIB Integrated Auto Attd/Voice Mail

Related Features

Digital Phone

3.64.2 Pilot Hunt Group

Description

A Station can be grouped for Pilot Hunt Feature. Users may select incoming calls in the group to re-route to other stations (local or networked), station groups, the VMIB according to ring mode (Day/Night/Timed). A member of the Pilot Hunt Group may have Pilot Hunt Ring Access authority set for call coverage on another member Station in a group.

Operation

If programmed, Pilot Hunt Group is automatic.

Conditions

1. Pilot Hunt Ring Access of a member should be enabled to receive the Pilot Hunt Group calls.
2. Pilot Hunt Group members can register a Forward as a Day destination using the {Pilot H. CFW Register} feature code.
3. If a Station's busy service is set to Pilot Hunt and the Station is in conversation with an internal/external party, and another station calls, the call is routed to an idle member in the same Pilot Hunt.

Programming

Numbering Plan	1. Feature Numbering Plan (PGM 113)
Station Data	1. Call Forward Access (PGM 132-Index2) 2. Pilot Hunt Ring Access (PGM 134-Index6) 3. Busy Service (PGM 131-Index5)
CO Line Data	1. Incoming CO Alternate (PGM 169) 2. Outgoing CO Alternate (PGM 173)
Station Group	1. Pilot Hunt Group (PGM 210) 2. Pilot Group Forward Attributes (PGM 211)
Table Data	1. Ring Table (Web Admin. PGM 265)

Related Features

Call Forward
Pilot Hunt

Hardware

3.64.3 Pick Up Group

Description

A Station can be assigned to a Call Pick-Up group and may pick-up (answer) calls to other stations in the group employing the System's Group Call Pick-Up feature.

Station Groups can be added as Pick -Up Groups with Pick-Up Attributes. Pick-up Groups can be set to pick-up all calls, internal calls only or external calls only.

Operation

To use Group Call Pickup:

1. Dial the {Group Call Pick Up} feature code.

To use Direct Call Pickup:

1. Dial {Direct Call Pick Up} feature code.
2. Dial DN number to pick up the call.

Conditions

Programming

- | | |
|-----------------------|--|
| Station Group | 1. Station Group (PGM 200-Index5)
2. Pickup Group (PGM 204) |
| Numbering Plan | 3. Feature Numbering, Group Pick-Up (PGM 113) |

Related Features

Group Call Pick-Up

Hardware

3.64.4 Push To Talk (PTT) Group

Description

Each Phone can be assigned as a member of one or more of the System Push-To-Talk (PTT) groups. The Phone user may log-in or log-out of any one, or all PTT groups to which it is assigned. Once logged in, the user may place or receive one-way page announcements to/from other users who are logged in to the same PTT group.

Additionally, each user can log in or log out PTT group using the {PTT Group Log-In/Out Feature Code}.

Operation

Log-in PTT Group with Feature Code:

1. Dial the {PTT Group Log-In/Out Feature Code}; the PTT Group status and registration will be displayed.
2. Dial a PTT Group Number.

Log-out PTT Group with Feature Code:

1. Dial {PTT Group Log-In/Out Feature Code}; the PTT Group status and registration will be displayed.
2. Dial * (Log-Out Code).

To make a PTT Group Call:

1. Press the programmed {PTT} button.

2. When finished, press the {PTT} button to end the call.

Conditions

1. Only a SADN can be assigned as a member of Paging Group (PTT Group).
2. SIP Stations cannot be assigned to Paging Groups.
3. PTT Group 0 is a specific PTT Group, so if a member of the PTT group 0 tries to make a PTT announcement, all of the members in all of groups will receive the announcement. Additionally, each group member can make PTT calls, then all members of Group 0 will receive the announcement.

Programming

- | | |
|-----------------------|--|
| Station Group | 1. PTT Group (PGM 208) |
| Numbering Plan | 2. Feature Numbering Plan - PTT Group Log IN/OUT (PGM 113) |

Related Features

Hardware

3.64.5 Command Conference Group

Description

A Station or external telephone number can be assigned as a member of a Command Conference Group.

Stations and external contacts (up to 12 members) can be arranged in groups so that a user may create a conference with all members of the group through a single call. Additionally, the user can make paging calls with same group. There are 2 kinds of Command Conference Groups:

- **Command One Way:** A user can make announcements (paging) to members of the Command Group.
- **Command Conference:** A user can make conference calls with members of the Command Group.

On-Hook Service – An internal user can receive a command call while the Station is On-Hook.

When an internal user receives a command group call and the call goes unanswered, the System will make a recall to the user station.

One- or Both-Way Busy – When an internal user receives command group call while in busy status, the command group call is ignored, and can be queued. When an internal user receives a command group call while in busy status, the current call will be disconnected and the command group call automatically is connected.

Operation

To initiate a Command Group (One Way):

1. Dial the {One Way Command Group Call} feature code.
2. Enter the Command Group number.

To initiate a Command Group Conference:

1. Dial the {Conference Command Group Call} feature code.
2. Enter the Command Group number.

Conditions

Programming

- | | |
|-----------------------|--|
| Station Data | 1. Command Group Access (PGM 152) |
| Station Group | 1. Command Conference Group (Web Admin PGM 206) |
| Numbering Plan | 2. Feature Numbering Plan, Command Group (PGM 113) |

Related Features

Hardware

3.64.6 Interphone Group

Description

To call Stations using a simple, one-touch digit, Stations can be gathered into an Interphone Group (up to 10 members).

Operation

To use the Interphone Group feature:

1. Dial {Interphone Group Access} feature code.
2. Dial One Digit.

Conditions

Programming

- | | |
|-----------------------|---|
| Station Group | 1. Interphone Group (PGM 209) |
| Numbering Plan | 2. Feature Numbering, Interphone Group Access (PGM 113) |

Related Features

Hardware

3.64.7 Paging Group

Description

A Station is permitted to access page facilities for each Paging Group, to connect and transmit voice announcements to any or all System Paging Groups.

Operation

To perform an Internal Page:

1. Dial {INT Page code}.

2. Dial Page Group number (01 – 30).

To perform an External Page:

1. Dial {EXT Page code}.

Conditions

1. Only SADN can be assigned as the member of a Paging Group.
2. SIP Station cannot be assigned as member of a Paging Group.

Programming

- | | |
|-----------------------|---|
| Station Data | 1. Page Access (PGM 134-Index2)
2. Meet-Me Access (PGM 134-Index3)
3. Page Group Access (PGM 151) |
| Station Group | 1. Page Group (PGM 205) |
| Numbering Plan | 2. Feature Numbering, Internal Page Calling Answer (PGM 113) |

Related Features

Hardware

3.65 STATION MESSAGE DETAIL RECORDING (SMDR)

3.65.1 Call Cost Display

Description

Each SMDR call record includes a Cost field; a calculated estimate for the cost of the call. When set, the call cost will update in real-time and display on the Digital Phone LCD in place of the call duration.

The cost is determined by:

- Fixed charge per Call Meter Pulse
- ISDN Advice of Charge
- Estimated cost updated based on the Elapsed Call Timer and any assigned costing.

The technique selected to determine cost is based on the type of facility (analog CO, ISDN, or VoIP), services provided by the carrier, and the system database.

Analog CO – When the Call Metering Pulse service is available from the carrier, the system will apply the SMDR Cost per Unit Pulse and the SMDR Decimal to Call Metering received to estimate call cost.

When no “Metering Type” is selected, the system call duration is used with the cost/pulse and decimal values to estimate the cost of the call. The cost is updated periodically using the “Elapsed Call Timer” duration.

ISDN – ISDN providers may support Advice of Charge information in the ISDN Facility Message. If assigned, the system will employ this information to display and output the call cost.

VoIP – For VoIP calls, the system employs the call duration, cost/pulse and decimal values to establish a call cost estimate. The cost is updated periodically at intervals of the Elapsed Call Timer.

Operation

System

When set, Call Cost is estimated automatically and output to Digital Phone displays and the SMDR RS-232 port.

Conditions

1. If enabled, Call Cost display begins after the SMDR Start Timer expires, or at receipt of the first Call Meter Pulse.
2. Once connected to the system, the call duration includes the total time the call is connected including periods when the call is on hold, in queue, etc.
3. To enable Call Cost Display, the SMDR Cost per Unit Pulse and SMDR Decimal must be assigned; when not assigned, call duration is provided by the System.
4. SMDR MAX record message number is 5000; alarm message is automatically received at the Attendant Station if recorded number is 4000 or 4500.

Programming

- | | |
|---------------------|--|
| CO Line Data | 1. Metering Type (PGM 160-Index12) |
| System Data | 1. SMDR Cost per Unit Pulse (PGM 232-Index8) |
| | 2. SMDR Fraction (PGM 232-Index9) |

Related Features

Station Message Detail Recording (SMDR)
Traffic Analysis
CO/IP Call Warning Tone Timer

Hardware

RS-323 device to capture SMDR

3.65.2 SMDR Call Records

Description

SMDR provides detailed information on incoming and outgoing calls. Assignable options in the System database permit recording of all calls, all outgoing calls or toll calls and calls that exceed a fixed duration. Call records are output either upon completion of the call (real-time) or in response

to a request from the System Attendant. SMDR may be sent periodically via e-mail to a defined e-mail address.

The various fields or items for a Call Record are:

- 8-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)
- 7-digit time of day call originator field
- 1 digit call identification digit-first digit in digit dial field
- 24-digit collected dialed digit field
- 5-digit pulse metering count field
- 10-digit call cost field
- 12-digit account code field

Operation

System

For real-time SMDR, records are output after completion of the call.

System Attendant

To print SMDR records (based on Station range):

1. Press the **[PGM]** button.
2. Dial 011 {SMDR Print code}.
3. Enter the desired station range.
4. Press the **[SAVE]** button.

To delete stored SMDR records (based on Station range):

1. Press the **[PGM]** button.
2. Dial 012 {SMDR delete code}.
3. Enter the desired station range.
4. Press the **[SAVE]** button.

To print Non-Station Based SMDR records:

1. Press the **[PGM]** button.
2. Dial 013 {SMDR Print code}.
3. Press the **[SAVE]** button.

To delete Non-Station Based SMDR records:

1. Press the **[PGM]** button.
2. Dial 014 {SMDR delete code}.
3. Press the **[SAVE]** button.

To print ALL SMDR records (all of SMDR):

1. Press the **[PGM]** button.
2. Dial 015 {SMDR print code}.

3. Press the [SAVE] button.

To delete All SMDR records (all of SMDR):

1. Press the [PGM] button.
2. Dial 016 {SMDR delete code}.
3. Press the [SAVE] button.

Conditions

Programming

- | | |
|---------------|------------------------------|
| System | 1. SMDR Attributes (PGM 232) |
|---------------|------------------------------|

Related Features

- Call Cost Display
- Traffic Analysis Authorization Codes (Password)

Hardware

- RS-323 device to capture SMDR

3.66 SYSTEM ADMIN PROGRAMMING

3.66.1 Keypad Administration

Description

The System database can be accessed and modified using the Keypad and Flex buttons of a Digital Phone. The Digital Phone LCD is employed to view items on the System database. The user may be required to enter a password for access to Keypad Admin. Based on a set-up of Multi-level passwords, the User may have access to specified System database program codes.

For detailed information on database administration and maintenance, refer to the **Admin. Programming Manual**.

Operation

Keypad Administration operation is detailed in the Admin. Manual.

Conditions

1. Only stations assigned with Admin. access can enter and change System database items. As a default, the First station (Station 100, Administrator and/or Attendant) can access the database.

Programming

- | | |
|----------------|---|
| Station | 1. Admin. Access (PGM 121-Index5) |
| System | 1. Admin. Access Authority (Web Admin.)
2. System Password (PGM 226) |

Related Features

Web Administration
Multi-Level Admin Access

Hardware

3.66.2 Multi-Level Admin Access

Description

Access to the System Admin. database is password protected. Up to three (3) levels of access to the database can be established by assigning a different password to each level. The Maintenance level has access to the entire database, all maintenance routines, and defines the Admin. Access Authority of the two remaining passwords. A User password can only access specific database items and cannot access Station Program pages. The Admin. password has access to specific database items as well as Station Program pages. In Web Admin, the Maintenance password user can establish the Access Authority for each password selecting the Admin. Program Codes available to each password level.

Operation

Detailed operation of Admin. access and assigning access authority for each level is given in the Admin. Programming Manual.

Conditions

1. Admin. Access Authority is defined only in Web Admin.; it cannot be defined using Keypad Admin.
2. Admin. Access Authority applies to all Admin. access whether accessed via a Digital Phone, ISDN or IP channels.

Programming

- | | |
|----------------|--|
| Station | 1. Keypad Admin (PGM 121-Index5) |
| System | 1. Admin Access Authority (Web Admin.)
2. System Password (PGM 226) |

Related Features

Web Administration

Hardware

3.66.3 Web Administration

Description

The System database can be accessed and modified using a Digital Phone, the LAN interface or an ISDN BRI/PRI line. Both the LAN and ISDN access the System Web server delivers the database as a set of Web pages to the a Web browser. Under the proper conditions, both also allow for remote access to the System database.

For remote access with a LAN interface, the System must be assigned a remotely accessible IP address. The IP address should be fixed either as a public IP address or through a NAPT server with port forwarding. The User's browser should be pointed to the System IP address and the TCP port assigned in the System database.

For an ISDN BRI/PRI, a Point-to-Point Protocol (PPP) connection can be established between a standard ISDN modem and the BRI/PRI board. The connection can be set-up over a normal or DID BRI/PRI line. The System will negotiate access using 2 User IDs and matching passwords assigned in the database. In addition, for DID access, the Called Party number must match the assigned System PPP destination. Once a connection is established over the BRI/PRI, the User Web browser can be opened and pointed to the System IP address and assigned a Port for access to the database.

When accessed, the System will return the Administration Web page. From this page, selecting Admin & Maintenance will return the login page where the user must enter a registered password. Based on the password entered, the user is permitted access to specified System program codes (refer to **Admin. Programming Manual**).

Operation

Operation is detailed in the Admin. Programming Manual.

Conditions

1. For Web Admin., a password can be encrypted using the LG-Ericsson Java Virtual Encryption plug-in. A Java Virtual Machine (MS or Sun) must be installed on the User PC to support password encryption.

Programming

Pre-programmed Data 1. System IP Address Plan (PGM 108)

System

1. WEB Password Encryption (PGM 223-Index1)
2. Admin Access Authority (Web Admin.)
3. BRI/PRI PPP Web Admin. Attributes (PGM 235)

Related Features

Keyset Administration
Multi-Level Admin Access

Hardware

3.66.4 Web User Manual

Description

The Web Admin. User Guide is available on-line as part of Web services. The main Web page permits access to the Web User guide. The guide is an HTML document, which can be modified by replacing the HTML ROM image in the System with an external ROM image in accordance with the appropriate LG-Ericsson R&D STI.

Operation

Operation is detailed in the Admin. Programming Manual.

Conditions

1. Access to the User Guide is not password protected.
2. To support local languages, the HTML ROM image of a translated Web User guide may be loaded into System memory.

Programming

Related Features

Web Administration

Hardware

3.67 SYSTEM DECT

Description

The iPECS-MG System supports office building mobility employing Digital Enhanced Cordless Telecommunications (DECT). LG-Ericsson's DECT Base stations (GDC-400B/GDC-600B) connect to the Wireless Telephone Interface Module (WTIB). The WTIB manages up to 8 base stations; up to 2 WTIBs may be installed in the System. DECT handsets (GDC-400H and GDC-450H) can roam and maintain uninterrupted communications link to features and resources through the base station to the WTIB.

Operation

DECT operation is automatic when configured.

Conditions

1. Multiple WTIBs may be installed to support DECT.

Programming

DECT Data

1. DECT Registration (#0)
2. DECT Attributes (PGM 492).

Related features

Hardware

WTIBs
GDC-400B/GDC-600B Base stations
GDC-400H/GDC-450H Handsets

3.68 SYSTEM NETWORKING

3.68.1 Centralized Control T-NET (LM)

Description

The System supports Centralized Control T-NET (Transparent Network) as a role of the Local Mode (LM). If the iPECS-MG System is set for LM, all modules and terminals which are physically connected to the System can transparently access all the features and functions of the central iPECS as well as connected resources. An iPECS System set to work in Central Mode (CM) controls all remote modules and terminals connected to the System as if they are connected transparently without a local iPECS-MG System.

The voice connection provided locally will not be controlled by a central iPECS directly. Therefore, a VOIP channel should be configured for voice relay between phones in the Local iPECS-MG system and those in the Central iPECS system.

Under normal circumstances, the Central iPECS controls devices in Local MG-System. However, should the WAN connection between the Central system and remote devices fail, the Local MG-System will assume Call Server responsibility for the local devices. The Local MG-System provides local survivability and may provide PSTN back-up service (Fail-over) for internal calls that normally route over the WAN, based on configuration.

Under certain operating conditions, this equipment cannot be relied upon for Emergency calls. Alternative arrangements should be made for access to Emergency services.

Operation

System

Operation of Centralized Network is automatic when configured and defined.

Conditions

1. A VOIB channel in the iPECS-MG System is required.
2. In a Centralized Network, the maximum number of channels available is the maximum number of channels supported by the central iPECS.
3. In a Centralized T-NET, miscellaneous functions (Relay support, MOH, BGM, Alarms and External Page) are not supported.
4. When NAPT or other firewall functions are implemented, packet relay for RTP packets is required; packet relay requires VoIP channels for each simultaneous call desired.
5. The local MG-System will take over operation of registered devices if the Central iPECS does not respond to three consecutive poll attempts over a period of 10-

seconds; once connection to the WAN is re-established, the central iPECS will automatically re-gain control.

6. iPECS can be installed behind a NAT, however Fixed NAT-Port forwarding is required for the host to be reachable using remote devices.

Programming

T-Net Data

1. T-Net Attributes (PGM 330)
2. CM Attributes (PGM 331)
3. Fail-over PSTN Attributes (PGM 333)
4. T-Net Board Attributes (PGM 334)
5. IP Phone T-Net Enable (PGM 335)

Numbering Plan

1. Feature Numbering-T-NET Login/Logout (PGM 113)

Related Features

Hardware

VOIB

iPECS System

3.68.2 Distributed Control Network

Description

In the Distributed Control Network, each System maintains control over the devices registered to it. Networked systems communicate allowing other networked systems access to resources over the network. In addition, other features and functions as detailed in the following sections of this manual are available to users provided by a distributed network environment. The System permits remote access to various resources through registered gateway Modules and Terminals.

In addition, the System will request access to resources of remote systems. A user-dialed number is analyzed and the call routed according to the NET Numbering Table. Should the main path fail to respond, the System re-routes the call employing the designated Alternative Speed Dial route.

The System supports 2 standard protocols (QSIG over ISDN, and H.450 over IP), for basic networking functions. QSIG employs ISDN PRI channels only with support for ESTI standards ETS 300-237/238/256/257/260/261/361/362/363/364.

Operation

Operation of Distributed Networking is automatic when configured and defined.

Conditions

1. To use the networking features, software lock-key installation is required (available from LG-Ericsson Distributor); there are two types of software lock-keys, for QSIG-based networking, and VOIP-based networking.
2. Unified Dialing Plan (UDP) specifies that each Station can have a unique number up to 8 digits in the Networked Systems, depending on the Numbering Plan.

3. An Alternative Speed Dial route can be used to place a call, but is not a Networked call, so Distributed Control Network features are not available.

Programming

- | | |
|-----------------------|---|
| Numbering Plan | 1. System Numbering Plan (PGM 111) |
| Voice Network | 1. Voice Network (PGM 320)
2. Voice Network Numbering Plan (PGM 321) |

Related Features

Hardware

3.68.2.1 Net Call

Description

A Station user can make a call to a Station on another System by dialing just the Station number, as an Intercom call within the same System.

Operation

To configure Net Call Numbering Plan programming:

Press **[PGM] + 111** (Prefix Code) + (1) + Prefix code for Networking Numbering + **[SAVE]**

To perform a Net Call:

1. Lift Handset or press the **[SPEAKER]** button; dial tone will be provided.
2. Dial a Station number on another System, or press the {NET DSS} button for the other System.
3. The Station seizes the network CO Line according to the Net Routing Table, and the System sends a modified digit stream.
4. The called System receives a digit stream sent by the Calling Party, and analyzes it using the net routing table to determine the right destination, sending it to the Called Party (ring signal).
5. When Net Call is established, a network CO Line is used; when Net Call is cleared, the network CO Line is released.
6. The [Network CO] button LED will be extinguished when Net Call is cleared.

Conditions

1. Net call must be used without seizing a CO line.
2. User hears an error tone if there is no idle networking path.
3. In spite of ICM mode, the called party receives a ringing signal for Net Calls.
4. When the System detects a fatal error from the Network, the System sends digit streams to the Network using the Alternate Dial bin (not a Net Call).
5. The CO Call Restriction Timer is also applied to Net Call (PGM180-Index14).

Programming

- | | |
|-----------------------|------------------------------------|
| Numbering Plan | 1. System Numbering Plan (PGM 111) |
|-----------------------|------------------------------------|

CO Line Data	1. CO line Attribute (PGM 160, PGM 161)
Station	1. CO/IP Group Access (PGM 150)
Voice Network	1. Network Basic Attribute(PGM 320) 2. Network Numbering Plan (PGM 321)

Related Features

Hardware

3.68.2.2 Net Transfer

Description

A Station User can transfer any kind of CO call to a Station on other systems by pressing the **[TRANS]** button and dialing a transfer destination Station, as a Call Transfer within the same system. There are two kinds of Transfer, Screened and Unscreened.

NOTE: There are two kinds of standard transfer method in QSIG and H.450; Transfer by Join and Transfer by Rerouting. The main difference is how the connecting path is controlled between the transferring and transfer destination Stations. Transfer by join uses an additional connecting path for transferring the call to another station. Transfer by rerouting, uses a new connecting path to transfer the call and the old connecting path of transferring station is cleared.

Operation

To perform a Screened Transfer to another Station on a different Networked System:

1. Press the **[TRANS]** button at a Station during a CO Line conversation; the CO line is placed on Exclusive Hold.
2. Dial the destination Station number on another System to transfer the call; the Station on the other System receives a ring signal.
3. Announce the call when the call is answered; the CO Call is held in Exclusive Hold while the Station communicates with the destination Station.
4. Hang-up to complete the transfer.

To perform an Unscreened Transfer to another Station on a different Networked Station:

1. Press the **[TRANS]** button at a Station during a CO Line conversation; the CO line is placed on Exclusive Hold.
2. Dial the destination Station number on another System to transfer the call; the Station on the other System receives a ring signal.
3. Hang-up to complete the transfer.

Conditions

1. If both of the transferred and destination Stations are located on the same system, the networking path is cleared; the transfer call will be setup as an Intercom call.
2. The Transfer will be canceled when the transferring Station User presses the flashing **[CO line access code]** or **[TRANS]** button.
3. Net Transfer calls do not recall at the origination Station.
4. The User will hear an error tone if the Networking Path is unavailable.

5. Net transfer is not activated at a Busy station.

Programming

- | | |
|-----------------------|---|
| Numbering Plan | 1. System Numbering Plan (PGM 111) |
| CO Line Data | 1. CO COS (PGM 177) |
| | 2. CO to CO Attribute (PGM 179) |
| Voice Network | 1. Voice Network (PGM 320) |
| | 2. Voice Network Numbering Plan (PGM 321) |

3.68.2.3 Identification Service

Description

When a user makes a Net Call, the System provides the name registered at the Station to the Called Party between Systems.

Operation

If set, CNIP operation is automatic.

Conditions

Programming

- | | |
|-----------------------|-------------------------------------|
| Numbering Plan | 1. System Numbering Plan (PGM 111) |
| Voice Network | 1. Network Basic Attribute(PGM 320) |
| | 2. Network Numbering Plan (PGM 321) |

3.68.2.4 Call Completion

Description

There are two kinds of Call Completion:

- **Completion of Calls to Busy Subscribers (CCBS)** – After calling a User on 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 when that notification is received, the call can be reinitiated to the destination of the other system again.
- **Completion of Calls on No Reply (CCNR)** – After calling a User in another System using basic call and encountering No Reply (No Answer), the caller can set to be notified when the destination becomes idle again. If the caller wants to make a call to the destination when notification is received, the call can be reinitiated to the destination on the other System again.

Operation

To perform a CCBS (Call Back):

1. Dial the Station of the other System.

2. When a busy tone is received, the User can either,
 - (a) Press the [CALLBK] button,
 - (b) Press [TRANS] button and dial {Call Back Register Feature Code},
OR
 - (c) Dial the Call-Back digit set as Tenant Intercom Busy One-Digit service.
3. The call is cleared after the confirmation tone is received.
4. When the busy station returns to Idle; the originator will receive a Call Back ring.
5. When the originator answers the call-back ring, a new call to the Called Station will be activated.

Conditions

1. A stand-alone IP Phone that supports H.450 can activate Call Completion.
2. A station can leave or have only one callback message, and a new request will be left as a message wait indication on the busy Station.
3. A voice message cannot be left even though the VMIM/VSF is installed at a local System.
4. If the Call Originator does not answer the Call Back ring within Net Timer, the call will be cleared.
5. There are two modes: Connection Mode and Connectionless Mode; this can be set at PGM 320-Index 8.

Programming

- | | |
|-----------------------|--|
| Voice Network | <ol style="list-style-type: none">1. Network Basic Attribute (PGM 320)2. Network Numbering Plan (PGM 321) |
| Numbering Plan | <ol style="list-style-type: none">1. System Numbering Plan (PGM 111)2. Feature Numbering Plan (PGM 113) |
| System | <ol style="list-style-type: none">1. Intercom Busy One-Digit Service (PGM 237) |

3.68.2.5 Call Offer

Description

A busy user on one node can be given notification that another call is waiting from another node. It is similar to the Camp-On function.

Operation

To activate Call Offer:

1. Dial a Station number on another System; when busy tone is received, press the [Call Wait] button, or {Call Wait Feature Code}.
2. 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

2. The muted ring is changed to a normal CO ring the current call is disconnected; the offered call then can be answered.

Conditions

1. Call Offer is only applied to a Station that in Busy status.
2. During Conference or Paging, Call Offer is not activated.
3. The System does not support the standard QSIG specification path reservation mode.

Programming

- | | |
|-----------------------|--|
| Voice Network | <ol style="list-style-type: none">1. Network Basic Attribute (PGM 320)2. Network Numbering Plan (PGM 321) |
| Numbering Plan | <ol style="list-style-type: none">1. System Numbering Plan (PGM 111)2. Feature Numbering Plan (PGM 113) |
| System Data | <ol style="list-style-type: none">1. Intercom Busy One-Digit Service (PGM 237) |

3.68.2.6 Net Conference

Description

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 can conference-in a party on any other node. Members of the conference can consist of up to 3 stations on the network.

Operation

To perform Net Conference:

1. Press the [CONF] button during a Net Call conversation.
2. The existing call is placed on hold and an ICM dial tone is provided.
3. Make a Net Call to another Station on another node.
4. Press the [CONF] button when the 3rd party answers.
5. The second call is placed on Hold and an ICM Dial tone is provided.
6. Press the [CONF] button again at the Master Station; all parties will be connected.

To clear a Net Conference:

1. Any Station in the Net Conference can hang-up during the conference.
2. After all parties disconnect, the net conference will be cancelled and the network path will be cleared.

Conditions

1. Multi-line conference time is not applied on networked 3-party conference.

Programming

- | | |
|----------------------|--|
| Voice Network | <ol style="list-style-type: none">1. Network Basic Attribute (PGM 320)2. Network Numbering Plan (PGM 321) |
|----------------------|--|

3.68.2.7 Message Waiting Indication (MWI)

Description

Message Waiting Indication (MWI) is the same as Calling Line Indication (CLI) message wait service. On a normal ISDN call, a Station can leave a Message Wait for an unavailable station on another node (CLI message wait must be enabled). MWI is indicated by the CLI message on the Station LCD panel. Additionally, it can be supported by supplementary service (without CLI) by pressing the {MSG WAIT} button when the Net Call Ring-Back tone is received.

Operation

To register and retrieve MWI:

1. Initiate a Net Call to another station on another node; the caller will hear a Ring-Back tone.
2. While hearing the Ring-Back tone, the caller can either,
 - (a) Press the [MSG/Callbk] button.OR
 - (2) Press the [TRANS] button and dial the {Message Wait Register Feature Code}.
3. MWI is left to the called station, and a flashing [MSG/Callbk] button indicates the message waiting.
4. At the called Station, press the flashing [MSG/Callbk] button; the Calling Station number is displayed.

Conditions

1. MWI only can be registered at the Station hearing a ring back tone.
2. MWI notification is displayed at the Called Station LCD.
3. When the System presents the Call-Back according to MWI data, the CO Line is selected within the Network CO group.

Programming

- | | |
|----------------------|--|
| Voice Network | <ol style="list-style-type: none">1. Network Basic Attribute (PGM 320)2. Network Numbering Plan (PGM 321) |
|----------------------|--|

- | | |
|-----------------------|--|
| Numbering Plan | <ol style="list-style-type: none">1. System Numbering Plan (PGM 111)2. Feature Numbering Plan (PGM 113) |
|-----------------------|--|

3.68.2.8 Net Call Forward (Unconditional/Busy/No-Answer)

Description

A User can remotely forward to another Station immediately over the network.

NOTE: The System supports both Rerouting and Join methods according to Admin. Programming.

Operation

To activate Net Call Forward:

1. Lift handset, or press the [Speaker] button.
2. Dial {Net Call Forward code}.
3. Dial 0 (Remote Forward).

4. Dial the Station Number to forward calls (register).
5. Enter the Station password.
6. Press * or # key.
7. Select Net Call Forward Type (1:Unconditional, 2:Busy, 3:No Answer).
8. Dial the Net Call Forward Station Number on another node.
9. Press the [SAVE] button.

To deactivate Net Call Forward:

1. Lift handset, or press the [Speaker] button.
2. Dial {Net Call Forward code}.
3. Dial 0 (Remote Forward).
4. Dial the Station Number to forward calls (register).
5. Press * or # key.
6. Enter the Station password.
7. Press the [SAVE] button.

Conditions

1. If both the Originating and Forwarded To Stations are located within the same System, the Networking Path will be cleared; the Forwarded Call will be setup as Intercom Call.
2. The System does not check the status of the diverted-to station in DND, CFW or Empty.

Programming

- | | |
|----------------------|--------------------------------------|
| Voice Network | 1. Network Basic Attribute (PGM 320) |
| | 2. Network Numbering Plan (PGM 321) |

3.68.2.9 CO Transit-In

Description

The incoming DID call can reroute to a Net Call destination on another System.

Operation

CO Line programming:

Press [PGM] + 177 (CO COS) + Select the desired Incoming CO Line + Assign COS 01-15 (Toll Exception Table is also applied to Transit Calls).

If configured, CO Transit-In operation is automatic.

Conditions

1. Outside caller hears a busy tone when a Networking Path is not available during transit.
2. A Network CO Line is seized automatically, and the call will be transferred to the Network Destination; the destination will receive ringing with CLI from PX, however the outside User will hear a Ring-Back tone.
3. Both the outside User and the Destination Station will be connected when the destination station answers the ringing.

Programming

- | | |
|----------------------|---|
| CO Line Data | 1. CO COS (PGM 177) |
| Voice Network | 1. Network Basic Attribute (PGM 320)
2. Network Numbering Plan (PGM 321) |

3.68.2.10 CO Transit-Out

Description

This feature enables a user to maximize use of the network and reduce call costs by routing outgoing calls to the nearest appropriate point on the network. The system should provide sufficient digit translation or string analysis options to enable the switch to route the call correctly. A sub-system with no public network connection can gain access to public network using the Main System connection.

Refer to the *Administration Programming Manual* for a full description of programming for this function.

Operation

To use CO Transit-Out, perform the following:

1. A Station of a Sub-System seizes a CO line.
2. The Dummy CO Dial tone (PRI=real dial tone) is provided from the Main System or the Sub-System; according to the CO Dial Send Mode (En-Block or Overlap), the origination of the CO Dial tone is determined.
3. Dial the Telephone number of Public Network User; the Called Station will receive the ringing, and Station placing the call will hear the Ring-Back tone.
4. When Call is answered, the public network telephone and the station of sub-system will be connected.

Conditions

1. To use CO transit-out, the Sub-System User must seize the CO Line.
2. The Original Station COS will receive toll restriction as configured.
3. The Outside Caller will hear a busy tone if a Networking Path is not available during the Transit-Out attempt.

Programming

- | | |
|----------------------|--|
| CO Line Data | 1. CO COS (PGM 177)
2. CO to CO Attribute (PGM 179)
3. CO Group Access (PGM 180) |
| Voice Network | 1. Network Basic Attribute (PGM 320)
2. Network Numbering Plan (PGM 321) |

3.68.2.11 Do-Not-Disturb (DND)

Description

A call to a Station in DND mode can be denied though it is received from a Station on another System; the calling party will receive a busy tone.

Operation

To use DND in a Networked environment:

1. Press the [DND] button to activate DND mode.
2. When a Station on another System calls in to the Station in DND mode, the busy tone will be received, and DND will display on the LCD.

Conditions

1. When a Station is in DND mode, the [Station] button of the DND Station will flash (BLF manager must be activated).

Programming

- | | |
|----------------------|--|
| Voice Network | <ol style="list-style-type: none">1. Network Basic Attribute (PGM 320)2. Network Numbering Plan (PGM 321) |
|----------------------|--|

3.68.2.12 Attendant Call (CAS)

Description

An Attendant Call from any node can be routed to the Centralized Attendant (CAS); the call will be queued when all Centralized Attendants are busy.

Operation

If programmed, Attendant Call operation is automatic.

Conditions

1. A Recall Call is not routed to CAS on the Master System.
2. The CAS DN Number can be converted to Attendant Call code using the Digit Conv. Table.
3. An Attendant Call can be rerouted to CAS according to the Forward type and Destination
4. If all Attendants press [DND] button, CAS will be covered by the Night Attendant Group.
5. If the Forward Destination of Night Attendant group is CAS, the Attendant Call will be routed to CAS.

Programming

- | | |
|----------------------|---|
| Tenant Data | <ol style="list-style-type: none">1. Attendant Attribute (PGM 272-Index3, Index5)2. Night Attendant Group Assign (PGM 275)3. Digit Conv. Table (PGM 251, PGM 252) |
| Voice Network | <ol style="list-style-type: none">1. Network Basic Attribute(PGM 320)2. Network Numbering Plan (PGM 321) |

3.68.2.13 BLF Presentation

Description

Stations on one node are able to program a busy lamp appearance at an extension on another node. The Busy Lamp Field (BLF) key can also be used to call the remote extension. The BLF presentation can be utilized with BLF manager software based on programming settings.

Operation

BLF function is automatic.

Conditions

1. If BLF manager S/W is used for BLF Presentation, the BLF manager should be installed at one system for completely networked systems.
2. The number of Net DSS can be restricted according to the capability of each system.
3. 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.
4. CO BLF is not supported, and also ringing signal does not update a status of that station (ICM / CO / Transfer / CO Recall ring).

Programming

Voice Network 1. Network Basic Attribute (PGM 320-Index6-10)

Application

BLF Manager Software

3.68.2.14 DECT Mobility

Description

When one DECT is registered to more than two networked systems at the same time and the user of DECT moves to another networked system, the incoming call to DECT will be routed to the appropriate networked system automatically.

Operation

DECT Mobility is automatic.

Conditions

1. DECT mobility information is sent through the LAN port of MPB.
2. The physical port number of the DECT should be same as on whole systems.
3. DECT must be registered to more than two systems for this functionality to work (refer to the *iPECS DECT Installation Manual*).

Programming

Voice Network 1. Network Basic Attribute (PGM 320-Index6-10)
 2. Network Numbering Plan Table (PGM321-Index8)

3.69 TRAFFIC ANALYSIS

Description

The System monitors, stores and periodically or upon request outputs various traffic statistics covering System resources. The output from the System can be used to:

- Monitor and evaluate system performance
- Observe usage trends and recommend possible corrective actions,
- Determine possible trunk problems (ex., blocking level too high), and/or
- Recommend System upgrades.

The Traffic report selected by the Attendant will be output only upon request. It is sent to the defined System RS-232 or TCP port.

System resources covered by Traffic Reports are:

- Tenant Call Summary Report (Hourly)
- Call Type Summary Report (Hourly)
- CO Group Summary Report (Hourly)

Operation

System Attendant

To print a Tenant Traffic Report:

1. Press the **[PGM]** button.
2. Dial 021 {Tenant Traffic Report code}.
3. Select Tenant number (0: All, or 1-9)
4. Select Day Information (0: Yesterday, 1: Today)
5. Press the **[SAVE]** button.

To print a Call Type Traffic Report:

1. Press the **[PGM]** button.
2. Dial 022 {Call Type Traffic Report code}.
3. Select Day Information (0: Yesterday, 1: Today)
4. Press the **[SAVE]** button.

To print a CO Group Traffic Report:

1. Press the **[PGM]** button.
2. Dial 023 {CO Group Traffic Report code}.
3. Select Day Information (0: Yesterday, 1: Today)
4. Press the **[SAVE]** button.

Conditions

1. Traffic Reports printing can generate Attendant, Call Summary and CO/IP Summary Traffic Reports.

Programming

- System Data**
1. RS-232 Port Settings (PGM 230)

2. Serial Port Function Selections (PGM 231)

Related Features

SMDR Call Records

Hardware

Printer

3.70 SYSTEM TIME MANAGEMENT

Description

The System provides 9 system time zones, applied to all Time related features. The System time table consist of Weekly, LCR, and Holiday. A user can select the System Time zone, Daylight and Ring mode for each time table.

The following are contained in the Time tables.

- Directory Number
- Incoming CO
- Outgoing CO
- Station Group
- Pilot Hunt Group
- DNT Time / LCR Time
- Tenant

Operation

System Time Management is automatic when programmed.

Conditions

Programming

- | | |
|-------------------|---------------------------------|
| Table Data | 1. System Time Table (PGM 253) |
| | 2. Weekly Time Table (PGM 254) |
| | 3. LCR Time Table (PGM 255) |
| | 4. Holiday Time Table (PGM 256) |

3.71 SYSTEM DATABASE BACKUP TO USB

Description

The system database can be downloaded to USB memory card automatically on MPB based on programmed intervals.

Operation

Conditions

Programming

- System Data**
- 1) DB Auto Download, Weekly (PGM223-Index9)
 - 2).DB Download Time (PGM223-Index10)

Related Features

Hardware

USB interface

3.72 TENANT GROUP

Description

One System can be divided into several systems; each Station and Co line is assigned to a specific Tenant Group. Stations in a group are allowed or denied the ability to place intercom calls to Stations in other groups on a Group-by-Group basis.

Each Tenant Group has an Attendant Group. If a user dials 0 {Attendant Call Feature Code}, the call is routed to the assigned Attendant Group. Additionally, the assigned Attendant member can control the Day/Night Ring mode for Stations in the group switching from Day to Night mode. Each Group is assigned a separate Auto Ring Mode Table for changing the Ring and COS mode automatically during the Day and Night service mode (as applicable).

Operation

System

Operation of Tenant Groups is automatic when programmed.

Conditions

1. A Station denied access will return an Error tone when attempting to make a call to a particular Tenant Group.
2. Tenant does not affect the Station Numbering Plan in the system; all stations in the System must have different Station numbers even if they are assigned to different Tenancy groups.
3. When the Attendant of a Tenant Group sets Day/Night/Timed mode, it will affect only the assigned Tenant Group.

Programming

- Tenant Data** 1. Tenant Data Settings (PGM 270-290)
- Station Data** 1. Station Tenant No. (PGM 131-Index2)
- CO Line Data** 1. CO Tenant No (PGM 160-Index5)

Related Features

Hardware

3.73 UNIVERSAL ANSWER (UA)

Description

UA allows a user to be alerted via an external loud bell and answer defined CO/IP calls by dialing a UA code. While primarily intended for alternate answering, UA will also function in other modes providing UA in all service modes. Calls will appear on the {CO}/{IP} appearance or a {DN} button. An External Control Contact can be assigned to activate an external Loud Bell to alert users of incoming calls.

Digital Phones may program a Flex button as a {UA} button.

Operation

Digital Phone

To assign a Flex button as a {UA} button:

Press [PGM] + {FLEX} + Button Feature Type (1) + {UA Feature Code} + [SAVE]

To access an incoming UA call:

1. Lift the handset or press the [SPEAKER] button.
 2. Dial {UA Feature Code}; the UA call is connected.
- OR
3. Lift the handset or press the [SPEAKER] button.
 4. Press the {UA} button; the UA call is connected.

SLT

To access an incoming UA call:

1. Lift the handset.
2. Dial {UA Feature Code}.

Conditions

1. System will search a CO line for UA from first accessible CO line.

Programming

- | | |
|-----------------------|--------------------------------------|
| Numbering Plan | 1. Universal Answer Code (PGM 113) |
| CO Line Data | 1. Universal Answer (PGM 166-Index7) |

Related Features

Loud Bell Control (LBC)

Hardware

3.74 VMIB INTEGRATED AUTO ATTD/VOICE MAIL

3.74.1 VMIB

Description

The Voice Message Interface Board (VMIB) includes processing and memory for the System integrated Auto-Attendant, Voice Mail and System announcement applications. The memory is employed to store Auto-Attendant announcements, voice mail, greetings and messages, and various system prompts. The System prompts (time, date, etc.) are employed by the Auto-Attendant and Voice Mail applications as well as other system features.

3.74.2 VMIB-Auto Attendant

Description

When a call comes in to the System through an ICM Call, a DID or a DISA line, the call may be routed to one of 70 User-recorded VMIB Announcements. An announcement is assigned as a Station Group announcement or as an Auto-Attendant announcement with Caller Controlled Routing (CCR). Station Group announcements are played when a call is routed to the group based on definitions in Station Group Attributes.

For an Auto-Attendant Announcement the System will play the announcement and monitor for digits from the connected external party. A CCR Table defines a dialed digit (0-9, #, and *) to a designated route; each individual digit corresponds with a route:

- Station
- Station Group
- CO Group Access Code
- Internal Page Zone
- Voice Mail Access Code
- CCR Access Code + VMIB Announcement
- CCR Access and Drop Code + VMIB Announcement
- Conference Room
- Net number

In addition, the System will monitor digits for a Station number; if the User dials a Station number, the Auto-Attendant will complete an unsupervised call transfer to the station.

Operation

System Attendant

To record an Auto-Attendant Announcement:

1. Press the **[PGM]** button.
2. Dial '062', the Message Record code.
3. Dial the VMIB slot number.
4. Dial the Announcement number (01-70).

5. Enter the Language number (1-3, if supported); the announcement "Press the # key to record," will be presented. If there is already a recorded message in the number dialed, the corresponding message will be played.
6. Dial '#'.
7. After the beep-tone, record message.
8. Press the **[SAVE]** button to stop recording and save the message.

To delete a recording:

1. Press the **[PGM]** button.
2. Dial 062 {Message Record code}.
3. Dial the VMIB slot number.
4. Dial the Announcement number (01-70).
5. Enter the Language number (1-3, if supported); the announcement "Press the # key to record," will be heard. If there is already a recorded message in the number dialed, the corresponding message will be played.
6. Press the **[SPEED]** button during playback to erase message

System

Operation of CCR Audio Text Tables and Auto-Attendant are automatic.

Conditions

1. There are no individual time limits on Auto-Attendant announcements.
2. The external caller may experience a Ring-Back tone before playback of a VMIB announcement.
3. The Attendant Station must "Save" a recording before returning to the on-hook state, otherwise the existing recording is used and new recording will be lost.
4. To record or delete an Auto-Attendant message, all of the VMIB channels must be in the idle state.
5. The external caller may dial at any time during an Auto Attendant announcement and must dial prior to the expiration of the CCR Analysis timer.
6. If the external caller dials an invalid selection or station, the System will present the Invalid Entry prompt and initiate a re-entry according to the DISA Retry Counter.
7. If the external caller dials more than a single digit, the call is routed based on the System Numbering Plan.
8. Calls answered by an Auto-Attendant (CCR) Announcement are interactive DISA calls and are subject to conditions of a DISA call.
9. A CCR Announcement may be programmed to disconnect the call after playing.
10. The Auto-Attendant Announcement feature is supported for DISA and DID calls.
11. To allow back-tracking in call routing, assign one of the CCR destinations of a current step as the CCR announcement number for the previous step.
12. The remote Caller's voice mailbox access can be supported by assigning the {VMIB Access Code} to a CCR destination.

Programming

Tables

1. CCR Tables (PGM 260)

Related Features

Hardware

VMIB

3.74.3 VMIB Voice Mail

3.74.3.1 Message Storage

Description

When a station activates Call Forward to the {VMIB Access Feature Code}, the call is transferred to a VMIB mailbox or a transferred call recalls to the VMIB, the call is handled by the System's Voice Mail application. The caller can connect with the called Station User Greeting followed by a beep tone.

A remote Caller can record a message and hang-up or dial '*' for further options. When disconnected, the VM application will store the message in the Called User Voice Mailbox and activates the Message Waiting Indication (MWI) at the User Station. If VM back-up is assigned at the back-up station, Phontage or UCS Client is also notified.

Operation

Remote Caller

To leave a voice message:

1. After receiving the Greeting and beep on an attempted call, record the desired message.
2. Hang up to quit recording or dial * for further options.

Conditions

1. Two timers are provided to control voice message length,
 - VMIB-Message Minimum Record Timer: establishes the minimum voice message length; voice messages shorter than this timer are not stored.
 - VMIB-Message Maximum Record Timer: establishes the maximum voice message length; when the VMIB-Message Maximum Record Timer expires while a voice message is being recorded, a confirmation tone is heard and the message is saved for the destination station.
2. If all VMIB channels are in use, a Ring-Back tone is provided until a VMIB channel is available.
3. All active Stations including SLTs can leave and receive voice messages.
4. Individual User Greetings and Voice Mails are protected from loss of AC power.

Programming

System Data

1. VMIB-Message Minimum Record Timer (PGM 220-Index7)
2. VMIB-Message Maximum Record Timer (PGM 220-Index8)

Numbering Plan

1. VMIB Access (PGM 113)

Station Data 1. Station VMIB Attributes (PGM 145)

Related Features

Hardware

VMIB

3.74.3.2 Message Retrieval

Description

A user can access their Mailbox locally from a Digital Phone by dialing the {VMIB Access Feature Code}, by pressing the **[MSG/CALLBK]** button, or by pressing a pre-assigned {VMAILBOX} Flex button when Off-Hook receiving Intercom dial tone.

Prompts are presented to guide the User in the Voice Mailbox operation. The User must enter a Mailbox number (Station number), and a corresponding password in response to the Request for Mailbox number (*"Please enter your Mailbox number."*) and Request for Password (*"Please enter your password code."*) prompts. Additional prompts and mailbox operation is described in Operation, or refer to your **Phone** or **Voice Mail User Guide**.

Operation

Digital Phone

To assign a {VMAILBOX} Flex button:

Press **[PGM] + {FLEX} + Button Feature Type(1) + {VMIB Access Feature Code} + [SAVE]**

To retrieve Voice Mail locally:

1. Lift the handset or press the **[SPEAKER]** button
2. Press **[MSG/CALLBK]** button.
3. Dial 2 to select VMIB Messages; the Mail Box & Password prompts will be played.
4. Enter the Mailbox (Station number) and corresponding password; if entry is valid, the Number of Messages prompt will be presented.
5. Dial desired option code,
 - 1. Play New Messages
 - 2. Play Saved Messages
 - 8. Set Greeting or Password
 - #. Disconnect
 - 0. Operator
 - 9. Replay Prompt
6. Following selection, additional prompts will be presented.
7. At completion of session, hang-up to return to idle.

OR

1. Lift the handset or press the [speaker] button
2. Press {VMAILBOX} button.
3. Enter the Mailbox password corresponding to the Station; if entry is valid, the Number of Messages prompt will be presented.

4. Dial desired option code (refer to Step 5 above).
5. Following selection, additional prompts will be presented.
6. At completion of session, hang-up to return to idle.

To attach a memo to the current voice message:

1. After message playback, dial 7 during or after message option prompt.
2. Following the beep, record the desired memo.
3. Press * key to stop recording and store the memo.
4. During or after the New/Old Option Prompt, dial 4 to forward the message and memo.

SLT

To retrieve Voice Mail locally:

1. Lift the handset.
2. Dial the {VMIB Access Feature Code}; the Mailbox & Password will be presented sequentially.
3. Enter the Mailbox number (Station number) and corresponding password; if entry is valid, the Number of Messages prompt will be presented.
4. Dial desired option code,
 - 1. Play New Messages
 - 2. Play Saved Messages
 - 8. Set Greeting or Password
 - #. Disconnect
 - 0. Operator
 - 9. Replay Prompt
5. Following selection, additional prompts will be presented.
6. At completion of session, hang-up to return to idle.

To attach a memo to the current voice message:

1. After Message Playback, dial 7 during or after Message Option Prompt.
2. Following the beep, record the desired memo.
3. Dial * to stop recording and store the memo.
4. During or after the New/Old Option Prompt, dial 4 to forward the message and memo.

Conditions

1. If no new/old messages are available, pressing 1 or 2, is an invalid operation and the User receives the "Invalid Entry" prompt or "No Message" prompt.
2. If the dialed number is not recognized, the "Invalid Entry" prompt is played; after the second invalid entry, the User is disconnected.
3. The User may dial digits at any time during Voice Mail playback, System Prompt or silence; the user must dial a digit in response to a System Prompt within the CCR Analysis timer or the System will return an error tone and disconnect the call.
4. Messages can be retrieved in either a FIFO (First in First out) or LIFO (Last in First out).

Programming

Station

1. Station VMIB Attributes (PGM 145)

Numbering Plan 1. VMIB Access (PGM 113)

Related Features

Message Retrieval Options
Remote Message Retrieval
Multiple Voice Mailbox Support

Hardware

VMIB

3.74.3.3 Remote Message Retrieval

Description

The System permits remote Users access to their Mailbox. After accessing the VMIB Voice Mail, operation follows the local procedures.

Operation

Remote Caller

To access Voice Mailbox from a remote location:

1. Lift the handset.
 2. Dial the telephone number of a DISA assigned CO Line assigned for answer by a VMIB Auto-Attendant.
- OR
3. Dial a Station Group number assigned for answer by a VIMB Auto-Attendant.
 4. Upon answer, dial {VIMB Access Feature Code}; the Request for Mail Box Number prompt will be presented.
 5. Follow local access procedures.

Conditions

1. The conditions associated with Message Retrieval and Message Retrieval Options apply.
2. The conditions associated with DISA/DID apply.

Programming

Station 1. Station VMIB Attributes (PGM 145)
Numbering Plan 1. Feature Numbering Plan, VMIB Access (PGM 113)

Related Features

Message Retrieval Options
VMIB-Auto Attendant
Message Retrieval

Hardware

VMIB

3.74.3.4 Message Retrieval Options

Description

The user may dial the digit 9 to receive the VM Long Options prompt while in the Voice Mailbox, including during or after a Voice Message or System Prompt except when an option has been selected that requires user dialing. The VM Long Options prompt is:

"To play New Messages, press 1. To play Saved Messages, press 2. To set Station Forwarding, press 7 (available only for remote access). To set Greeting or Password, press 8. To Disconnect, press #. Press 0 for the Operator. Press 9 to hear this message again."

The VMIB Voice Mail will respond to incoming digits as shown in the following table.

VMIB Voice Mail Input

Digit	Function	Prompt
1	Play New Msg	
2	Play Saved Msg	
7	Set Cancel/Fwd	
8	Mail Box Setting	Mailbox Settings (greeting/password)
9	VM Long Options	VM Long Options
#	Drop	Goodbye
0	Attendant Group Call	Call to System Attendant

When the user responds by dialing 1, the first New Message is played. At the end of message playback, the New Message option prompt is presented:

"To Replay Message, press 1. To listen to the next Message, press 2. To Delete Message, press 3. To Forward Message, press 4. To Call the Sender, press 5. To Skip Message, press 6. To return to Main Menu, press 9."

This process is repeated until the last new message is played and the No Message prompt ("No Messages") is played.

When the user dials 2 in response to the Number of Messages prompt, the Oldest Saved Message is played. At the end of Message Playback, the Old Message option prompt is presented:

"To Replay Message, press 1. To listen to the next Message, press 2. To Delete Message, press 3. To Forward Message, press 4. To Call the Sender, press 5. To return to Main menu, press 9."

This process is repeated until the last new message is played and the No Message prompt ("No Messages") is played.

In addition to the options indicated in the prompt, a user can dial 7 to record a memo, and attach it to the current voice mail. The current voice mail and memo can then be forwarded to other users.

When the user dials 9 in response to the Number of Messages prompt, or during or at the end of a message the VM Long Options prompt is played.

Operation

Digital Phone

To access a Message Retrieval option:

1. At any time after the Number of Messages prompt, dial a Message Retrieval Option digit; the System initiates the selection providing any subsequent prompts.

SLT

To access a Message Retrieval option:

1. At any time after the Number of Messages prompt, dial a Message Retrieval Option digit; the System initiates the selection providing any subsequent prompts.

Conditions

1. The User must begin dialing within the CCR Analysis timer in response to a system prompt; if the timer expires, the User will receive an error tone and the System will disconnect the call.
2. When the call sender option results in an external call, dialing restrictions will be applied based on the Station COS.
3. If the User remains Off-Hook after a call placed through the Voice Mail is complete, the User will be returned to the previous place in the Voice Mailbox. If the user hangs up, the VMIB will Recall the User, and will present the Request Mailbox Number prompt.

Programming

- | | |
|-----------------------|---|
| Station | 1. Station VMIB Attributes (PGM 145) |
| Numbering Plan | 1. Feature Numbering Plan (VMIB Access) (PGM 113) |

Related Features

Message Retrieval
Remote Message Retrieval
Voice Mailbox Settings
Class of Service

Hardware

VMIB

3.74.3.5 E-Mail Notification

Description

The System stores VMIB voice messages and sends an e-mail to the e-mail address associated with the station as notification of the new e-mail. The voice message is attached to the e-mail as a .wav file.

Operation

System

If configured, System automatically sends e-mail to notify User of new Voice Messages.

Conditions

1. Voice Messages are stored in the VMIB as well as being attached to the e-mail. The Voice Message must be deleted from the VMIB even if the e-mail is deleted.
2. The e-mail will be sent to the address assigned for the Station with the Sender address defined for the VMIB.

NOTE: The Sender address must be defined, as many e-mail servers will reject anonymous e-mails.

3. The e-mail address for the VMIB and the station is defined under the Web Admin.
4. The Voice message can be attached to the e-mail notification as a .wav file, if the Attach Message option is enabled. If disabled, the notification e-mail does not include an attached .wav file.

Programming

Station Data

1. Station VMIB Attribute (PGM145)
2. SMTP Mail Server IP Address (Web Admin.)
 - User Mail Address
 - SMTP Mail Server ID
 - SMTP Mail Server Password
 - Attach Message Option

Related Features

VMIB Integrated Auto Attd/Voice Mail

Hardware

VMIB

3.74.3.6 Voice Mailbox Settings

Description

The User can program personal Mailbox settings including a security password and a greeting. When a user presses 8 while retrieving messages, the Mailbox Setting prompt, (*"To edit your greeting, press 1. To edit your password, press 2. To return to Main Menu, press 9."*) is played.

Operation

To program Mailbox settings while using the Voice Mailbox:

1. Press 8 (Mailbox settings), the Mailbox Setting prompt is presented.

To modify the Password:

1. Dial 2, the Password Entry prompt will be presented ("Please enter your new password and press # when finished.").
2. Enter desired new password and then press the # key; the Reenter Password prompt will be presented ("Please re-enter your password to confirm and press # when finished.").
3. Enter the new password again, then press the # key; the Password Confirmation prompt is presented ("Your password is saved.").

To modify the Greeting:

1. Dial 1, the Greeting Option prompt will be presented ("To listen to your current Greeting, press 5. To Record a new Greeting, press 7. To return to the Main Menu, press 9.").
2. Dial 5, to hear your Greeting.
OR
3. Dial 7, the Record Greeting prompt is presented ("At the tone, record your new greeting, press # when done.").
4. After the beep, record Greeting speaking in a normal voice.
5. When finished, press the # key, the Greeting Confirmation prompt is presented ("Your greeting is saved.").

To modify Mailbox Settings:

1. Dial 9, the Mail Box Setting prompt is presented ("To edit your greeting, press one, to edit you password, press two. To return to main menu, press nine").

Conditions

1. If the User is external (remote), the User must begin dialing within the CCR Analysis time, if not the call is released.
2. If the dialed number is not recognized, the Invalid Entry prompt is played.
3. The User must assign a password (Authentication Code= up to 12 digits) before access to the mailbox will be allowed.

NOTE: Greeting does not need to be recorded.

Programming

Related Features

Message Storage
Message Retrieval
Remote Message Retrieval
Message Retrieval Options

Hardware

VMIB

3.74.3.7 Call Forward from VMIB

Description

External Users can activate or deactivate Call Forward for their station. Pressing 7 while retrieving messages will return the Mailbox Set Forward prompt.

Operation

To activate Call Forward while using the VM:

1. Press 7 for Mailbox Set Forward, the corresponding prompt is received.

2. Dial 1, the Password Entry prompt is presented ("Please enter the number to forward to ...").
3. Dial the desired Station Number as follows:
 - To Forward to another Station, dial the Station number and press the * key.
 - To Forward Calls Off-Net, dial CO seize number and dial phone number press the * key.

To deactivate Call Forward:

1. Press 7 for Mailbox Set Forward, the corresponding prompt is received.
2. Dial 2, the Station Forwarding Cancel prompt is presented.

To return to the Main menu

1. Dial 9, the Mailbox Settings prompt.

Conditions

1. If the User is external (remote), the User must begin dialing within the CCR Analysis time and dial subsequent digits within the VMIB Inter-Digit Timer; if not the call is released.
2. Mailbox Set Forward is only available for External Users.

Programming

- | | |
|-----------------------|--|
| Station Data | 1. Station VMIB Attributes (PGM 145) |
| Numbering Plan | 1. Feature Numbering Plan, VMIB Access (PGM 113) |

Related Features

Message Storage
Message Retrieval
Remote Message Retrieval
Message Retrieval Options

Hardware

VMIB

3.74.3.8 Delete All VM Messages

Description

All User VM Messages can be deleted using the **{Delete All VM Delete}** feature code.

Operation

To delete all a user's VM Messages:

1. Dial {Delete All VM Delete} feature code.
2. Enter User's Authorization code.
3. Dial 1 (Delete All).

Conditions

Programming

- System Data** 1. Delete All VM Message Code (PGM 113)

Related Features

Hardware

3.74.3.9 Direct VM Transfer

Description

Internal/External Calls can be directly transferred to a designated Station Voice Mail Box.

Operation

To activate Direct VM Transfer:

1. While on a CO/ICM Call.
2. Press [Trans] button and dial the **{Direct VM Transfer}** feature code.
3. Dial desired Station Number.
4. Go on-hook; VM Transfer will be completed.

Conditions

1. VMIB Access option must be ON.
2. If VMIB channel is all used, recalling is served to transferring station.

Programming

- System Data** 1. Direct VM Transfer Code (PGM 113)

Related Features

Hardware

3.74.3.10 Phontage Message Backup and Delete

Description

iPECS-MG Softphone (Phontage or UCS Clients) can be notified of voice mail for a Registered Station on MIB boards. Softphone users can check their own voice mail and hear recorded voice mail of registered stations. Also, Softphone users can backup voice mail to their PC, and then can manage their voice mails. User can delete voice mail on VMIB boards, according to their assigned authority.

Operation

Message Backup and Delete function is automatic (refer to Softphone User Guide for user operation).

Conditions

1. Phontage backup will be operated when messages are saved at VMIB boards.

2. If Backup Message Delete is executed, voice mails at target station will be deleted in VMIB boards.

Programming

- Station Data**
1. Record Message Backup Phontage Number (PGM 145-Index6)
 2. Backup Message Delete (PGM145-Index7)

Related Features

Hardware

3.74.4 System Voice Memo

Description

This feature provides several general Voice Memos to provide the System Time and Date as well as Station number, and settings over the Digital Phone speaker or the handset for SLTs.

Operation

Digital Phone

To hear Date & Time Prompt:

1. Dial {System Voice Memo Announcing Date/Time code}.
2. Announcement for Date & Time is heard, "Date is May 2nd. Time is xx:xx pm."

To hear Station Number Prompt:

1. Dial {System Voice Memo Announcing Station Number code}.
2. The Station number announcement for Station is heard, "This is station 150."

To hear Station Settings:

1. Dial {System Voice Memo Announcing Station Configuration code}.
2. Status for the Station is reported. Items reported are as follows:
 - Station IP Address
 - Station Mac Address
 - Station ICM Mode (Handsfree/Tone/Privacy)
 - Listed message x (x: number of all messages waiting)
 - Wake-Up Time (hh:mm)
 - Do not disturb
 - COS x

SLT

To hear Date & Time Prompt:

1. Lift the handset.
2. Dial {System Voice Memo Announcing Date/Time code}.
3. Announcement for Time is heard, "Date is May 2nd. Time is xx:xx pm."

To hear Station Number Prompt:

1. Lift the handset.
2. Dial {System Voice Memo Announcing Station Number code}.
3. Announcement for Station is heard, "This is station 150."

To hear Station Settings:

1. Lift the handset.
2. Dial {System Voice Memo Announcing Station Configuration code}.
3. Status for Station is reported. Items that will be reported are as follows:
 - Station number
 - Station IP Address
 - Station Mac Address
 - Station ICM Mode Handsfree/Tone/Privacy)
 - Listed message x (the number of all messages waiting)
 - Wake-Up Time (hh:mm)
 - Do Not Disturb
 - Queued CO/IP xx
 - Locked (temporary COS change)
 - COS x

Conditions

1. For Station Status, items from "Listed message x" to "COS x" will be not be announced if not active.

Programming

- | | |
|-----------------------|--|
| Station Data | 1. Station VMIB Attributes (PGM 145) |
| Numbering Plan | 1. Feature Numbering Plan, VMIB Access (PGM 113) |

Related Features

Hardware

3.75 WAKE-UP ALARM

Description

This feature allows a User or Attendant to set a Wake-Up time or desired time to be alerted. When the time is reached, the System will signal with an audible and visual notification.

Operation

System Attendant

To register a Wake-Up Alarm:

1. Press the [PGM] button.
2. Dial 045 {Attendant Station Program code}.
3. Dial the desired Station range; for a single station, enter the same Station Number twice.

4. Enter Time for Alarm (hh:mm).
5. For a Daily reminder (Repeating Alarm), dial #.
6. Press **[SAVE]** button.

To erase Wake-Up Alarm:

1. Press the **[PGM]** button.
2. Dial 046 {Attendant Station Program code}.
3. Dial the desired Station range; for a single station, enter the same station number twice.
4. Press **[SAVE]** button.

Digital Phone

To register Wake-Up Alarm:

1. Press the **[PGM]** button.
2. Dial 13 {Set Wake-up code}.
3. Dial Time for Alarm (hh:mm).
4. For a Daily (Repeating Alarm), dial #.
5. Press **[SAVE]** button.

To stop the Alarm when alerting:

1. Lift the handset or press **[SPEAKER]**.

To erase Wake-Up:

1. Press the **[PGM]** button.
2. Dial 14 {Erase Wake-up code}.
3. Press **[SAVE]** button.

SLT

To register Wake-Up:

1. Lift the handset.
2. Dial the {Program Mode Access code}; confirmation tone is heard.
3. Dial 13 {Set Wake-up code}.
4. Dial Time for Alarm (hh:mm).
5. For a Daily (Repeating Alarm), dial #.
6. Press Hook-flash, and conformation tone is provided.

To stop the Alarm when alerting:

1. Lift the handset.

To erase Wake-Up Alarm:

1. Lift the handset.
2. Dial the {Program Mode Access} code; confirmation tone is heard.
3. Dial 14 {Erase Wake-up code}.
4. Press Hook-flash, and a conformation tone is provided.

Conditions

1. When receiving a Wake-Up signal, lifting the handset will return Wake-Up Answer Tone.
2. The Wake-Up alarm ring signal follows the Ring Table.
3. If the User does not answer the Wake-Up Alarm ring, it is repeated according to the Wake-Up Retry Counter with the interval of Wake-Up Retry Timer.

NOTE: The value 'N' of the Wake-Up Retry Counter means that 'N' repetition of Wake-Up Retry can occur after the first Wake-Up Alarm ring occurs. The wake-up alarm ring will recur according to the Wake-up Retry Time and continues during Wake-Up Alarm Ring Timer. If no action is taken by the User until the end of the Wake-Up Retry Counter, the ring signal is presented at the Attendant Station with a display designating the Station number that did not respond to Wake-Up Alarm.

4. Time (hh:mm) must be entered in the Military format (24-hour).
5. The Daily Alarm will reset and repeat each day until erased (cancelled), however, One-Time Alarm will reset and cancel automatically.
6. When registering Wake-up Time, if user dials 1, then the One-Time Alarm will be set or if User dials 2, then the Daily Alarm will be set.

Programming

Station Data	<ol style="list-style-type: none">1. Wake-Up Time (PGM 134-Index8)2. Repeat Wake-up (PGM 134-Index9)
Table Data	<ol style="list-style-type: none">1. Wake Up Answer Tone (Web Admin. PGM 290-Index65)2. Wake-Up Indication Ring (Web Admin. PGM 265-Index10)
Tenant Data	<ol style="list-style-type: none">1. Wake-Up Retry Count (PGM 280-Index 5)2. Wake-Up Retry Time (PGM 280-Index 6)

Related Features

Hardware

4. INTERCOM

4.1 DIRECT STATION SELECT/BUSY LAMP FIELD (DSS/BLF)

Description

When a Flex button on a Digital Phone or DSS Console is assigned as a {DSS} button, it also serves as a Busy Lamp Field (BLF). The LED indicates the status of the associated Station or System facility.

{DSS} button indicates the following conditions:

- In use at Station
- In use by another Station
- DND
- Receive Incoming Call
- Hold
- Call Forward
- Conference at Station
- Conference by another Station
- Conference Initiator
- Lifted handset
- Leave pre-selected (custom) message

Operation

Digital Phone

To assign a {DSS} button to a Flex button on a Digital Phone or a DSS Console:

Press [PGM] + {FLEX} + Button Feature Type(1) + {Station Number} + [SAVE].

DSS/BLF operation is automatic for assigned Flex buttons.

Conditions

1. A Station receiving an ICM call is considered Busy, and associated station LEDs will flash at all other stations.
2. A Station receiving an ICM call will receive visual LED Flex button indication (flashing) associated with the Calling Station.
3. The LED Flash Rate can be adjusted by Admin. Programming.

Programming

- | | |
|---------------------|---|
| Station Data | 1. Station Flexible Button Assignment (PGM 126) |
| System Data | 1. LED Color/Flash Rate (PGM 234) |

Related Features

Intercom Call (ICM Call)

Hardware

4.2 INTERCOM CALL (ICM CALL)

Description

A non-blocking ICM is available to all Stations in the System. Users may place an Intercom call to other Stations in the System by dialing applicable digits as defined in the System Numbering Plan.

Operation

Digital Phone

To place an Intercom call:

1. Lift the handset or press the **[SPEAKER]** button to receive the ICM dial tone.
 2. Dial Station number or press the **{DSS}** button.
 3. For Ring-Back tone, await answer.
- OR
4. For Intercom splash-tone, speak and await answer.

SLT

To place an intercom call:

1. Lift the handset to receive ICM dial tone.
 2. Dial Station number.
 3. For Ring-Back tone, await answer
- OR
4. For Intercom splash-tone, speak and await answer.

Conditions

1. Intercom Dial tone will time-out if action is not taken within Dial-Tone Time or if the time between digits exceeds the Inter-digit Timer (Error tone is presented when time-out occurs).
2. ICM Dial tone is removed after dialing the first digit.
3. If the Called Station is Busy, Intercom Busy tone is provided for the Busy Tone time (7 seconds), then the Error tone is sent by the System; the caller may disconnect or activate a feature such as Message Wait/Callback prior to the time-out.
4. For Digital Phone users, consecutive Intercom calls can be placed without the need to regain ICM dial tone between calls (no need to hang-up); the user simply presses another **{DSS}** button.
5. An Intercom call to a Station in the Tone or Voice Announce mode (T or P Intercom Signaling Mode) is not considered answered unless the Called User lifts the handset or presses the **[SPEAKER]** button (goes off-hook).

Programming

- | | |
|-----------------------|---|
| Numbering Data | 1. Flexible Station Number (PGM 112)
2. Feature Numbering Plan (PGM 113) |
| System Data | 1. LED Color/Flash Rate (PGM 234)
2. Inter Digit Timer (PGM 222-Index3) |

Related Features

Intercom Signaling Mode
Speakerphone

Hardware

4.3 INTERCOM CALL HOLD

Description

While on an active ICM Call, the Station User can place an ICM Call on Hold. The Held Station will receive the assigned Music-on-Hold (MOH). The Held call is placed on Exclusive Hold and will Recall the after expiration of the Hold Recall Timer.

Operation

Digital Phone

To place an active ICM call on Hold:

1. Press the [HOLD] button, the [DN] button LED will flash at the designated Hold rate.

To retrieve the held ICM call:

1. Press the {DN} button associated with the Held station, then the ICM call is connected again.

SLT

To place an active ICM Call on Hold:

1. Press the Hook-switch during a conversation.
2. Dial {System Hold Code}; confirmation tone is heard.
3. Replace the handset.

To retrieve the held ICM call:

1. Lift the handset; station is connected with the Held party.

Conditions

1. After placing a Call on Hold, the Station returns to idle and the User can make another call.
2. If the User is in an Off-Hook Status when placing a call on Hold, the dial tone is heard.

Programming

- Numbering Data** 1. Feature Numbering Plan (PGM 113)

Related Features

Music-On-Hold (MOH)
Intercom Call (ICM Call)
Hold & Hold Recall

Hardware

4.4 INTERCOM CALLER CONTROLLED ICM SIGNALING

Description

A User can change the Signaling mode of an ICM call from Tone ring to Voice announce.

Operation

To change the ICM Signaling mode:

1. Dial {Force HF Calling Code}.
2. Place intercom call,
OR
1. Place intercom call.
2. Dial '#'

Conditions

1. The ICM Signal mode cannot be changed if the Called Station number is {MADN}.
2. If the Signaling mode is changed, the Call is not subject to Call Forward, No Answer.
3. The Signaling mode for a specific Intercom call can only be changed once and cannot be changed back to the original Signaling mode.
4. Changing the Signaling mode does not affect privacy at the Called Station.

Programming

- | | |
|-----------------------|--|
| Station Data | 1. Forced Hands free Access (PGM 132–Index1) |
| Numbering Plan | 1. Forced Hands free Call Code (PGM 113) |

Related Features

Intercom Signaling Mode

Hardware

4.5 INTERCOM LOCK-OUT

Description

If the User takes no action after going Off-Hook for the Dial Tone Timer, or fails to dial an additional digit within the Inter-Digit Timer, the Station will receive an Error Tone for 30 seconds and phone will be placed out-of-service (locked-out). The LED of the associated {DSS} buttons as well as the Station [ICM] button will flutter (flash) rapidly to indicate the out-of-service state.

For Digital Phone users, if the [SPEAKER] is used, the Station will receive the Error tone for 30 seconds and then automatically return to idle.

Operation

System

Operation of Intercom Lockout is automatic based on the Dial Tone & Inter-Digit Timers.

Conditions

1. If the Station is assigned with the Howler Tone, the Error tone is presented at the expiration of the Error Tone Timer. The Howler Tone is presented for the Howling Tone Timer expiration, and the Station is placed out-of-service (lockout and silence).

Programming

- | | |
|---------------------|--|
| Station Data | 1. Howling Tone (PGM 121-Index7) |
| Tenant Data | 1. Dial Tone (Web Admin. PGM 290-Index1-2) |

Related Features

Hardware

4.6 INTERCOM STEP CALL

Description

When the Busy Tone is received on a dialed Intercom call, the User may place a call to another Station by dialing the last digit of the Station number. The System replaces the last digit of the previously dialed Busy Station with the dialed digit and places an Intercom call to the new Station number.

Operation

Digital Phone

To activate Step call:

1. While receiving Busy notification on a dialed ICM call, dial a digit other than the last digit of the busy Station intercom number.
2. The System will attempt an ICM call to the new Station.

Conditions

1. If the user dials the last digit of the Busy Station, Camp-On will be activated.
2. After receiving a Busy tone, if the user takes no action for the Busy Tone Timer (7sec.), the System will start the Intercom Lockout procedure.
3. If programmed, Step call is supported.

Programming

- | | |
|--------------------|---|
| Tenant Data | 1. Internal Busy Tone (Web Admin. PGM 290-Index8) |
| | 2. Step Call (PGM 237-Index1) |

Related Features

Hardware

4.7 INTERCOM TRANSFER

Description

Users can Transfer an active Intercom call to other Stations in the System. Intercom calls can be Transferred after announcing the call (Screened) or without announcing the call (Unscreened).

The Intercom Station is placed on Exclusive Hold, and the Transfer Recall Timer is initiated; if the timer expires before the Intercom call is answered, the call will Recall at the Transferring Station until answered or abandoned.

Operation

Digital Phone

To perform a Screened ICM transfer, while on an ICM call:

1. Press **[TRANS]** button.
 2. Dial the Station to receive the call.
 3. When answered or on Splash tone, announce the call.
 4. Hang-up, and return to Idle.
- OR
1. Press the **{DSS/BLF}** button for the desired Station.
 2. When answered or on Splash tone, announce the call.
 3. Hang-up, and return to Idle.

While on an Intercom call, to perform an Unscreened call transfer:

1. Press the **[TRANS]** button.
 2. Dial the Station to receive the call.
 3. Hang-up, and return to Idle.
- OR
1. Press the **{DSS/BLF}** button for the desired Station.
 2. Hang-up, and return to Idle.

SLT

To perform a Screened transfer of an active Intercom call:

1. Press the Hook-switch.
2. Dial the Station to receive the call.
3. When answered, or on Splash tone, announce the call.
4. Hang-up, and return to Idle.

While on an Intercom call, to perform an Unscreened call transfer:

1. Press the Hook-switch.
2. Dial the Station to receive the call.
3. Hang-up, and return to Idle.

Conditions

1. The **[ICM]** button provides an appearance for the transferred station; the LED indicates status and pressing the button connects to the station.

2. If the receiving Station is busy, the Transferring Station may Camp-On the call at the Busy Station (refer to Camp-On).
3. A Station in DND or out-of-service cannot receive a Transfer, and such attempts will result in an Error tone.

Programming

Related Features

- Hold Recall
- Do Not Disturb (DND)
- Call Transfer

Hardware

4.8 INTRUSION

Description

A Station User can Intrude upon an active Station conversation. If assigned, when the Station Intrudes an Intrusion Tone is provided, and a conference is established among the Intrusion Station, and the Station and CO/IP party.

Operation

To perform an Intrusion:

1. Lift Handset or press [Speaker], and place call in the normal manner.
2. When Busy tone is heard, press **[TRANS]** button and dial the {Intrude Register Feature Code}.
3. When finished with Intrusion call, Hang-Up and return to Idle.

Conditions

1. A Station must be enabled for Intrusion feature to work.
2. If the Called Station has the Auto Privacy option set to ON then the call cannot be Intruded upon.

Programming

- | | |
|---------------------|--------------------------------------|
| Station Data | 1. Intrusion Access (PGM 132-Index5) |
| | 2. Auto Privacy (PGM 134-Index11) |

Related Features

- One Digit Service

Hardware

4.9 MESSAGE WAIT/CALL BACK

4.9.1 Short Message Service (SMS)

Description

The Short Message Service (SMS) provides the ability to send and receive text messages to and from Digital Phones equipped with a display including the Phontage, UCS Client, and the WLAN Phone. The text can be comprised of words, numbers, or an alphanumeric combination, and each short message can be up to 100 characters in length when a Latin-based alphabet is used.

Operation

To send SMS:

1. Press the **[PGM]** button.
2. Dial 43.
3. Enter Station Number; for several stations, Press **{ADD}** Soft button to enter another Station.
4. Press **{FINISH}** Soft button.
5. Enter message.
6. Press **{SEND}** Soft button.

To check SMS:

1. Press the **[CALL BACK]** button and dial '3',
OR
Press the **[PGM]** button and dial 44.
2. Received message is displayed on the LCD.
3. To check the next message, Press **{NEXT}**.
4. To delete, Press **{DELETE}**.

Conditions

1. Stations can send a SMS to up to 20 stations simultaneously.
2. SMS is supported in LIP-8000, LIP-7000, LDP-7024LD /w graphic LCD Keypad and LKD-30DH.
3. Station can save 20 SMS; when the 21st SMS message is received, the 1st SMS is deleted and 21st SMS is saved.

Programming

Related Features

Hardware

Digital Phone w/Display
Phontage
UCS Client

4.9.2 Station Message Wait/Call Back

Description

When a Called Station does not answer or is in DND, a Station User can activate a Message Wait Indication (MWI) to request a Call Back. A Station may receive a MWI from any number of other Stations in the System. The Station receiving the MWIs can return the calls using the **[MSG/CALLBK]** button.

When a Busy Station is called, the Calling User may request to be placed in a queue to receive a Call Back. When the Called Station returns to Idle, the System notifies the Initiating Station with Call Back ring. When the User answers, the previously Busy Station is called.

When a Message is waiting, the **[MSG/CALLBK]** button LED will flash; when MWI is received at a SLT, the MW lamp will flash.

Operation

Digital Phone

To leave a Message Wait:

1. While receiving a Ring Back tone or No answer on a Call Announce (H or P mode).
2. Press the **[MSG/CALLBK]** button; a confirmation tone will be heard.
3. On Hang-up, the MWI is activated.

To leave a Message Wait:

1. While receiving a DND tone, press the **[MSG/CALLBK]** button; a confirmation tone will be heard.
2. On Hang-Up, the MWI is activated.

To leave a Call Back (queue for a station):

1. While receiving a Busy Signal on an attempted call, press the **[MSG/CALLBK]** button; the confirmation tone will be heard.
2. Hang up, and return to Idle.

To respond to a Call Back Recall Request:

1. When the Busy Station returns to Idle, the System initiates a Call Back to the Originating Station.
2. Lift the handset or press the **[SPEAKER]** button.
3. Previously Busy Station is called.

To retrieve Station MWIs:

1. At the Station, press the **[MSG/CALLBK]** button; either the message contents summary will be displayed (shown), or the Station Messages Waiting notification will be displayed.

MWI(05) VMS(03) ENTER (MWI:1, VMS:2)

2. Dial 1 or 2,
 - 1. MWI: Absent Call

- 2. VMS: VMIB Message
- 3. Press the **[VOL UP]/[VOL DOWN]** button to scroll.

To return a Call from the Current Station Message:

1. While message is active, press the **[HOLD/SAVE]** button.

SLT

To leave a MWI:

1. While receiving a Ring Back tone or No Answer on a Call Announce (H or P mode), press the Hook-Switch.
2. Dial the {Message Wait/Call Back code}.
3. Hang-up, the MWI is activated.

To leave a MWI:

1. While receiving a DND tone, press the Hook-Switch.
2. Dial the {Message Wait/Call Back code}.
3. Hang-Up, the MWI is activated.

To retrieve a Station Message Wait:

1. Dial the {Message Wait/Call Back Answer code}.

To leave a Call Back (queue for a station):

1. While receiving a Busy Signal, press the hook switch.
2. Dial the {Message Wait/Call Back code}.
3. Hang up, and return to Idle.

To respond to a Call Back Recall:

1. When the Busy Station returns to Idle, the System initiates a Call Back.
2. Lift the handset, or press the [Speaker] button.
3. The previously Busy Station is called.

Conditions

1. A Message Wait/Call Back Return Call will always ring at the Receiving Station overriding the Intercom Signaling mode selected.
2. A Station can leave only one Call Back request.
3. The **[MSG/CALLBK]** button LED will continue to flash until all MWIs and Call Back Requests, have been serviced (including Voicemails).
4. If a Station is attempting to leave a message and the System MWI queue is full, the oldest MWI will be deleted.
5. A MWI Reminder Tone can be enabled to remind the user of MWIs.
6. A Station in Call Forward can leave a MWI.
7. A MWI is left at the Originally Called Station even if the call is Forwarded.
8. A Digital Phone with LCD may Call Back to Stations that left messages in any desired order, or the normal ("oldest first") order.
9. Placing an Intercom call to a Station will cancel any existing MWI from that Station.

10. A Station can support up to 99 VMS messages.
11. If a Station requests a Call Back at a Busy Station, a Call Back Request Station will check the Busy Station's status every 5 sec., and receives Call Back Ring when the Status Check Timer is expired (after the Busy Station returns to Idle). For this reason, Call Back ring may be delayed after a Busy Station returns to Idle, and when several Stations request a Call Back at a Busy Station, the Call Back Ring may not be provided sequentially.

Programming

- System Data**
1. Message Reminder Tone Timer (PGM 222-Index2)

Related Features

Hardware

4.9.3 Message Wait Reminder Tone

Description

In addition to the [MSG/CALLBK] button LED, the Digital Phones can be sent a tone as a periodic reminder to the user of MWIs in queue. This tone is sent to the Station only while Idle, and is heard over the Speaker.

Operation

System

When programmed, the MWI Reminder Tone is automatically sent to Stations in Idle as appropriate.

Conditions

1. The interval between tones can be 00 to 60 minutes; the 00 setting will disable the MWI Reminder Tone.
2. The Reminder Tone will continue repeating notification until all messages have been retrieved.
3. A Busy Station or Station in DND will not receive the MWI Reminder Tone.
4. SIP Phone and DECT do not support the MWI Reminder Tone feature.

Programming

- System Data**
1. Message Reminder Tone Timer (PGM 222-Index2)

Related Features

Hardware

Digital Phone

4.10 PAGING

4.10.1 Internal/External & All Call Page

Description

A Station that is set-up for using the Page features, can connect and transmit voice announcements to any or all of the System Internal/External Page Zones. Stations are grouped into “Zones” to receive Pages directed at each Zone. Stations not assigned to any Zone will not receive a Page including All Call Pages.

A Page Ring will be provided to the Page Zone(s) prior to the Audio Connection. The User is allowed to continue the Page for the specified Page Time-out Timer after which the User is disconnected and the Page Zone(s) is returned to idle.

The default Page Zone Dial codes are:

Page Type	Page Zone Dial Codes	
	iPECS-MG 100	iPECS-MG 300
Internal Page Zones	543 + Group Number	543 + Group Number
Internal All Call Page	543 = 00	543 + 00
External Page Zone	546	546
All Call Page	547	547

Flexible buttons of a Digital Phone may be assigned to access a Page Zone as a {PAGE ZONE} button.

Operation

Digital Phone

To assign a Flex button as a {PAGE ZONE} button:

Press [PGM] + {FLEX} + Button Feature Type(1) + {Paging Code} + [SAVE]

To initiate a Page:

1. Lift the handset.
2. Dial the desired Paging code or press a {PAGE ZONE} button.
3. Dial 1 (Speaker Paging).
4. If assigned, after the Page Ring, make the desired announcement.
5. When finished, replace the handset and go On-Hook.

SLT

To initiate a Page:

1. Lift the handset.
2. Dial the desired Paging code.
3. Dial 1 (Speaker Paging).
4. If assigned, after the Page Ring, make the desired announcement.
5. Replace the handset, go On-Hook.

Conditions

1. Stations that are not enabled to Page, will receive an Error tone when any Page Access code is dialed.
2. Stations in DND or Busy will not receive Page announcements.
3. A Station accessing a Page Zone is considered Busy.
4. Stations, that are not included in any Internal Page Zone, will not receive any page, including All Call.
5. For External Paging, an External Amplifier and Speaker(s) are required.
6. The System External Control Contacts may be assigned to activate when External Page is accessed.

Programming

Station Data	1. Internal Page Group Access (PGM 151)
Station Group Data	1. Page Group (PGM 205)
Tenant Data	1. Page Tone (Web Admin. PGM 290-Index33)
System Data	1. Paging Time-out Timer (PGM 220-Index 4)
Numbering Plan	1. Internal Page Answer Code (PGM 115-Index 2) 2. Page Auto Answer (PGM 115-Index69)

Related Features

Meet Me Page Answer

Hardware

External Amplifier & Speakers

4.10.2 Meet Me Page Answer

Description

Any station may respond to a Meet Me Page request over an Internal or External Page Zone. The user answers the page from any Station and is connected to the Paging Party. The Paging Party can answer the page by pressing the [HOLD/SAVE] fixed button and other users can answer by dialing the {Meet Me Page Feature Code}.

Two kinds of Page Answer codes are supported:

- {Auto Page Answer Code} – When activated, the System will search the available communicated Page Call automatically from External Page to Last Page Group Call.
- Searching rules: External Page > Page Group1 > Page Group2 > > Last Page Group
- {Int Page Answer Code} – User can choose exact available communicated Page Group; if user dials code (can be assigned at Flexible buttons of a Digital Phone), user can choose the Group number.

Operation

Digital Phone

To assign a Flex button as a {MEET ME} button:

1. Press the [PGM] + {FLEX} + Button Feature Type (1) + {Int Page Answer Code} + [SAVE].

To answer a Page by the Paging Party:

1. Lift the handset.
 2. Press the [HOLD] button
- OR
1. Press [HOLD] button.

To answer a Page with Meet Me Page Code from another Station:

1. Lift the handset or press the [SPEAKER] button.
 2. Dial the {Int. Page Answer Code} or press the {MEET-ME} button.
 3. Dial the Group number.
- OR
1. Lift the handset or press the [SPEAKER] button.
 2. Dial the {Auto Page Answer code} from a Station; after the system locates the appropriate Station Group, the call it will be connected to the Paging Station.

Conditions

1. A Meet Me Page must be answered within the Page Time-out Timer.
2. A Station may answer a Meet Me Page from any Station regardless of Pick-Up/Paging Group assignments and Page access permission.
3. The Paging Party must remain Off-Hook until the Paged Party answers the Meet Me request. The Initiator/Paging Party may press the Mute button to eliminate transmitting over the Page circuit while waiting for the Paged Party to answer.
4. A SLT user cannot answer a Meet Me Page.

Programming

- | | |
|-----------------------|---|
| System Data | 1. Paging Timeout Timer (PGM 220-Index4) |
| Station Data | 1. Meet-Me Access (PGM 134- Index3) |
| Numbering Plan | 1. Int Page Answer Code (PGM 115-Index2)
2. Page Auto Answer (PGM 115-Index69) |

Related Features

Internal/External & All Call Page

Hardware

4.10.3 VM Paging

Description

Users can distribute a page they have recorded using the {VM Page Message Record} feature code.

Operation

Digital Phone

To record a VM Message:

1. Lift the handset.
2. Dial the **{VM PAGE MESSAGE RECORD}** feature code.
3. Record VM Message.

To delete a recorded VM Message:

1. Lift the handset.
2. Dial the **{VM PAGE MESSAGE RECORD}** feature code.
3. Dial **[Speed]** Button for deletion.

To make a VM Page:

1. Lift the handset.
2. Dial the desired paging code or press a **{PAGE ZONE}** button.
3. Dial 2 for VM paging.
4. If assigned, after the Page Ring, VM message will be played.
5. Replace the handset, go on-hook.

SLT

To make a page:

1. Lift the handset.
2. Dial the appropriate paging code.
3. Dial 2 for VM paging.
4. If assigned, after the Page Ring, make announcement, or hear VM message.
5. Replace the handset, go on-hook.

Conditions

1. If paging user is making a VM-Page, then user will hear the recorded VM message.
2. During VM paging, Meet Me Page Answer function is available.

Programming

- | | |
|---------------------------|--|
| Station Data | 1. Internal Page Group Access (PGM 151) |
| Station Group Data | 1. Page Group (PGM 205) |
| Tenant Data | 1. Page Tone (Web Admin. PGM 290-Index33) |
| System Data | 1. Paging Timeout Timer (PGM 220-Index4) |
| Numbering Plan | 1. VM Page Message Record (PGM 113-Index88)
2. Int Page Answer Code (PGM 115-Index2)
3. Page AUTO Answer (PGM 113-Index14) |

Related Features

Internal/External & All Call Page
Meet Me Page Answer

Hardware

4.11 PUSH-TO-TALK PAGING

Description

Digital Phone can be assigned as a member of one of the System's Push-To-Talk (PTT) Page Groups. The Digital Phone User may log-in or log-out of any one or all PTT Groups it is a member of. Once logged in, the User may place or receive One-Way Page announcements to/from other Users who are logged-in to the same PTT group. To place a PTT Page announcement, the User must press and hold the **{PTT}** Flex button.

An Attendant may log-in/-out other Stations to/from PTT groups.

Operation

Digital Phone

To assign a {PTT} Flex button:

Press **[PGM] + {FLEX} + Button Feature Type (1)+ {PTT} + [SAVE]**

To Log-In to a PTT group:

1. Dial the {PTT Log-in/-out code}.
2. Dial the desired PTT Group Number.

To Log-Out of the PTT group(s):

1. Dial the {PTT Log-in/-out code}.
2. Dial '*'.

To place a Page to the active PTT Group:

1. Press and hold the **{PTT}** Flex button.
2. Make the desired Page announcement after hearing confirmation tone.

Conditions

1. Conditions associated with Internal/External & All Call Page apply to PTT Paging.
2. To access PTT Paging, the Station must be permitted access to System Paging.
3. If allowed access to all PTT Groups, a Station may log-into all Groups (PTT Group 0) to place announcements to all Groups simultaneously and receive announcements from any group.
4. A Station can only Log-In to PTT groups to which it is assigned as a member.
5. The Station must have a **{PTT}** button to place or receive PTT announcements; by default, the WLAN phone is assigned a PTT button.
6. The Station may be assigned and Logged-In to the default active PTT group in the System database.

Programming

- | | |
|---------------------------|---|
| Numbering Data | 1. PTT Log-In/-Out code (PGM 113) |
| Station Group Data | 1. PTT Group (PGM 208) |
| Station Data | 1. Fixed PTT Flex Button Assignment (PGM 126) |

Related Features

Paging

Hardware

Digital Phone
Phontage
UCS Client

5. CO/IP

5.1 ALTERNATIVE ROUTE SELECTION

Description

This feature is enabled if there are several paths in order to connect toward a destination System. If a selected path is not available for some reason (All Busy, Line Fault, etc), Alternative Route Selection (ARS) will connect calls using another designated path.

Operation

If set, Alternative Route Selection operation is automatic.

Conditions

1. ARS is optional and must be programmed to be operational.
2. Up to 2 ARS pats can be assigned for each CO Group Access code.
3. ARS digits should be contained in the CO Group Access Code, and will be applied when feature is initiated.
4. ARS can be used in coordination with Last Number Redial, Station Speed Dial, and System Speed Dial.

Programming

- | | |
|-----------------------|---|
| Numbering Plan | 1. CO Group Access Code (PGM 114) |
| CO Line Data | 1. CO Group Access Code Attributes (PGM 180) <ul style="list-style-type: none">- ARS Service (Index5)- ARS Digit (Index6-7)- AND Digit (Index4) |

Related Features

Hardware

5.2 AUTOMATIC NETWORK DIALING

Description

When a Station User dials a CO Group Access Code and a CO Line is seized, Automatic Network Dialing (AND) enables an automatic digit dialing. Using this feature, a CO Group Access code can be dialed automatically when a User is making an external call through another System.

Conditions

1. AND can place calls automatically by dialing all of the necessary digits or some digits, as programmed.
2. AND digits (up to 10) for placing calls can be registered on each CO Group Access codes.
3. If the CO line is seized using the {CO Line Access} feature code, AND digit is not supported.

4. AND is enabled only when CO line is seized using a CO Group Access Code.

Programming

Numbering Plan	1. CO Group Access Code (PGM 114)
CO Line Data	1. CO Group Access Code AND Digit (PGM 180-Index4)

Related Features

Hardware

5.3 CO GROUP ACCESS CODE

Description

The CO Group Access code can be set dynamically for each Outgoing CO Group. One Outgoing CO Group can have several access codes. Additionally, there are various kinds of services for each access code:

- **Access Code Name** – Access code name can be displayed when a user seizes a CO line with CO Group Access code.
- **CO Line Choice** – There are 3 ways to seize a CO line (Round / First / Last).
- **Outgoing CO Group** – When a user dials a CO Group Access code, the System will seize a CO Line with the Outgoing CO Group.
- **AND** – When using CO Group Access code, it can add max. 10 digits to be able to send towards PX automatically.
- **ARS Service** – When a selected path is not available for some reasons(all busy, line fault, etc), this feature connects a call using another path which is preset automatically

Operation

Conditions

1. This feature can be set for each Outgoing CO Group.
2. If CO Line has been seized directly, not using a CO Line Access Code, this feature is not enabled.
3. When Outgoing CO Group is none, a CO Line can be seized according CO Group Access.

Programming

Numbering Plan	1. CO Group Access Code (PGM 114)
Station Data	1. CO/IP Group Access (PGM 150)
CO Data	1. CO Group Access Code Attributes (PGM 180)

Related Features

Hardware

5.4 CO LINE FLASH

Description

Analog CO Lines recognize a brief open or ground connection (Flash), as a request for a new dial tone. When used behind a PBX, a CO Line Flash is often used to activate a PBX feature or Call Transfer.

Operation

Digital Phone

While connected to an Analog CO line:

1. Press the {CO Flash}.button, the System automatically will generate a Flash on the CO Line.

SLT

While connected to an Analog CO line:

1. Press the Hook-switch.
2. Dial {CO Flash Feature Code}.

Conditions

1. Stations may Flash on a CO Line defined for PABX operation and will experience COS dialing restrictions if a PABX Trunk access code is dialed.
2. During Flash, the LED for the CO Line button will remain lit.
3. A Flash may be stored as a part of a Station or System Speed Dial number.
4. While connected to an Internal Call or Dial Tone, pressing the {CO Flash} button will re-access the ICM Dial tone.
5. Flash function is not available on Digital CO Lines like ISDN, VoIP and R2.

Programming

- | | |
|-----------------------|--|
| Numbering Plan | 1. Feature Numbering Plan – CO Flash (PGM 113)
2. Feature Numbering Plan – CO Line Access (PGM 113) |
| System Data | 1. LED Color/Flash Rate (PGM 234) |
| CO Group | 1. CO Incoming BLF Usage (PGM 166–Index2)
2. CO Outgoing BLF Usage (PGM 171–Index4) |

Related Features

Station Speed Dial

Hardware

5.5 CO/IP LINE GROUPS

Description

All CO Lines are included on one Outgoing CO Group and one Incoming CO Group individually. One CO Line cannot be a member of several CO Groups at the same time.

The CO/IP Lines in the System can be placed together into Groups for assigning access by Stations and common Access Dial codes (up to 24 Groups on iPECS-MG 100, and 72 Groups on iPECS-MG 300).

Operation

Conditions

1. Outgoing CO Line Groups and Incoming CO Line Groups are separated.
2. CO/IP Lines in each CO Group can also be accessed individually by dialing the {CO Line Access Feature Code} and the CO/IP Line number.
3. The System will select a CO/IP Line from a group based on the Round Robin, First Choice or Last Choice determined by database assignments.
4. Multiple {CO Group Access Code} can be assigned to a CO Group.
5. If a CO Line is not included in an outgoing CO line group, this line has to be seized using a Flex button or System feature code.

Programming

System Data

1. CO Group Access Code (PGM 114)

CO Data

1. Outgoing Group Number (PGM 160-Index3)
2. Incoming Group Number (PGM 160-Index4)
3. CO Group Access Code Attribute (PGM 180)
4. CO Line Choice (PGM 180-Index2)

Related Features

Hardware

5.6 CO LINE SERVICE

The System can set Service attributes and several options according to each CO Line.

5.6.1 Incoming CO Line Option

Description

There are some options to support the Incoming CO line:

- **Progress Indicator (Send)**
- **CID Prefix Code** – Add to prefix code before CLI.
- **Own Code Add to Transit CID** – Own code can be added to incoming CLI when external User places an outgoing call through CO Line in the System.
- **Provide Dial Tone** – Dial Tone can be provided to a Station if PX is not given the dial tone.
- **CPT Detect** – System can release the CO Line by detecting the External User's disconnection.
- **Own Code** – Own code is added before the Station number when CLI information is available.

- **Max Ring Time** - System releases the CO Line if the transferred/held incoming CO line is not answered prior to Timer Expiration.

Operation

Conditions

1. This feature can be set for each outgoing CO line.

Programming

- | | |
|---------------------|---|
| CO Line Data | 1. CO Line Data (PGM 160) |
| | 2. Incoming CO Attributes (PGM 165-166) |

Related Features

Hardware

5.6.2 Outgoing CO Line Option

Description

The following options support the Outgoing CO line:

- **Screen Indicator** – Determines if Screen Indicator information is included when it consists of ISDN CLI information.
- **Calling Party Number** – Determines if Calling Party Number information is included when it consists of ISDN CLI information.
- **ISDN Calling Number** – Determines if the ISDN Setup Message includes CLI information or not.
- **ISDN Bearer Capability** – Includes the specific Bearer Capability information on the Setup Message ignoring the received Bearer Capability if an ISDN Internal User or PRI Transit Incoming Call places an Outgoing Call.
- **Digit Sending Complete IE** – Determines if Digit Sending Complete IE message is included on an ISDN information message.
- **CLI Information when Transit** – Determines if CLI Information is included on Outgoing Transit Calls when no information is provided on Incoming Transit Calls.
- **CLI Information to Transmit Toward PX on Outgoing Transit Calls** – Creates new CLI by adding or deleting the received CLI Information.
- **Calling Party Numbering Plan** – Determines Calling Party Numbering Plan information if it consists of ISDN CLI information on Outgoing Transit Call.
- **RBT Support** – Determines if System supports RBT if the PX does not support RBT when an Internal User places an Outgoing Call seizing the Trunk CO Line.

Operation

If set, Outgoing CO Line operation is automatic.

Conditions

1. This feature can be set for each outgoing CO line.

Programming

- CO Data**
1. CO Line Data (PGM 160)
 2. Outgoing CO Attributes (PGM 170-171)

Related Features

Hardware

5.6.3 Alternate Incoming CO Service

Description

If the system cannot answer for an incoming CO call, a programmed service can be provided or it can be routed into a programmed destination. This feature is applied by incoming CO Group base Administrator to select the Routing Destination for the Incoming CO line on a case-by-case basis as follows:

- **Busy** – When calling a Busy User, the call can be set for routing destination, Disconnect, ATD, CO Ring Assign, Alt Ring Table, Tone, or Pilot Hunt.
- **No Answer** – When an Incoming Call goes unanswered, it can be set for routing destination, Disconnect, ATD, CO Ring Assign, Alt Ring Table, Tone, or Pilot Hunt.
- **Transfer No Answer** – When an Incoming Call goes unanswered after an unscreened transfer, the call can be set for routing destination, Disconnect, ATD, CO Ring Assign, Alt Ring Table, Tone, Pilot Hunt, or Transfer Station.
- **Recall No Answer** – When an Incoming Call goes unanswered after a recall on a CO call, the call can be set for routing destination, Disconnect, ATD, CO Ring Assign, Alt Ring Table, Tone, or Pilot Hunt.
- **Vacant** – If dialing analysis determines an Incoming CO Call is to a vacant number, the call can be set for routing destination, Disconnect, ATD, CO Ring Assign, Alt Ring table, or Tone.
- **DND** – If an Incoming CO call is attempted to a DND User, the call can be set for routing destination, Disconnect, ATD, CO Ring Assign, Alt Ring table, Tone, or Pilot Hunt.
- **Handset Lifted** – If an Incoming CO Call is received at a Station where the handset is lifted (Off-Hook), the call can be set for routing destination, Disconnect, ATD, CO Ring Assign, Alt Ring table, Tone, or Pilot Hunt.
- **Blocking** – If an Incoming Call is placed to a Blocked User, the call can be set for routing destination, Disconnect, ATD, CO Ring Assign, Alt Ring table, Tone, or Pilot Hunt.

Operation

If set, Alternate Incoming CO Service operation is automatic.

Conditions

1. This feature can be set for each incoming CO Group.
2. The destination is set differently according to Day, Night, and Time modes.
3. If the destination is a Ring Table, all features of the Ring Table can be used together.

Programming

- CO Line Data**
1. CO Line Data (PGM 160)
 2. Incoming CO Line Attributes (PGM 165-166)
 3. Incoming CO Line Alternative Destination (PGM 169)
 4. Alternative CO Ring Table (PGM 181)

Related Features

Hardware

5.6.4 Alternate Outgoing CO Service

Description

A User can place an outgoing CO call and then can Hold or Transfer the Call to another User. If the System cannot answer for an external Outgoing Call, a programmed service can be provided or it can be routed into a programmed destination. This feature is applied by CO Line Group by the Administrator.

- **Recall No Answer** – Station does not answer the Hold Recall, the call can be set for routing destination, Disconnect, ATD, CO Ring Assign, Alt Ring table, Tone, or Pilot Hunt.
- **Transfer No Answer** – Transferred call goes unanswered, the call can be set for routing destination, Disconnect, ATD, CO Ring Assign, Alt Ring table, Tone, Pilot Hunt, or Transfer Station.
- **No Answer** – If an incoming call goes unanswered, the call can be set for routing destination, Disconnect, ATD, CO Ring Assign, Alt Ring table, Tone, or Pilot Hunt.

Operation

If set, Alternate Outgoing CO Service is automatic.

Conditions

1. This feature can be set for each CO line.
2. If attendant is not assigned and it calls an incoming CO call, it plays error tone.
3. If there is no voice mail resource or all voice mail channels are busy, it plays error tone.
4. The destination is set differently according to Day, Night and Timed modes.
5. If the destination is a ring table, all features of ring table can be applied.

Programming

- CO Line Data**
1. CO Line Data (PGM 160)
 2. Outgoing CO Alternative Attribute (PGM 173)

Related Features

Hardware

5.6.5 Digit Sending Mode

Description

For calls that are initiated using an Access Code, the sending digits can be set for en-blocked or overlapped.

Operation

If set, Digit Sending Mode is automatic.

Conditions

Programming

CO Line Data 1. Digit Sending Mode (PGM 162-Index2)

Related Features

Hardware

5.6.6 CO COS

Description

The permission/restriction level can be set for each CO line. If an Internal user with lower-level permissions attempts a call or trunk dials, COS will be applied.

Operation

If set, CO COS operation is automatic.

Conditions

Programming

CO Line Data 1. CO COS Assign (PGM 177)

Related Features

Hardware

5.7 CO/IP LINE PRESET FORWARD

Description

Each CO/IP Line can be assigned a Ring-No-Answer Preset Forward destination. An incoming call on the CO/IP line will be routed to the designed Ring Destination, following expiration of the CO/IP Line Preset No Answer Forward Timer. Preset Forward destination, determined according to the Ring Assignment Table.

Operation

System

If set, CO/IP Line Preset Forward is automatic.

Conditions

1. This feature can be set for each Normal-type CO Line, not DID-type CO Lines.
2. CO/IP Line Preset Forward is available only when Incoming CO Ring Group destination is DN; not activated when destination is a Station Group or CCR.
3. CO/IP Line Preset Forward will override Call Forward No-Answer at a Station.
4. If Destination Station has an External No Answer Preset Call Forward destination set and the timer is same as CO Preset Forward Timer, CO Preset Call Forward will take priority. Otherwise the timer that expires first will be applied and Forward Destination will be applied according to the expired timer.
5. CO/IP line Preset Forward is disabled if the Preset Forward Timer is set to 0.
6. CO/IP line Preset Forward destination cannot be a VM Group.
7. CO/IP line Preset Forward is not applied to the DID line.

Programming

CO Line Data

1. CO Service Type (PGM 160–Index2)
2. Preset Forward Timer (PGM 168–Index4)
3. Preset Forward Ring Table Index (PGM 168–Index5)
4. Alternative CO Ring Table (PGM 181)

Related Features

Call Forward
Call Forward, Preset

Hardware

Related Features

5.8 CO OWN CODE SERVICE

Description

If a Station places a CO Call, it can send CLI including its CO Own Code towards the receiving PX. Alternatively, CO Own Code can be included into CLI Information if an Incoming CO Call makes an Outgoing CO Call. The CO Own Code is sent adding ahead of station number or received CLI.

Operation

Conditions

1. This feature can be set for each outgoing/incoming CO line.
2. The outgoing CO own code and the incoming CO own code are independent.
3. The maximum own code length is 16.

Programming

CO Line Data

1. Outgoing CO Own Code (PGM 170-Index12)
2. Incoming CO Own Code (PGM 165-Index9)

Related Features

Hardware

5.9 CO/IP RING ASSIGNMENT

Description

Each station in the system can be programmed to provide an audible signal when the system detects an incoming call on specified CO/IP lines. Separate ring assignments are made for Day, Night and Timed Ring operation mode. In addition, the audible signal at the station can be delayed by 1 to 9 ring cycles allowing other stations to answer the call first.

Operation

System

Operation of this feature is automatic.

Conditions

1. Separate assignments are made for Stations to ring in the Day, Night and/or Timed Ring mode.
2. Audible Alerting for an Incoming VoIP call is based only on the derived IP Address.
3. A Busy Station receives Muted Ring or Call Waiting tones as appropriate for the Station's Off-hook Ring Assignment.
4. The system Ring mode can be selected manually or automatically,
 - Automatic mode: Day/Night selection is determined based on the Automatic Ring Mode Selection table.
 - Manual mode: The Attendant has manual control over the Ring mode selection.
5. The LCD of the Attendant Station will display Night and Timed Ring Mode.
6. If CO/IP Ring is assigned to only one Station and that Station assigns Call Forward, the CO/IP Call will be routed according to Call Forward settings.

Programming

CO Line Data

1. CO Ring Table (PGM 167)

TABLE DATA

1. System Time Table (PGM 253)
2. Weekly Time Table (PGM 254)
3. Holiday Time Table (PGM 256)

Related Features

Day/Night/Timed Ring Mode
Auto Service Mode Control
Off-Hook Signaling

Hardware

5.10 CO LINE RELEASE GUARD TIME

Description

To assure that the PSTN switching equipment has sufficient time to restore to the Idle condition, the System will hold CO Lines in a Busy State to Users after release of a CO Line by a Station. The time between Station disconnect and when the System changes the CO Line status from Busy to Idle is the CO Line Release Guard Time. If CO receives 'Release Ack' signal from PSTN before Release Guard Timer expired, then the CO line is released instantly.

Operation

System

Operation of this feature is automatic.

Conditions

Programming

- | | |
|---------------------|--|
| CO Line Data | 1. Incoming CO Release Guard Timer (PGM 166-Index8)
2. Outgoing CO Release Guard Timer (PGM 171-Index5) |
| System Data | 1. LCO Release Guard Timer (PGM 221-Index6) |

Related Features

Hardware

5.11 CO RING DETECT

Description

The System incorporates timers for Ring-On and Ring-Off durations to assure proper alerting. When the duration of the Ring Signal exceeds the Ring-On Timer, alerting will start. When the Ring is not present for a period exceeding the Ring-Off Timer, alerting will stop. This allows the System Ring cycle detection to be matched to the many and varied PBX systems.

Operation

System

Operation of Ring detect is automatic.

Conditions

1. Ring-On time and Ring-Off time are assigned on a System basis.
2. The CO Ring Detect is applied to Analog CO Lines only.

Programming

- | | |
|--------------------|---|
| System Data | 1. LCO Ring ON Timer (PGM 221-Index4)
2. LCO Ring OFF Timer (PGM 221-Index5) |
|--------------------|---|

Related Features

Hardware

5.12 CO TRANSIT SERVICE

Description

The System provides a function for Incoming CO Calls to make a call on an outgoing CO Line directly. In this case, transit switch-over can be supported for using different types of signaling. Incoming Calls from R2 transit can be turned into a PRI transit call. The System supports R2, ISDN, H.323, SIP, and analog CO methods of transit.

Operation

If set, CO Transit Service operation is automatic.

Conditions

1. The CO transit service can be set through Admin. programming of optional permissions.
2. The System provides an inter-working feature for all CO Lines.
3. For a R2 CO to R2 CO call, set for link-by-link or end-to-end transit.
4. For a PRI CO to PRI CO call, all messages are Forwarded transparently.

Programming

- CO Line Data** 1. CO to CO Attributes (PGM 179)

Related Features

Hardware

5.13 DIAL PULSE SIGNALING

Description

An Analog CO line will send dial Pulse Signals to the Central Office. If programmed as a Pulse CO line, the System will send open loop pulses at 10 pps with the assigned Break/Make Ratio.

Operation

System

Operation of this feature is automatic when programmed.

Conditions

1. The Break/Make Ratio is System programmable as 60/40 , 66/33 or 50/50.

Programming

- CO Line Data** 1. CO Line Dialing Type (PGM 160-Index10)
System Data 1. Pulse Dial Break Ratio (PGM 223-Index2)

Related Features

Dial Pulse to Tone Switchover

Hardware

5.14 DIRECT INWARD DIAL (DID)

Description

A Carrier Service, known as Direct Inward Dial (DID), sends digits to the System so that the call may be routed directly to a Specific Station or System Facility. DID service is available over digital and packet networks. ISDN lines can provide two-way, incoming DID and normal Outgoing Service, and requires no special signaling.

After collecting the digits from the Carrier, the System routes the Call to the Destination:

- Incoming DID number is compared with Digit Conversion Table. If matched, received DID number is converted according to the Table. Separate Digit Conversion can be applied according to DAY/Night/Timed Ring Mode.
- DID Destination is decided with the converted DID number.
- Destination can be Station, Station Group, Outgoing CO Call, Voice Mail, Net Station, Paging, or Conference Room.

Operation

If set, DID operation is automatic.

Conditions

1. If ICLID routing is assigned for the CO/IP Line, the received Caller ID is first compared to the ICLID Table for routing. If Caller ID does not match an entry in the ICLID Table, the normal DID call processes are used.
2. DID calls that encounter a Busy signal, are not answered in the DID/DISA No-Answer Timer, or are received at a Vacant or Invalid number can be routed to the Attendant, Tone, Station Group, or VMIB announcement. When the Attendant receives such calls, the call is appropriately identified by the Attendant Digital Phone Display.
3. For a Station that is part of a non-pilot Station Hunt group, DID calls will follow the Group Hunt process if the Station is Busy or does not answer the call.
4. DID calls are subject to Group Call Pick-up and Directed Call Pick-up.

Programming

Station Data	1. Station DN Number (PGM130)
CO Line Data	1. CO Line Attribute (PGM160) 2. Incoming CO Group Alternative (PGM169)
Numbering Plan	1. Feature Numbering Plan (PGM 113)
Table Data	1. Digit Conversion Table (PGM 251/252)

Related Features

Hardware

DID, BRI or PRI Module

5.15 DIRECT INWARD SYSTEM ACCESS (DISA)

Description

Each CO/IP path may be assigned for DISA Service, which allows an Incoming Caller to gain access to System resources and/or features. The System will answer the outside call and provide the User Intercom Dial tone or route the call based on the VMIB Auto Attendant announcement settings where Caller Controlled Routing (CCR) may be defined. The DISA caller may then access the desired resource using dial codes.

If an Authorization Code is required for DISA access, when the System answers an Incoming DISA call, DND Warning tone is provided to indicate an Authorization Code must be entered.

Operation

System

Incoming calls enabled for DISA service:

1. The System will recognize the Incoming call.
2. The System will answer the call and connect the caller to the Intercom Dial tone or AA announcement.
3. The Call will be processed based on the entered digits/programming.

DISA Caller

To remotely access System resources:

1. Place call to the System DISA facility.
2. On receipt of the Intercom Dial tone/AA Announcement, dial as desired.

Conditions

1. Each CO/IP path is separately assigned for DISA Operation during Day, Night and/or Timed System Operation modes. DISA operation is active only when the System is in the assigned operating mode(s).
2. A DISA Caller can be required to enter an Authorization Code to access System external outgoing resources, facilities or features. If required, the caller is permitted to retry entry of a valid Authorization Code based on the DISA Retry count (continued failure will cause disconnection).
3. DISA callers are subject to COS dialing restrictions. If Authorization Codes are required and the code matches a Station Authorization Code, the Station COS will apply.
4. The System will disconnect an Outgoing DISA call if the Unsupervised Conference timer expires or Disconnect Supervision is received; a Disconnect Warning tone is provided 15 seconds prior to disconnect.
5. If a DISA caller encounters a System All Lines Busy, Busy tone is received for 5 seconds before ICM Dial tone is presented again and the DISA caller may try another call.
6. LEDs associated with the DISA CO Line appearance will provide normal status indications at all Stations except Attendants; the LED for the line at an Attendant Station will flutter at 240 ipm when Busy.

7. If a DISA Caller accesses a CO/IP Line, the Transit option is applied (CO to CO Attribute); this function can make a call recovered after conversation during an assigned time.

Programming

- | | |
|-----------------------|--|
| Numbering Plan | 1. Feature Numbering Plan, DISA Tone Service (PGM 113) |
| CO Line Data | 1. CO COS (PGM 177) |
| | 2. CO Access From DISA (PGM 168-Index1) |
| | 3. DISA Account Code (PGM 168-Index2) |
| | 4. DISA Retry Count (PGM 168-Index3) |
| | 5. CO to CO Attribute (PGM 179) |
| | 6. CO Ring Assignment (PGM 167) |
| | 7. DISA Supervision Timer (PGM 166-Index12) |
| | 8. VMIB Play Delay Timer (PGM 166-Index13) |

Related Features

VMIB-Auto Attendant
Day/Night/Timed Ring Mode
Dialing Restrictions
Authorization Codes (Password)
Unsupervised Conference
VMIB Integrated Auto Attd/Voice Mail
Auto Service Mode Control

Hardware

5.16 DUAL TONE MULTI-FREQUENCY (DTMF) SIGNAL SENDING

Description

CO Lines can be assigned for Dual Tone Multi-Frequency (DTMF) signaling.

Operation

System

Operation of this feature is automatic when programmed.

Conditions

1. The System will mute the User's voice transmission to reduce interference while sending DTMF tones.

Programming

- | | |
|---------------------|---|
| CO Line Data | 1. CO COS (PGM 177) |
| | 2. Outgoing CO Line Inter-Digit Timer (PGM 174) |

Related Features

Hardware

5.17 H.323 MULTI ROUTE SERVICE

Description

The system can set-up several destination IPs for one prefix. The destination IP is selected circularly when user tries to make a H.323 call.

Operation

Operation of this feature is automatic when programmed.

Conditions

1. This feature is for only VoIP (H.323).

Programming

- | | |
|--------------|--|
| H.323 | <ol style="list-style-type: none">1. H.323 Route Attribute (PGM 360)2. H.323 VOIB Attributes (PGM 361)3. H.323 Incoming Attributes (PGM 362) |
|--------------|--|

Related Features

Hardware

5.18 INCOMING CALLING LINE ID (ICLID) CALL ROUTING

Description

The system can employ Incoming Calling Line ID (ICLID) to determine the routing of Incoming external calls. Each CO/IP Line, including DID Lines, may be assigned to employ ICLID routing. The System will compare the received ICLID to entries in the ICLID Routing Table. If a match is found, the call will be routed to the destination defined in the ICLID Ring Assignment Table. Destinations can be the VMIB, an external Voice Mail, a Station or a Station Group.

Operation

System

The System will implement routing automatically based on database entries and the received ICLID.

Conditions

1. If the received ICLID does not match an entry in the ICLID Route Table, the call is routed according to CO Ring Assign Table.
2. For analog CO Lines, the System will await receipt of a valid ICLID for the ICLID Ring Timer. At expiration of the timer, if ICLID is not received, the call is routed based on the type and other programming (Ring assignments, etc.) CO/IP Line.
3. The ICLID received from the CO/IP Line must be a telephone number to match an ICLID Route Table entry.
4. If ICLID routing is enabled for a DID line, DID Call Wait is disabled.
5. Beside the System ICLID Table, each station can have up to 10 individual ICLID numbers.

Programming

- | | |
|---------------------|---|
| CO Line Data | 1. ICLID Service (PGM 165-Index7)
2. Alternative Ring Table (PGM 181)
3. CO Ring Assign Table (PGM 167) |
| Table | 1. ICLID Table (PGM 262) |

Related Features

Hardware

5.19 IP TRUNKING

5.19.1 H.323 v4 Service

Description

When assigned to support H.323 protocol, VoIP channels provide protocol conversion between H.323 v4 and SIP. This permits the VoIP channel to connect to external H.323 networks or terminals and to support H.323v4 supplementary services. In addition, H.323 VoIP channels can register with an external H.323 Gatekeeper to support Gatekeeper call routing.

Supplementary services support employing H.450.1 - H.450.12 standards, which define the following supplementary services:

- H.450.1
- H.450.2
- H.450.3
- H.450.4
- H.450.5
- H.450.6
- H.450.7
- H.450.8
- H.450.9
- H.450.10
- H.450.11
- H.450.12

Operation

System

Operation of H.323 Service is automatic.

Conditions

Programming

- | | |
|--------------|--|
| H.323 | 1. H.323 Call Setup (PGM 361) |
| SIP | 1. SIP User ID Data (Web Admin. PGM 370)
2. SIP CO Service (Web Admin. PGM 371) |

Related Features

System Networking

Hardware

VOIB

5.19.2 Session Initiation Protocol (SIP) Service

Description

When assigned to support Session Initiation Protocol (SIP), VoIP channels provide protocol conversion between SIP and H.323. This permits the VoIP channel to connect to external SIP Networks for call services. In addition, to the IETF RFC-3261 SIP draft standard, System's VoIP channels support other SIP related RFCs including:

- RFC-2617 – HTTP Authentication, Basic & Digest
- RFC-3515 – Refer Method
- RFC-3264 – Offer/Answer Model
- RFC-3265 – SIP Basic Call Flow Examples
- RFC-3891 – SIP “Replaces” Header

Using the SIP database assignments, the System will register and authenticate with the SIP Proxy Server permitting the System to interoperate employing SIP to establish, manage and terminate real-time voice sessions with external parties.

Operation

System

Operation of SIP Service is automatic.

Conditions

Programming

SIP

1. SIP User ID Data (Web Admin. PGM 370)
2. SIP CO Service (Web Admin. PGM 371)

Related Features

Hardware

VOIB

5.20 IP WAN DIALING AFTER ANSWER

Description

The System permits sending and receiving DTMF signals after connecting to an external VoIP party. The DTMF signal can be DTMF tone, Text String, or DTMF protocol (H.323 specification) based on System Programming.

Operation

System

To make IP Call:

1. Dial telephone number starting with digit in PGM 360; the system makes IP call to the assigned IP address.

To receive IP Call:

1. If system gets call from the assigned IP address in PGM 362; the system routes the call to the incoming CO group number in PGM 362.

Conditions

1. DTMF digits will not be recognized unless the connected VoIP party is transmitting.
2. DTMF mode can be assigned in CO Line Group base.
3. If system gets IP call from unassigned IP address in PGM 362, system ignores it.

Programming

H.323

1. H.323 Routing Attributes (PGM 360)
2. H.323 Call Setup (PGM 361)
3. H.323 Incoming Attributes (PGM 362)

Related Features

Hardware

5.21 INTEGRATED SERVICE DIGITAL NETWORK (ISDN)

Description

The System supports both Basic Rate Interface (BRI) and Primary Rate Interface (PRI) ISDN circuits.

5.21.1 ISDN Advice of Charge (AOC)

Description

When ISDN Advice of Charge (AOC) service is provided from the ISDN, the system will deliver charge information for display on the Digital Phone LCD and include the AOC in SMDR records. AOC is implemented in accordance with ETSI ISDN AOC Specifications.

Operation

System

ETSI standard AOC operation is automatic.

Conditions

1. AOC information implemented based on ETSI AOC standard, can be sent during call set-up (AOC-S), during the call (AOC-D) or at the end of call (AOC-E).
2. This feature may not be available in the specific ISDN service area or may be a subscription service.

Programming

- | | |
|------------------------|----------------------------------|
| System Data | 1. SMDR Attributes (PGM 232) |
| CO/IP Line Data | 1. Charge Mode (PGM 160-Index11) |

Related Features

- Station Message Detail Recording (SMDR)
- Call Cost Display

Hardware

PRIB, BRIB

5.21.2 Calling/Called Party Identification (CLIP/COLP)

Description

The System receives Calling Line Identification Presentation (CLIP) in the ISDN call Set-up Message. The Answering Party Identification, which may be different from the Called Party, is received in the ISDN Connect Message, is called Connected Line Identification Presentation (COLP). When provided, the Digital Phone LCD the identification, which is also included in call records. The received identification may be sent to a selected serial port.

Digital Phone Display



```
LINE XXX RINGING
03438502821
```

CLI Serial output

- 1) When CLI is entered through a normal CO Line or not dialed, the destination through DID line:
 - AAA:BBBBBBBB
- 2) When CLI is entered and the Station number is called through DID line:
 - AA A:BBBBBBBB : DDDDDDDDD(EEE) -> CCCCCCCC
- 3) When CLI is entered and Station answers the ring:
 - AAA:BBBBBBBB : DDDDDDDDD(EEE) -> FFFFFFFF

Where:

- AAA – ISDN CO line number
- BB...B – CLIP
- CCCCCCCC – Called Station
- DD...D – Speed Dial Name
- EEE – Speed Dial Bin Number (3 or 4)
- FFFFFFFF – Answering Station

The System will also compare the identification to the Speed Dial bins. If a match exists, the Name of the Speed Dial bin may be displayed in place of the number, CO/IP Name Display.

The System will send calling and answering party identification in the appropriate messages to the ISDN based on the database. Identification messages may be restricted, and not reported to the far-end user. Calling Line Identification Restriction and Connected Line Identification Restriction may be enabled in the System database.

Operation

Operation of this feature is automatic.

Conditions

1. This feature may not be available in the specific ISDN service area or may be a subscription service.

Programming

System Data 1. CLI Print (PGM 223-Index6)

Related Features

Hardware

5.21.3 Keypad Facility

Description

The ISDN Keypad Facility Information Element (IE) may enable the User to activate certain ISDN services (ex., Off-Net Forward). To access this facility, the Station must be enabled and have a Flex button programmed for {KEYPAD FACILITY}. When activated, the digits dialed by a User are sent in the Keypad Facility IE instead of the Called Party Number IE.

Operation

Digital Phone

To program a {KEYPAD FACILITY} button:

Press [PGM] + {FLEX} + Button Feature Type (1) + {Keypad Facility} feature code + [SAVE]

To activate the keypad facility after seizing an ISDN line:

1. Lift Handset, or press [Speaker] button.

2. Press the {KEYPAD FACILITY} button.
3. Dial desired digits, other actions will disable the Keypad facility feature.

Conditions

1. This feature can be activated or deactivated only after a CO Line (ISDN) is seized.
2. Once activated, the system will continue to send dialed digits as Keypad Facility IE messages regardless of ISDN messages; in the connected mode, DTMF tones are not sent to the connected party, only the keypad message is sent.
3. This feature may not be available in the specific ISDN or may be a subscription service.
4. The System can handle only a single Call Reference; services that require handling of two simultaneous Call References, cannot be supported.

Programming

- | | |
|-----------------------|---|
| Numbering Plan | 1. Keypad Facility Code (PGM 113) |
| Station Data | 1. Flex Button Assignment (PGM 126)
2. Keypad Facility usage (PGM 133-Index11) |

Related Features

Station Speed Dial

Hardware

Digital Phone
PRIB, BRIB

5.21.4 Multiple Subscriber Number (MSN)

Description

The Multiple Subscriber Number (MSN) feature enables multiple stations to receive ISDN incoming calls at a specific destination. Additionally, it enables a station to place an outgoing call using a specific CLI.

Operation

If set, MSN operation is automatic.

Conditions

1. If a CO line uses Representative CLI, that information is sent before other CLI options.

Programming

- | | |
|---------------------|--|
| Station Data | 1. Station DN Type (PGM 130 –Index1)
2. Station CLI Number (PGM 135 –Index6)
3. Station Flex Button Assign (PGM 126) |
| CO Line Data | 1. CO Digit Conv. Table Index (PGM 160 –Index6) |
| System Data | 1. Digit Conversion Table (PGM 251) |

Related Features

Hardware

PRIB/ BRIB

5.21.5 ISDN CLI

Description

When programmed, the ISDN will send CLI information on incoming and outgoing Calls. On Incoming Calls, CLI information is delivered to the Calling Party System. On Outgoing Calls, CLI information is delivered to the Called Party System, and also from the Called Party System to the iPECS-MG.

Operation

If set, ISDN CLI operation is automatic.

Conditions

1. If a CO line is using Representative CLI, that information is sent before other CLI options.
2. If a Station that places or answers the CO call has a CLI number programmed, only the CLI number of the station is sent as CLI information (similar to Long CLI option of ipLDK system).
3. If a CO line is not using Representative CLI and the Station is not using a programmed CLI number, then the {CO Own Code} + {Station number} is used as CLI information.
4. The Own Code of a CO Line is programmed separately to Incoming CLI and Outgoing CLI.
5. If the CLIP Display option at the Station is Disabled, then Caller CLI is not displayed on the Station LCD when an incoming call is received.
6. If the COLP Display option at the Station is Disabled, then CLI is not displayed on the Station LCD even after the Called Party answers the Station Outgoing Call.

Programming

Station Data

1. Station CLI Attributes (PGM 135)

CO Line Data

1. Own Code (Incoming) (PGM 165 –Index9)
2. Own Code (Outgoing) (PGM 170 –Index12)
3. Representative CID (Outgoing) (PGM 170 –Index11)

Related Features

Hardware

PRIB/ BRIB

5.22 ISDN SUPPLEMENTARY SERVICES

In many cases, the ISDN Service Provider will offer enhanced services available for User Subscription. The System allows access to these ISDN Supplementary Services implemented under the ETSI standard as described.

5.22.1 ISDN Call Deflection

Description

When ISDN Supplementary Service Call Deflection is supported, a User can Forward Incoming Calls on the ISDN Line directly through the ISDN without the need to establish a connection through the System.

When Call Deflection is used, the System sends a Call Deflection Request Message with the telephone number to the ISDN. The ISDN then sends Incoming Calls to the desired telephone number. In this case, the System does not set-up a CO Line-to-CO Line (Unsupervised Conference) connection for the call.

Operation

Digital Phone

To activate ISDN Call Deflection for an External Phone Number:

1. Lift the handset or press the **[SPEAKER]** button to receive dial tone.
2. Press the **[FWD]** button.
3. Dial Forward condition (1-4, or #)
4. Dial CO Access Code and desired External Phone Number.
5. Replace the handset, and return to idle.

To deactivate ISDN Call Deflection:

1. Press flashing **[FWD]** button, Call Forward will deactivate and the **[FWD]** button LED will be extinguished.

Conditions

1. The ISDN must support Call Deflection Supplementary Service as defined by the ETS300-202/206/207 Standard protocol.
2. ISDN lines that support Call Deflection must be assigned in the System database.

Programming

- | | |
|---------------------|-----------------------------|
| CO Line Data | 1. ISDN CD (PGM 161-Index9) |
|---------------------|-----------------------------|

Related Features

Hardware

- ISDN line
- Digital Phone

5.23 REPRESENTATIVE CLI SERVICE

Description

If a user makes a CO Call, the System can send Representative CLI instead of Individual CLI.

Operation

If set, Representative CLI Service operation is automatic.

Conditions

1. This feature can be set for each CO Line.
2. The maximum CLI length is 16 alphanumeric characters.
3. Representative CLI has the highest priority over other CLIs.

Programming

- | | |
|---------------------|---|
| CO Line Data | 1. Representative CLI Usage (PGM 170-Index10) |
| | 2. Representative CLI (PGM 170-Btn 11) |

Related Features

Hardware

6. DIGITAL PHONE

Following Digital phones and Optional Boards are available for use with the iPECS-MG 100 and 300 Systems.

- Digital Phones
 - LIP-6000 Series (LIP-6012D, 6030D, ...)
 - LIP-7000 Series (7024LD, 7024D, 7016D, 7008D)
 - LIP-8000 Series (8042L, 8024D, 8012D, 8004D)
 - LDP-6000 (6030D, ...)
 - LDP-7000 Series (7024LD, 7024D, 7016D, 7008D, 7004D, 7004N)
 - LKD Series (LKD-30D, LDK-8D, LKD-2NS)
- Optional Boards
 - BTU (LIP-8000 Series)
 - MU/FU/MFU/BTU/USB (LDP-7000 Series)
- DSS
 - DSS/LSS (LIP-8000 Series: LIP-8012DSS, LIP-8012LSS)
 - LKD-48DSS
 - LDP-7048DSS
- Door Phone
 - LDP-DPB

6.1 AUTO CALLED NUMBER REDIAL (ACNR)

Description

A Station User can request to have the System retry a Busy or No Answer External Call until the call is connected or the feature is cancelled.

Operation

System

If programmed, System operation for ACNR is automatic.

Digital Phone

To activate ACNR:

1. While receiving a Busy notification or No Answer, press the **[MSG/CALLBK]** button,
OR
2. Press the **{ACNR}** Soft button.
3. Hang-up handset, or press **[SPEAKER]**.

To cancel ACNR while phone is in Idle state:

1. Press the flashing **[MSG/CALLBK]** button.

To cancel ACNR during an ACNR attempt:

1. Lift the handset, press the **[MUTE]** button, or press the flashing **[MSG/CALLBK]** button

Conditions

1. The applicable Timers and Retry Counter must be programmed,
 - ACNR Pause Timer: Determines the time allowed between ACNR retries.
 - ACNR Retry Count: Determines the number of times the System will retry before ACNR is automatically cancelled.
2. The call will be placed on the same path as originally used; if the path is Busy, an available CO/IP line in the same group will be seized.
3. The ACNR Retry Counter decreases by one each time the System completes the Dialed Number.
4. When the ACNR Pause Timer expires, if the station is in a Busy state, the ACNR Delay Timer is invoked.
5. Upon completion of dialing, the System will monitor the call for progress signals.

Programming

System Data	1. ACNR Pause Timer (PGM 220-Index3)
Tenant Data	1. ACNR Retry Counter (PGM 280-Index4)
Station Data	1. ACNR Access (PGM 133-Index5)

Related Features

Last Number Redial (LNR)
Speakerphone
Mute

Hardware

Digital Phone

6.2 AUTO RELEASE OF [SPEAKER]

Description

After completion of certain features, the [SPEAKER] will turn OFF automatically, returning the Digital Phone to Idle.

Operation

System

Auto Release of [SPEAKER] operation is automatic for supported features (refer to Conditions).

Conditions

1. Auto Release of [SPEAKER] also applies to features including Call Park, Call Back, Call Forward and CO/IP Queuing.
2. If erroneous data is entered in Station Programming, an Error Tone is received and the User must correct the error before the station will return to Idle automatically.

Programming

Related Features

- Call Park
- Call Back
- Call Forward
- CO/IP Queuing

Hardware

- Digital Phone

6.3 AUTOMATIC SPEAKER SELECT

Description

Digital Phones programmed for Auto Speaker Select can access a CO/IP line or an Internal Call by pressing the appropriate button without the need to lift the handset or press the **[SPEAKER]** button. Audio from the CO/IP Line or Called Station is sent to the Speaker as if the User pressed the **[SPEAKER]** button and the Speakerphone MIC is activated.

Operation

Digital Phone

To access an Internal or External System Resource:

1. Press an assigned **{FLEX}** button for the appropriate resource; the **[SPEAKER]** will be activated.

Conditions

1. This feature does not apply to Digital Phones not equipped/assigned with Speakerphone; the User must lift the Handset.
2. Paging while on the Speakerphone may cause feedback from Paging Equipment; if Auto Speaker is enabled and a **{PAGE ZONE}** button is pressed, the display will show "LIFT THE HANDSET". To complete the page, User must lift the Handset within the predefined 5-second period or the phone will return to Idle.

Programming

- | | |
|---------------------|--|
| Station Data | 1. Auto Speaker Selection (PGM 121-Index1) |
|---------------------|--|

Related Features

Hardware

- Digital Phone

6.4 BACKGROUND MUSIC (BGM)

Description

A Digital Phone can receive audio (generally music), from an Internal or External Source while it is Idle. Music from the source is received over the Speaker and will be disabled during Ringing, Paging, or when the Station is Off-Hook.

Operation

Digital Phone

To listen to Background Music:

1. Press the **[PGM]** button and Dial 24.
2. Select the desired {BGM Source}.
3. Press **[SAVE]** button.

Conditions

1. Speaker volume can be adjustable at the Station using the **[VOL UP]/[VOL DOWN]** keys on the Digital Phone.
2. Silence is provided if no BGM Source is assigned.

Programming

Related Features

Music-On-Hold (MOH)
Internal/External & All Call Page

Hardware

BGM source properly connected to the MPB (refer to System Description and Installation Manual, Section 4.4.2)

6.5 CALL LOG DISPLAY

Description

Users of Digital Phones with Soft keys can view a Call Log of Incoming, Outgoing and Missed Calls on the Phone LCD (where applicable).

Operation

Digital Phone

To access the Call Log menu:

1. Press the **{LOG}** Soft button on the Digital Phone; a similar display will be presented.

```
>01 -> 107
02 M 106
SEND      SELECT      EXIT
```

Symbol	Description	Comment
→	Outgoing (Dialed) Call	
←	Incoming Answered (Received) Call	
M	Missed (Lost) Call	

2. Press the **[VOL UP]/[VOL DOWN]** buttons to select the Call Log to display.

```
01 -> 107
>02 M 106
SEND      SELECT      EXIT
```

3. Press the **{SEND}** button to make a call, press **{SELECT}** to check the detailed information, or press **{EXIT}** to return to Main Menu selections.

Conditions

1. One station can have up to 100 logged calls.
2. Call Log is displayed as one of three types of calls: Received Call, Dialed Call and Lost Call.
3. The Call Log is stored in volatile memory; it is not protected in case of power failure.

Related Features

Hardware

Digital Phones with 3 Soft keys

6.6 CO LINE NAME DISPLAY

Description

When a CO/IP Call is received or a User seizes a CO/IP Path, the CO/IP Number is displayed on the LCD. If the CO/IP Path is assigned a Name and CO/IP Name Display is enabled, the CO/IP name will be displayed instead of the CO/IP Number.

Operation

Digital Phone

To display the CO/IP Line Name while calling:

1. Dial CO/IP Access code, LCD displays CO/IP Line Name.

Conditions

1. Each CO Line and IP Group can be assigned a Name of up to 12 characters in the System Database.

Programming

- | | |
|---------------------|--|
| CO Line Data | 1. Incoming CO Line Name (PGM 165-Index1) |
| | 2. CO Line Access Code Name (PGM 180-Index1) |

Related Features

Hardware

Digital Phone with LCD

6.7 ONE TIME DND

Description

While a Station is ringing or receiving an Off-Hook Muted Ring, the User can press the **[DND]** button, to reject the call and terminate Ringing. When the Station returns to the Idle status, DND is automatically cancelled and the **[DND]** LED is extinguished.

If the DND button is pressed while on a call, any ringing to the Station regardless of destination DN, will be rejected and will not ring.

Operation

Digital Phone

To activate One Time DND while on a call

1. Press the **[DND]** button, the **[DND]** LED lights, station goes to DND state.

System

Deactivation

1. When the Station returns to Idle, DND automatically is cancelled and the **[DND]** LED will extinguish.

Conditions

1. If the **[DND]** button is pressed while on an active call, subsequent calls will not be received for the duration of that call.
2. One Time DND operates regardless of DND Access privilege.
3. If the **[DND]** button is pressed when Delayed Ring Service is enabled, the Station will continue to receive Incoming Call Ringing.
4. CO/IP recalls will override One Time DND.

Programming

Station Data

1. DND Access (PGM 132-Index4)

Related Features

Do Not Disturb (DND)

Hardware

Digital Phone

6.8 GROUP LISTENING

Description

All Digital Phones have a built-in speaker. If enabled, the User may use the speaker to monitor a call while using the Handset to converse with an Outside party. This enables a group of people in the room to listen to both parties in the conversation.

Operation

Digital Phone

While on a call using the Handset:

1. Press the **[SPEAKER]** button; the Speaker will activate, and the Speakerphone Microphone will be Muted while the Handset is Off-Hook.

Conditions

1. While using the Speakerphone, lifting the handset will turn-off the Speakerphone; to activate Group Listening, the **[SPEAKER]** button must be pressed while the Handset is Off-Hook.
2. While in Group Listening Mode, pressing the **[MUTE]** button will cause the TX path from the Handset to be Muted; the distant-end will still be heard over both the handset receiver and the Station Speaker.
3. If full Speakerphone operation is desired and available while in Group Listening Mode simply replace the Handset to go On-Hook.
4. Digital Phones without a built-in speaker (LIP8004D/LDP-7004/LDK-02N/etc.), are not able to use this feature.

Programming

- | | |
|---------------------|----------------------------------|
| Station Data | 1. Group Listen (PGM 121-Index4) |
|---------------------|----------------------------------|

Related Features

Speakerphone
Mute

Hardware

Digital Phone

6.9 INTERCOM SIGNALING MODE

Description

Each Digital Phone can select the applicable signaling mode used for incoming ICM calls while the station is Idle; there are three signaling modes available:

- **Call Announce with Hands free Answer-Back (H)** – When an ICM call is received, the User receives a splash tone followed by the ICM caller's voice. The user may respond to the Caller without the need to Lift the Handset or press the **[SPEAKER]** button.
- **Call Announce with Privacy (P)** – When an ICM call is received, the User will receive a Splash tone followed by the ICM Caller's voice; the User must lift the Handset or press the **[SPEAKER]** button to connect the call.
- **Tone Ring (T)** – An ICM call will cause the Digital Phone to provide an audible ICM ring tone; the user must lift the handset or press **[SPEAKER]** to answer.

NOTE: Digital Phones use Tone Ring Mode as a default, and SLTs always function in the Tone Ring Mode.

Operation

Digital Phone

To change ICM Signaling Mode:

1. Press the **[PGM]** button; the **[SPEAKER]** button LED will light steady.
2. Dial 11 {Station User Program code}, and a confirmation tone is received.
3. Dial the desired ICM Signaling Mode code (1=H,2=T, or 3=IP).
4. Press the **[SAVE]** button.

Conditions

1. Callback and Call Forward will ring in the tone mode, regardless of ICM Signaling Mode selected by the User.
2. The ICM Signaling Mode Selection does not affect Page announcements.
3. By default, the ICM Signaling Mode is Tone ring, and is stored in battery-protected memory.

Programming

- | | |
|--------|--|
| Tenant | 1. Intercom Busy One-Digit Service (PGM 237) |
|--------|--|

Related Features

Intercom Call (ICM Call)
Paging

Hardware

Digital Phone

6.10 MUTE

Description

A Digital Phone can turn off audio transmission from the Handset, Speakerphone or Headset Microphone (Mic Mute).

Operation

Digital Phone

To Mute the Microphone:

1. While on a call, press the **[MUTE]** button; the **[MUTE]** button LED will illuminate and the microphone (Handset, Speakerphone, Headset) will be muted, disabling audio transmission to the other party.

To activate the Microphone:

1. Press the illuminated **[MUTE]** button; the **[MUTE]** button LED will be extinguished, and the microphone is activated, transmitting audio to the connected party.

Conditions

1. Changing from Speakerphone to Handset or vice versa while the phone is Muted will re-activate the phone microphone, transmitting audio.
2. Returning to Idle or placing another CO/IP or Intercom call will cancel Mute, and re-establish audio transmission from the phone.

Programming

- | | |
|---------------------|----------------------------------|
| Station Data | 1. Headset Ring (PGM 121-Index3) |
|---------------------|----------------------------------|

Related Features

Speakerphone
Group Listening

6.11 OFF-HOOK SIGNALING

Description

This Feature is based on the DN Feature. When an Off-Hook Station receives a call or a CO/IP Call rings into the System for the Off-Hook Station, the Station will receive the assigned Off-Hook Ring signal for Intercom Calls, Camp-On, Call Wait, or Voice-Over Announcements. For Calls with a different DN, Off-Hook Ring signaling may be received with LED notification.

Off-Hook Ring signaling delivered to the Digital Phone Speaker either may be a Muted Normal Ring signal or a Single Burst Tone based on the System assignment.

Operation

System

If set, Off-Hook Ring signal operation is automatic.

Conditions

1. While using the Speakerphone, a Camp-On tone is provided over the Speaker in place of the assigned Off-Hook Ring signal.
2. Activating the DND or One-Time DND places the Station in DND, terminating any Off-Hook signaling.
3. Off-Hook Ring signals terminate when the call is Answered, Forwarded, or Abandoned.
4. The Station will receive Normal Ring signals following return to Idle status.

Programming

Station Data

1. DN Member (PGM 130-Index2)
2. Camp On Access (PGM 133-Index8)
3. Call Wait Access (PGM 133-Index7)

Related Features

Hardware

Digital Phone

6.12 ON-HOOK DIALING

Description

Digital Phones equipped with a Speakerphone can allow Users to place as well as receive calls while the Handset is On-Hook. Once the User activates the Speakerphone by pressing the **[SPEAKER]** button or Automatic Speaker Select, Dial tone is received and the User may dial the desired number.

Operation

Digital Phone

To activate On-Hook Dialing:

1. Press the **[SPEAKER]** button, and Dial tone is received; the **[SPEAKER]** button LED will illuminate.
2. Dial the desired number (ICM number, or select CO/IP path and dial).

Conditions

1. If the Outgoing call is not Answered, the User must press the illuminated **[SPEAKER]** button to return to Idle.
2. When the Speakerphone is used, the Microphone is active unless the **[MUTE]** button is pressed, and the **[MUTE]** button LED is On.

Programming

- | | |
|---------------------|--|
| Station Data | 1. Auto Speaker Selection (PGM 121-Index1) |
|---------------------|--|

Related Features

Mute
Speakerphone
Automatic Speaker Select

6.13 PRIME LINE IMMEDIATELY/DELAYED

Description

When a User Station goes Off-Hook, the System normally provides the ICM dial tone. If desired, a Station can be assigned to access a pre-selected Prime Line. The Prime Line can be set for:

- Seizing a CO/IP Line
- Calling another Station
- Feature Code (as selected)

Prime Line access can be defined as:

- Immediate – When phone goes Off-Hook, the System will provide access to the Prime Line.
- Delayed – The Station User will receive the normal Intercom Dial tone for the designated Prime Line Delay Timer and will access the Prime Line following Timer expiration.

Operation

Digital Phone

To access the Station Prime Line:

1. Lift the Handset or press the **[SPEAKER]** button; take no action (wait), Prime Line will be accessed as designated.

Conditions

1. Prime Line is based on DN.
2. When the User lifts the Handset or presses the **[SPEAKER]** button, the System will act as if the User pressed the pre-defined digits prior to going Off-Hook.
3. When Delayed Prime Line is set, the User must wait, taking no action until the Prime Line is accessed; the user receives the ICM Dial tone during this period, and may dial any valid numbering plan digit(s) or select a Flex button or Feature button.
4. If the Prime Line Delay Timer is greater than Dial Tone timer, Delayed Prime Line will not activate; it will be necessary to reduce the Delayed Prime Line Timer or extend the Dial Tone timer.
5. If Prime Line is set without delay, the Station can only place a call; to make any other action such as Call Transfer, or Conference Call, the Auto Dial Pause Timer must not be set to 0.

Programming

Station Data

1. Auto Dial Digit (PGM 138-Index1)
2. Auto Dial Pause Time (PGM 138-Index2)

Related Features

Speakerphone
Intercom Call (ICM Call)
Station Flexible Buttons

Hardware

6.14 DIFFERENTIAL RING

Description

The User can select one of 14 Ring tones so that the iPECS Phone ring can be distinguished from other nearby phones. Up to 8 Ring tones can be stored in the iPECS Phone permanent memory; the first 4 tones are fixed, and the other Ring tones can be downloaded from a library of 10 Ring tones stored in the System protected memory.

After downloading a tone from System memory, it can be selected as the Differential Ring Tone.

Operation

iPECS Phone

To download a Ring Tone from System memory:

1. Press the **[PGM]** button.
2. Dial 23 {Ring Tone Download code}.
3. Select the desired Ring tone location (5-8).
4. Select the Ring source using the **[VOL UP]/[VOL DOWN]** keys and press the **[SELECT]** button or dial the appropriate ring number to select.
5. While listening to music,
 - Dial 1 to save the selected music source
 - Dial 2 to return previous step.

To select the downloaded Ring Tone:

1. Press the **[PGM]** button.
2. Dial 21 (Intercom Ring tones), or 22 (CO/IP Ring tones).
3. Press the **[OK]** soft button or fixed button.
4. Dial the desired Ring tone digit (5-8).
5. Press the **[SAVE]** button.

Conditions

1. The downloadable Ring tone files are stored in System memory as *.wav files with a maximum length of 4 seconds. These files can be replaced as desired using the Web Upload function.
2. Ring tone download is only supported at LIP-series terminals.

Programming

Related Features

Differential Ring

Hardware

6.15 SAVED NUMBER REDIAL (SNR)

Description

The Last Dialed number on a CO/IP Call may be stored (up to 24 digits) in a buffer for future Redial. This number is saved in memory until the User stores a new number. Numbers dialed for subsequent calls do not affect the Save Number Redial (SNR) buffer.

Operation

Digital Phone

To save a Dialed number, while on a CO/IP Call:

1. After dialing, but before hanging up, press the **[SPEED]** button twice; the dialed number will be stored in the SNR buffer.

To dial a Saved number:

1. Lift the handset or press the **[SPEAKER]** button.
2. Press the **[SPEED]** button.
3. Dial #.

Conditions

1. The SNR can be a maximum of 24 digits.
2. Dialing the Saved Number will automatically seize the CO/IP line that was used for the original call; if the CO/IP Line is Busy, a CO/IP Line from the same group will be selected and the saved number dialed. If all CO/IP lines from the group are busy, the user will receive All Lines Busy tone and may select to Queue the call.
3. Dialing the SNR will automatically seize the DN that was used for the original call. If the DN is used by another Station, automatically Prime DN will be selected and saved.
4. SNR is saved in permanent memory, protected from power failure.
5. Manually dialing a Flash during a CO call will cause only those digits after the Flash to be stored and re-dialed as the SNR.

Programming

Related Features

Station Speed Dial
System Speed Dial
Last Number Redial (LNR)

Hardware

Digital Phone

6.16 SPEAKERPHONE

Description

Digital Phones equipped with a Speakerphone can use the telephone hands-free in two-way conversations.

Operation

Digital Phone

To activate the Speakerphone:

1. While on a Call or when placing a Call, press the **[SPEAKER]** button; the LED will illuminate.

To switch from Handset to Speakerphone:

1. While on a Call, press the **[SPEAKER]** button; the LED will illuminate.
2. Replace Handset, the Speakerphone is activated.

To terminate a Speakerphone call:

1. While on a Call using the Speakerphone, press the **[SPEAKER]** button; the LED will extinguish.

Conditions

1. If Automatic Speaker Select is enabled at the Station, pressing a DSS, DN, CO Line Access Code or Speed Dial button will automatically activate the Speakerphone.
2. The **[MUTE]** button LED indicates the status of the Microphone, when lit the Microphone is inactive.
3. When Group Listen is enabled, pressing the **[SPEAKER]** button while using the handset will send audio to both the Handset and Speaker. However, only the Handset microphone will be active; in order to activate the Speakerphone Microphone, the Handset must be On-Hook.
4. Each Digital Phone equipped with Speakerphone is allowed/denied Speakerphone operation based on System Database Admin. Programming.
5. When Headset operation is assigned for the Station, the Speakerphone is disabled and the **[SPEAKER]** button activates the Headset audio path instead of the Speaker.

Programming

Station Data

1. Headset Ring (PGM 121-Index3)
2. Headset Mode (PGM 121-Index2)

Related Features

Mute
Group Listening
Automatic Speaker Select
Green Power Save

6.17 STATION FLEXIBLE BUTTONS

Description

The Digital Phone incorporates a field of Flex buttons as well as the Fixed Feature buttons. The Flex buttons are assigned in the System database to access features, functions and resources of the System. Specifically, Flex buttons can be assigned as:

- Empty button – No system database assignment.
- **{DSS}** button – used to place One-touch ICM calls to a designated Station and display Station status.
- A Flex Numbering Plan, button activates the feature associated with the assigned digits from the Flexible Numbering Plan.
- A DN Flex button accesses and dials the assigned number.
- An External Telephone Number including CO Access Code to make external call.
- A CO Line Appearance button provides access to the individual CO Line assigned to the Flex button. The CO Line button LED provides the status of the CO Line.
- Fixed button activates the feature associated with the fixed button

With the exception of CO Line buttons, Flex buttons can be assigned at the station by the end-user. If allowed in the database, the user can also assign or reassign CO Line buttons.

Operation

Digital Phone

To assign a Flex button at the station:

1. Press the **[PGM]** button.
2. Press the desired Flex button.
3. Select the type.
 - 1: Telephone number or Feature code.
 - 2: Fixed Button.
 - 0: Delete.
4. Select the desired button type using the **[VOL UP]/[VOL DOWN]** key to assign a Fixed button,
OR
5. Dial the desired number to create the Telephone number button.
6. Press the **[SAVE]** button.

To change the ring option of DN button:

1. Press the **[PGM]** button.
2. Press the desired DN Flex button.
3. Dial '1' or '0'.
 - 1: Immediate Ring
 - 0: No Ring
4. Press the **[SAVE]** button.

Conditions

1. Fixed button programming is supported when the Phone does not already have the same Fixed button.
2. If a Fixed button or Dial Number button is set to not allow User modification according to Admin. Programming, the User cannot change the button.
3. DN button can be assigned or changed by Admin. Programming. The station can only change the ring option of DN button.

Programming

Numbering Plan 1. Flexible Numbering Plan (PGM 112-115)

Station Data 1. Station Flexible Button Assignment (PGM 126)

Related Features

Flexible Numbering Plan

Hardware

Digital Phone

6.18 STATION FLEXIBLE LED FLASH RATES

Description

The Flash Rates used with the various Flex buttons on the Digital Phone can be adjusted on a System-wide basis according to Customer needs. Up to 48 different functions can be assigned from of 15 different Flash Rates.

Operation

System

The System implements Flash Rates automatically based on Database entries.

Conditions

1. Available Flash rates and functions, which can be assigned, are given in the Admin & Programming Manual.

Programming

System Data 1. LED Data (PGM 234)

Station Data 1. Station Flexible Button (PGM 126)

Related Features

Hardware

Digital Phone

6.19 STATION ICLID CALL ROUTING

Description

The Station can employ Incoming Calling Line ID (ICLID) to determine appropriate routing for Incoming External Calls.

Operation

Digital Phone

To assign ICLID at the Station:

1. Press **[PGM]** button and Dial 71, then the empty ICLID bin automatically will be allocated.
2. Enter ICLID and press **[SAVE]** button.
3. Enter the Routing Destination, and press the **[SAVE]** button.

NOTE: Routing Destination can be a Station number, Hunt Group number or VMIB Forward code.

To view the assigned ICLID at the Station:

1. Press **[PGM]** button and dial 72; ICLID information is displayed.
2. Press **[VOLUME UP/DOWN]** to view the ICLID List.

To delete the assigned ICLID on the Station:

1. Press **[PGM]** button and Dial 72; ICLID information is displayed
2. Select the ICLID bin with **[VOL UP]/[VOL DOWN]** and press **[SAVE]** button.
3. Press **[SPEED]** button; confirmation tone will be presented.
4. Dial 1 (Delete).

Conditions

1. If the received ICLID does not match an entry in the Station ICLID List, the Station will receive CO/IP Line Ringing.
2. This feature applies to all digital and analog CO Lines.
3. The ICLID received from the CO/IP Line must be a telephone number to match a station ICLID List.
4. Each Station can have 10 ICLID numbers individually.

Programming

Related Features

Hardware

6.20 STATION USER PROGRAMMING & CODES

Description

Users can program an array of Functions and Features, and Access Status information. The Station User Program Codes used for these purposes are fixed as listed below.

User Program Code Chart

User PGM Code	Description	Remark
11	Intercom Answer Mode	1:H, 2:T, 3:P
12 + Name	User name creation	2 digit for each character
13 + Time	Set wake-up alarm time	HH/mm, 24-hour clock
14	Cancel Wake-up Alarm	
15	Set language for the display	00-14
16	LCD Date Mode Change	DD/MM/YY or MMDDYY
17	LCD Time Mode Change	12 Hour/24 Hour
18	Set Backlight	0-3
21	ICM Ring Type	
22	TRK Ring Type	
23	Ring Download	LIP-Series Only
24	Back Ground Music	
31	Temporary COS	Auth. Code required
32	Retrieve COS	Auth. Code required
33	COS Override (Walking COS)	Auth. Code required
34	Register Password	
35	Call Log Protect	
36	SMS Message Protect	LIP-Series/LDP6000-Series
41 + MSG number [xx]	Set Pre-defined Message.	0-9, MSG *: User Custom # Deactivation
42	Create a Station User Message	
43	Send SMS Message	LIP Series/LDP6000 Series
44	Receive SMS Message	LIP Series/LDP6000 Series
51 + x	Activate a mobile phone	X=1-2
52 + x	Register the mobile number	X=1-2
53 + x	Register the mobile CLI number	X=1-2
54 + Rm & Auth Code	Start a Conf Room	
55 + Rm & Auth Code	Close a Conf Room	
61	Speaker/Headset Mode	Speak/Headset/E-MIC
62	Headset Ring Mode	Speaker/Headset/Both
71	Register Station ICLID	
72	View Station ICLID	
81	View IP Address	IP Phone/ DTIM/SLTM
82	View Mac Address	IP Phone/ DTIM/SLTM
83	View IP Phone version	
80	Network Setting	LIP Series
91	System Version	
92	System IP Address	

DECT Program Code Chart

User PGM Code	Description	Remark
12 + Name	User name creation	2 digit for each character
13 + time	Set wake-up alarm time	HH/mm, 24-hour clock
14	Cancel Wake-up Alarm	
15	Set language for the display	00-14
31	Temporary COS	Auth Code
32	Retrieve COS	Auth Code
33	COS Override (Walking COS)	Auth Code
34	Register Password	
41 + MSG number [xx]	Set Pre-defined Message	0-9, MSG *: User Custom # Deactivation
42	Create a Station User Message	
51 + x	Activate a mobile phone	x= 1 –2
52 + x	Register the mobile number	x= 1 –2
53 + x	Register the mobile CLI number	X= 1-2
54 + Rm & Auth Code	Start a Conf Room	
55 + Rm & Auth Code	Close a Conf Room	
71	Register Station ICLID	
72	View Station ICLID	
91	System Version	
92	System IP Address	

Additionally, a Station User Program Menu display is provided on the Phone display to assist the User in setting the Station User Program Code Features and Functions. The **[VOL UP]/[VOL DOWN]** buttons are used to scroll through the Menu items and the Dial Pad is used to enter a selection.

Operation

Digital Phone

To activate a Station User Program Code Feature or Function:

1. Press the **[PGM]** button, the Station User Program Menu is displayed.
 2. Use the **[VOL UP]/[VOL DOWN]** to display the desired menu item.
- OR
3. Dial the desired Station User Program Code and additional inputs as required

Conditions

Programming

Related Features

Hardware

6.21 TWO-WAY RECORD

Description

A Digital Phone User can record any active conversation to the Station User's Internal/External Mailbox or to a Phontage Hard Disk drive or UCS Client (where applicable). A **{RECORD}** button must be assigned to access this feature and record Normal Incoming/Outgoing Calls.

Operation

Digital Phone

To assign a flexible button as a **{RECORD}** button:

Press **[PGM]** + **{FLEX}** + Button Feature Type(1) + {Two-Way Record Feature Code} + **[SAVE]**

To activate Two-Way Record:

1. While on an active CO/IP Call, press the **{RECORD}** button.
2. The Record tone will be heard, prompting the Parties on the Call that recording has begun.

NOTE: To manage recordings, follow the procedures outlined in the **Phontage** or **UCS Client User Guide**.

To stop Two-Way Record:

1. While on an active CO/IP Call that is being recorded by the Station, press the **{RECORD}** button again.
- OR
2. Hang-up, and return to idle.

Conditions

1. The **{RECORD}** button LED will flash at 120 ipm while recording.
2. Two-Way Recording feature is a manual recording, while the feature Call Recording is an automatic recording method.
3. This feature is available when using the VMIB, Feature Server, or an external AA/VM, using SMDI communications mode. When an external AA/VM system uses in-band (DTMF) mode, Two-Way Record is not available.
4. Using Feature Server or VMIB, Internal Calls can be recorded as well as External Calls.
5. If a Destination for recording is not defined for the **{RECORD}** button, the Call Record Destination defined is employed.
6. Conference Call is not supported for more than 2-parties.
7. When recording starts, the user hears two-way record warning tone. This tone is defined tenant tone table at 73 [Two-Way Record Warning Tone].

Programming

- | | |
|----------------|--|
| Station | <ol style="list-style-type: none">1. Auto Record Service (PGM 145-Btn 3)2. Two-way Record Access (PGM 145-Btn 4)3. Two-way Record Device (PGM 145-Btn 5) |
| System | <ol style="list-style-type: none">1. Tone Table (PGM290) |

Related Features

Hardware

Digital Phone
VMIB, Feature Server or External SMDI based AA/VM system
Phontage or UCS Client

6.22 ANSWERING MACHINE EMULATION (AME)

Description

When internal or external calls are forwarded to VMIB, the user can check and hear saved messages and also can record the call, using the {Mute} button. AME feature has LED Blinking mode and Speaker Hearing mode. User can program an {AME} button to a flex button, and can assign AME mode with this button.

- LED Blinking Mode—When a call is forwarded to VMIB, the {AME} LED button will blink as notification; the user can hear saved voice conversations by pressing the {AME} button.
- Speaker Hearing Mode—When a call is forwarded to VMIB, the speaker automatically will be open to hear the calling party. The user can listen to the call, save it, or pick-up the call by pressing the {AME} button or {Mute} button.

NOTE-

If user presses the {AME} button, the call will be connected without saving the message being left. If user presses the {Mute} button, the conversation will start and it will be saved at the same time.

Operation

Digital Phone

To assign a flexible button as {AME}:

[PGM] + {FLEX} + Button Feature Type(1) + {AME Access Feature Code} + [SAVE]

To assign LED Blinking Mode:

1. Press the {AME} button.
2. Press 1 (LED Blinking Mode).

To assign Speaker Hearing Mode:

1. Press the {AME} button.
2. Press 2 (Speaker Hearing Mode)

To Delete AME Feature:

1. Press the {AME} button.
2. Press 0 (Disable)

To Answer a call in LED Blinking Mode:

1. When a call has been forwarded to VMIB, the {AME} Button will blink.
2. Press {AME} button to go to Speaker Hearing mode.
3. Press {AME} button to communicate without saving conversation,
OR
4. Press {Mute} Button to communicate with saving conversation

To Answer a call in Speaker Hearing Mode:

1. When a call is forwarded to VMIB, press the {AME} button to communicate without saving conversation,
OR

2. Press {Mute} Button to communicate while saving the conversation.

Conditions

1. VMIB Access feature has to be ON (enabled)
2. AME Function will be start when VMIB Forward feature is set.

Programming

- | | |
|---------|---------------------------------|
| Station | 1. VMIB Access (PGM 145-Index1) |
|---------|---------------------------------|

Related Features

Hardware

Digital Phone
VMIB, Feature Server or External SMDI based AA/VM system
Phontage or UCS Client

6.23 VOICE OVER

Description

This feature allows Digital Phone Users, to receive a voice announcement through the handset receiver while on an existing call (CO, IP or Intercom). The Voice Over (OHVO) will be muted to not interfere with the existing conversation. The Called Station User may respond to the Calling Party using Call Wait.

Operation

Digital Phone

Placing a Voice Over (OHVO) while receiving a Busy signal:

1. Dial the {Voice-Over code}, selected in Tenant Intercom Busy One-Digit service.
OR
2. Press a pre-programmed {OHVO} button.
3. After the splash tone is heard, begin announcement.

Responding to a Voice Over announcement:

1. While on the active call, press the [HOLD] button, activating the Call Wait feature.

Conditions

1. The Receiving Station must be programmed to allow OHVO calls.

Programming

- | | |
|--------------|--|
| Station Data | 1. Voice Over Access (PGM 133-Index9) |
| System data | 1. Intercom Busy One-Digit Service (PGM 237) |

Related Features

Hardware

Digital Phone

7. ATTENDANTS

7.1 ATTENDANT GROUP

Description

The System can have an Attendant Group (up to 5 Attendants) per Tenant. Each Attendant position must be equipped with a multi-button Phone and may include multiple DSS Consoles.

There are 2 different destinations to cover Attendant duties in a Tenant as follows:

- **Night Attendant Group** – Can be assigned as Hunt Group and covers Attendant Calls when all Attendants in a Tenant are in **{DND}**.status or the System is in Night Ring mode.
- **Forward Destination** – This can be assigned as Station/Hunt Group/Telephone Number and it covers Attendant Call according to the Forward type of the Attendant group.

Operation

If set, Attendant Group operation is automatic.

Condition

1. The first Attendant (System Attendant) is assigned as Station 100 (default), and others are not assigned.
2. Attendant Calls (using the Attendant Call Code) is routed to first available Attendant according to Attendant Group Type (Terminal/Circular/Ring/Longest idle).
3. A member in Attendant Group can use all DN features.
4. If a member in the Attendant Group sets DND/FWD/Preselected Msg., Attendant Call will not be received.
5. LIP/LKD/LDP series set for SADN-type can be assigned as the member of an Attendant Group.
6. The Tenant of an Attendant member should be the same as Attendant Group.

Programming

Tenant Data

1. Attendant Group Assignment (PGM 270)
2. Attendant Group Greeting/Queuing Tone (PGM 271)
3. Attendant Group Attribute (PGM 272)
4. Night Attendant Group Assignment (PGM 275)
5. Night Attendant Group Greeting/Queuing Tone (PGM 276)
6. Night Attendant Group Attribute (PGM 277)

Numbering Plan

1. Feature Numbering Plan, Attendant Call (PGM 113)

Related Features

Hardware

Digital Phone

7.2 NIGHT ATTENDANT GROUP

Description

This feature allows an Alternate Answer Point while the Attendant station is in an unavailable mode. When in the unavailable mode, the next available Attendant in a Tenant will receive Attendant calls.

If All Attendants in a Tenant activates Night Attendant or the System is in Night mode, Calls will be routed to the Night Attendant Group.

Operation

If set, Night Attendant Group operation is automatic.

Conditions

1. Night Attendant activates when all members of an Attendant Group activate DND; the Night Group will receive Attendant Group Calls until Attendant Member Station is available.
2. Night Attendant activates when Ring mode is changed to Night mode.
3. LIP/LKD/LDP series Phones set to SADN-type can be assigned as a member of a Night Attendant Group.

Programming

Tenant Data

1. Night Attendant Group Assignment (PGM 275)
2. Night Attendant Group Greeting/Queuing Tone (PGM 276)
3. Night Attendant Group Attribute (PGM 277)

Related Features

Hardware

7.3 GREETING/QUEUING TONE SERVICE

Description

The System can provide a Greeting tone or Queuing tone when a call is routed to an Attendant Group or Night Attendant Group.

When a call is routed to the Attendant/Night Attendant Group, the pre-assigned Greeting or Queuing Tone will be provided to the caller. The tone will be provided according to the Tone Time/Delay Time/Repeat counters. There are 5 types of Tones:

- **NORMAL** – System Tone (01-19, Tone Freq. in PGM 264)
- **PROMPT** – VMIB Prompt
- **ANNOUNCEMENT** – VMIB Announcement
- **INT MOH**
- **EXT MOH**

Operation

If set, Greeting/Queuing Tone Service operation is automatic.

Conditions

Programming

Tenant Data	1. Attendant Group Attribute (PGM 271-272)
Table Data	1. Tone Frequency/Cadence (PGM 264)

Related Features

Hardware

7.4 FORWARD DESTINATION, OVERFLOW SERVICE

Description

This can be assigned as Station/Hunt Group/Telephone Number, covering Attendant Call according to the Forward type of the Attendant Group. There are 4 kinds of Forward type in an Attendant Group:

- Unconditional
- Queuing Overflow
- Queuing Timeout
- Queuing Overflow or Queuing Timeout

The Overflow Destination can be programmed as Station/Station Group/External number/NET Destination.

Operation

To use the Unconditional Forward Overflow Destination:

1. Dial the {Attendant Call code}.
2. The Call is Routed to the Forward Destination.

To use the Queuing Overflow Forward Destination:

1. Dial the {Attendant call code}.
2. The Call is Queued when all Member Stations are in Busy mode.

NOTE: The Call will be Routed to the Forward Destination when max. queue has been Overflowed.

To use Queuing Timeout Forward Destination:

1. Dial the {Attendant Call code}.
2. The call is Queued when all Member Stations are in Busy mode.

NOTE: Calls will be Routed to the Forward Destination when Queuing Time has expired.

To use Queuing Overflow or Timeout as Forward Destination:

1. Dial the {Attendant Call code}.
2. The Call is Queued when all Member Stations are in Busy mode.

NOTE: The Call will be routed to the Forward destination when Queuing Time expires or Max. Queue is overflowed.

Conditions

Programming

- Tenant Data** 1. Attendant Group Attribute (PGM 271-272)

Related Features

Hardware

7.5 ATTENDANT RECALL

Description

Unanswered or Abandoned CO/IP Calls that remain unanswered for the Hold or Transfer Hold Timer (as applicable), will Recall at the Station that placed the Call on Hold. If the Call remains unanswered for the assigned Recall Time, the first available Attendant will also receive the Recall. The Attendant and Station will simultaneously receive the recall signal for the Attendant Recall Timer period after which the System will disconnect the Call and return the CO/IP Line to Idle.

Operation

System

Attendant Recall operation is automatic.

Conditions

Programming

- CO Line Data** 1. Incoming CO Alternate (PGM 169)
2. Outgoing CO Alternate (PGM 173)
- Tenant Data** 1. Attendant Group Assign (PGM 270)

Related Features

Hold
Call Transfer

Hardware

7.6 ATTENDANT STATION PROGRAM CODES

Description

Using the Attendant Station Program Codes, the Attendant can Print SMDR and Traffic Reports on-demand, Assign Authorization Codes, Control certain User Features, Record VMIB Announcements, etc. Items are available using the Program Code directly or scrolling the multi-level display menu using a Digital Telephone LCD screen.

The following Table indicates the Display Menu, including digits for selecting the Item, the Item Description and any further required entries. The various levels of the Display Menu are indicated by indentation.

NOTE: Some Program Codes are only available to the System Attendant or Stations allowed access to Attendant Feature codes.

Example of User Program Code Chart

User PGM Code	Item Description	Remark
01 SMDR		
011	PRINT STATION SMDR	Station Range
012	DELETE STATION SMDR	Station Range
013	PRINT FAILED CALL SMDR	
014	DELETE FAILED CALL SMDR	
015	DELETE ALL SMDR	
016	ABORT PRINTING	
02 TRAFFIC		
021	PRINT TRAFFIC (TENANT)	
022	PRINT TRAFFIC (CALL TYPE)	
023	PRINT TRAFFIC (CO GRP)	
03 COS / PASSWORD		
031	TEMPORARY COS MODE	Station Range
032	RETRIEVE COS	Station Range
033	REGISTER PASSWORD	Station Range
034	CALL LOG PROTECT	Station Range
04 DATE / TIME		
041	SET SYSTEM DATE	
042	SET SYSTEM TIME	
043	LCD DATE MODE	Station Range
044	LCD TIME MODE	Station Range
045	SET WAKE UP	Station Range
046	RESET WAKE UP	Station Range
05 MULTI MESSAGE		
051	PRESELECTED MESSAGE	Station Range, MSG No
052	SET USER MESSAGE	Station Range

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User PGM Code	Item Description	Remark
06 VMIB ANNOUNCEMENT		
061	LISTEN VM ANNOUCEMENT	
062	RECORD VM ANNOUCEMENT	
07 USER PROGRAM		
071	STATION NAME	Station Range
072	LANGUAGE PROGRAM	Station Range
073	PREPAID CALL	Station Range
074	FEATURE CANCEL	Station Range
08 SYSTEM		
081	DAY/NIGHT PROGRAM	
082	MONITOR CONF ROOM	
083	FORCED DELETE CONF ROOM	
084	PPTP CONNECTION	Registered Server Number
09 USB		
091	SOFTWARE UPGRADE	
092	DB DOWNLOAD TO USB	
093	DB UPLOAD TO USB	
094	VMIB MSG DOWNLOAD	
0# WTU SUBSCRIBE		

Operation

Attendant

To activate an Attendant Station Program Code Feature or Function:

1. Press the **[PGM]** button, the Attendant Station Program Menu is displayed.
 2. Dial 0 to access the Attendant Station Program codes (Display Menu).
 3. Enter the desired code.
- OR
4. Use the **[VOL UP]/[VOL DOWN]** or **[NAVI UP]/[NAVI DOWN]** button to display the desired menu item and enter the desired code.
 5. Enter any additional inputs, if required.

Condition

Programming

- Tenant Data** 1. Attendant Group Assign (PGM 270)

Related Features

- Station Message Detail Recording (SMDR)
- Traffic Analysis
- Temporary Station COS/Lock Authorization Codes (Password)

System Clock Set
VMIB Integrated Auto Attd/Voice Mail
Auto Service Mode Control

Hardware

Digital Phone

7.7 ATTENDANT CALL/QUEUING

Description

Any Station can call the Attendant by dialing the {Attendant Call code}. When an Attendant Call encounters a Busy signal, the Call is Queued to the Attendant Group. The Call will be delivered to the first available Attendant.

Operation

To call the Attendant:

1. Dial the {Attendant Call Code}.

Condition

1. Call Routing order follows the order of entry in the Attendant Assignments program.
2. The Calling Intercom party will receive a Ring-Back tone or MOH, as specified.
3. Calls to the Attendant Station Intercom number are sent to the Attendant Station dialed.
4. When an Attendant Calls another Attendant by dialing Station Number and encounters a Busy signal, Busy tone is received and Call Wait/Voice Over is available.

Programming

- Tenant Data** 1. Attendant Group Assign (PGM 270)

Related Features

7.8 DAY/NIGHT/TIMED RING MODE

Description

The System Clock automatically controls Ring Mode; Ring assignments are applied based on the Time of Day and Day of Week. Three modes of ring (Ring Assignments) are provided: Day, Night and Timed.

The Attendant controls the System Ring Service mode changing from Auto Service Mode to Day, Night or Timed Service mode. Based on the Service Mode selected, different Ring Assignments, COS and Answering Privileges are invoked for System Users.

Operation

Attendant

To modify Day/Timed/Night Ring Mode manually (Attendant Only):

1. Press the [PGM] button.

2. Dial 0 8 1 {Day/Night Program code}.
3. Select Tenant Number (0=All, or 1-9).
4. Select Ring mode (0=AUTO, 1=DAY, 2:=NIGHT, 3=TIMED).

To set Day/Timed/Night Ring Mode automatically (Auto Service Mode Control):

1. Press the **[PGM]** button.
2. Dial 0 8 1 {Day/Night Program code}.
3. Select Tenant Number (0=All, or 1-9).
4. Select Auto Ring mode (0=AUTO).

Conditions

1. Only Attendants can change Day/Timed/Night Ring Mode for the System manually and program the Auto Ring Mode Selection Table.
2. A Station can receive Incoming Calls for CO Lines based on the Database assignments and the System mode (Day/Night/Timed) when the Call arrives.
3. When the Auto Ring Selection Table is programmed, Ringing, COS and CO/IP Access mode are changed automatically based on Times assigned in the Table.
4. The Attendant always has manual control of System mode by Enabling/Disabling the Auto Service Mode Control.

Programming

- | | |
|--------------------|---|
| Table Data | 1. System Time Table (PGM 253/254) |
| Tenant Data | 1. Attendant Group Assign (PGM 270)
2. Tenant Time Table Index (PGM 280) |

Related Features

System Clock Set
Loud Bell Control (LBC)
Dialing Restrictions
Auto Service Mode Control

Hardware

7.9 DSS/DLS CONSOLES

Description

The System allows an unlimited number of DSS/DLS Consoles to be installed in the System. Up to 5 DSS/DLS Consoles may be associated with a Station connected to a System. Each button on the Console can be assigned as CO Line, Intercom, or Feature key.

The User or Administrator may then change individual Flex buttons as desired. Operation of the DSS/BLF Console Flex buttons is the same as Flex buttons on the Digital Phone.

Operation

Conditions

1. Each DSS/DLS Console is assigned to operate in connection with a specific Station.
2. There is no limit to the number of DSS/DLS units in a System beyond the basic System capacities.

Programming

- | | |
|---------------------|-------------------------------------|
| Station Data | 1. Station Type (PGM 120) |
| Tenant Data | 1. Attendant Group Assign (PGM 270) |

Related Features

Station Flexible Buttons
DSS/BLF

Hardware

7.10 EZ-ATTENDANT

Description

The ez-Attendant is a Windows-based PC application that provides a visualization of the Attendant functionality to simplify Attendant control of Features and Functions including Call Display, and User and System status. ez-Attendant operates in conjunction with the Attendant Digital Phone to simplify overall operation (refer to ***ez-Attendant Installation and User Guide***).

Operation

Attendant

Operation of the ez-Attendant is described in detail in the ***ez-Attendant Installation and User Guide***.

Conditions

1. ez-Attendant requires installation of a System Lock-key.

Programming

- | | |
|--------------------|-------------------------------------|
| Tenant Data | 1. Attendant Group Assign (PGM 270) |
|--------------------|-------------------------------------|

Related Features

Hardware

7.11 SYSTEM CLOCK SET

Description

The Attendant can set the System Time/Date.

Operation

Attendant

To set the System Date:

1. Press the **[PGM]** button.
2. Dial 041 {Attendant Station Program code}.
3. Enter the Date (MM/DD/YY).
4. Press the **[SAVE]** button, a confirmation tone is heard.

To set the System Time:

1. Press the **[PGM]** button.
2. Dial 042 {Attendant Station Program code}.
3. Enter the Time (HH/MM).
4. Press the **[SAVE]** button, a confirmation tone is heard.

Conditions

Programming

- | | |
|--------------------|-------------------------------------|
| Tenant Data | 1. Attendant Group Assign (PGM 270) |
| System Data | 2. System Date & Time (PGM233) |

Related Features

Least Cost Routing (LCR)
Station Message Detail Recording (SMDR)
Auto Service Mode Control
Day/Night/Timed Ring Mode

Hardware

7.12 USB UPGRADE

Description

The Attendant can upgrade the System via USB memory. USB upgrade could be executed using the Attendant Keypad. Before upgrading, a User must save the System ROM file (GS55(56)MXXXX.rom) in USB memory.

Operation

Attendant

To upgrade the System using the Attendant Keypad:

1. Save System ROM file (GS55(56)MXXXX.rom) in USB memory.
2. Insert the USB memory to the USB port in MPB board.
3. Press the **[PGM]** button.
4. Dial 091 {Attendant Station Program code}.

MOUNT USB MEMORY
PLEASE WAIT...

5. Number of System ROM files in USB memory is displayed.

ROM FILE NUM : TOTAL 2
PRESS 0-1 TO VIEW FILE

6. Dial the Number of System ROM file to display the appropriate System ROM file name.

0 : GS56MA0Aa.rom
PRESS [HOLD] TO UPGRADE

7. Press the **[HOLD]** button to Upgrade the System.

0 : GS56MA0Aa.rom
PRESS [HOLD] TO UPGRADE

8. Following Upgrade Result is displayed and Keypad will return to Idle state.

SOFTWARE UPGRADE
USB UPGRADE SUCCESS

9. Restart the System.

Conditions

1. USB Upgrade in Attendant Keypad can support up to 10 ROM image files.

Programming

Tenant Data

1. Attendant Group Assign (PGM 270)

Related Features

Hardware

7.13 USB DB UP/DOWNLOAD FROM/TO USB

Description

The Attendant can upload/download the System database via USB memory. USB upload/download can be executed from the Attendant Station.

Operation

Attendant

To download the System Database using the Attendant Keypad:

1. Insert the USB memory to the USB port in MPB board.
2. Press the **[PGM]** button.
3. Dial 092 {Attendant Station Program code}; the display will be as shown.

DB DOWNLOAD TO USB
PRESS OK/SAVE KEY

4. Press the **[HOLD] / [OK]** button to download the database.

DB DOWNLOAD TO USB
PLEASE WAIT...

5. Following download, the result is displayed and Keypad will return to Idle.

DB DOWNLOAD TO USB
DOWNLOAD SUCCESS

To upload the System Database using the Attendant Keypad:

1. Insert the USB memory to the USB port in MPB board.
2. Press the **[PGM]** button.
3. Dial 093 {Attendant Station Program code}.

MOUNT USB MEMORY
PLEASE WAIT...

4. Number of DB files in USB memory is displayed.

DB FILE NUM : TOTAL 2
PRESS 0-1 TO VIEW FILE

5. Dial the Number of DB file to display the appropriate DB file name.

0 :MGDB_ALL090101.adm
PRESS [HOLD] TO UPGRADE

6. Press the **[HOLD]** button to upload the Database.
7. Following Upload, the result is displayed and Keypad will return to Idle.

DB UPLOAD FROM USB UPLOAD SUCCESS

8. Restart the System.

Conditions

1. DB Upload Feature can support up to 10 ROM image files at the Attendant Keypad.

Programming

- | | |
|--------------------|-------------------------------------|
| Tenant Data | 1. Attendant Group Assign (PGM 270) |
|--------------------|-------------------------------------|

Related Features

Hardware

8. SLT

8.1 BROKER CALL

Description

Broker Call allows an SLT User to engage in 2 Calls, alternating between the two Parties, so that the conversation with each Party is private.

There are two types of Broker Call:

- **Transfer Broker Call** – 2nd Call is originated by the SLT user.
- **Call-Wait (Camp-On) Broker Call** – 2nd Call is delivered to the SLT through a Call-Wait.

Operation

SLT

To activate a Transfer Broker Call:

1. While on an active Call, press the Hook-Switch to receive the Intercom Dial tone; the active Call is placed on Exclusive Hold.
2. Place a second Call.
3. To alternate between calls, shortly press the Hook-Switch.

To activate a Call Wait Broker Call:

1. While on an Active Call, and an Incoming Call-Wait is received, press the Hook-Switch to answer the incoming Call.
2. To alternate between Calls, shortly press the Hook-Switch.

Conditions

1. After a Hook-Switch (Flash), if the Call results in an Error, Busy, No Answer or an Abnormal State, the SLT User may shortly press Hook-Switch to retrieve the Held Call.
2. During a Transfer Broker Call, if the SLT User goes On-Hook, the Broker Call Parties are connected (Call Transfer).
3. During a Transfer Broker Call, if the active caller disconnects from the SLT User, the Held Party, if another Station, is connected to the SLT.
4. During a Call-Wait Broker Call, if the SLT User goes On-Hook, the Active Call is disconnected and the Held Call will Recall to the SLT.
5. During a Call-Wait Broker Call, if the Active Party disconnects from the SLT, the SLT User receives an Error tone; after the SLT User goes On-Hook a Recall will be received.
6. If after a Hook-Switch (Flash), the User takes no action for duration of the Dial Tone Timer, the SLT will receive an Error tone; once the SLT goes On-Hook, the SLT will receive a Recall automatically.

Programming

Station Data

1. Hook Flash option when transfer (PGM 124-Index6)

Related Feature

Call Transfer

Hardware

8.2 HOOK-FLASH MODE

Description

To prohibit any service after Hook-Flash, SLT Hook-Flash can be disabled or dropped or ignored.

- Disable Mode – When Hook-Flash detected, the previous Line will be continued, and the Line is not Dropped.
- Flash Drop – When Hook-Flash Flash detected, previous conversation will be disconnected.
- Flash Ignore – All Hook-Flash is ignored.
- Hold Release – During the conversation when a SLT user hook-flashes, the previous line is placed on hold; if the SLT user hangs-up without being transferred, the previous line will be disconnected automatically.

Operation

Conditions

Programming

Station Data 1. Hook Flash Mode (PGM 132-Index7)

Related features

Hardware

SLT

8.3 HOWLER TONE

Description

When an SLT station goes Off-Hook and does not initiate dialing for the duration of the Dial Tone Timer, delays dialing between digits in excess of the Inter-Digit Timer, or stays Off-Hook at the completion of activating a Feature or Program, the Station will present the Howler tone as an Error indication and the Call attempt will be abandoned. In order to complete the Call, the User must return to On-Hook and restart the Call.

Operation

System

The System will deliver a Howler tone automatically, as required.

Conditions

1. Howler Tone is sent after Error tone.
2. Lock-out occurs when Howler tone starts.

Programming

Station 1. Howler Tone (PGM 121-Index7)

Related Features

Intercom Lock-Out

Hardware

8.4 SLT MESSAGE WAIT INDICATION

Description

All SLT devices will receive a Stutter dial tone as an audible Message Wait Indication. In addition, Industry-standard Message Wait telephones may be connected to the System. Software included will cause the Message Wait lamp to Flash when a messaging is waiting.

Operation

System

The System will switch the 90 VDC lamp On and Off for enabled SLTs for visual Message Wait, and will provide a Stutter dial tone as an audible Message Wait indication.

Conditions

1. The System will switch a SLT 90 VDC lamp On and Off (Flash).
2. Although the SLT Battery Feed is removed during the 90 VDC On-cycle, the System will recognize an SLT Off-Hook event.
3. The SLT must incorporate a 90 VDC neon lamp that is connected directly across the tip and ring of the voice network.

Programming

Station Data

1. Station Type (PGM 120-Index1)

Related Features

Hardware

SLT w/90 VDC Neon lamp

8.5 SLT NAME REGISTRATION

Description

A SLT user has the capability to program the User Name so that a Calling Station with an LCD can see the associated Name instead of the Station number.

Operation

SLT

To register a name at the SLT:

1. Lift the handset.
2. Dial {Name Register Feature Code}.
3. Enter name (refer to Alphanumeric Chart).

Alphanumeric Entry Chart

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

4. Press the Hook-Switch; the confirmation tone will be heard.

To delete the Name at the SLT:

1. Lift the handset.
2. Dial {Name Register Feature Code}.
3. Press the Hook-Switch; confirmation tone will be heard.

Conditions

Programming

Related Features

Dial-by-Name
Station Speed Dial

Hardware

9. SIP PHONE

9.1 SIP TERMINAL REGISTRATION

Description

The System supports the LG-Ericsson SIP-based videophone, LVP-2000, as well as other third party SIP phones. Compatible SIP phones support the Internet Engineering Technical Committee standard RFC3261 for real-time communications over the Internet. Once registered, the System will deliver services to the SIP Phone. Operation of the SIP Phone generally follows the steps outlined for an SLT.

Operation

Web Admin

To register a SIP Phone:

1. Select IP Phone Registration Table.
2. Enter the Auth ID, Password, and User ID of SIP Phone.
3. Click the Save button.

SIP Phone

1. Configure SIP Phone settings (ex. IP address, Subnet mask, Gateway, Telephone number, Proxy Address, Expiration Timer, etc.); the Telephone is the Station number to be assigned to Phone by System and the Proxy Address is the System MPB IP address.
2. Boot the SIP Phone, which will activate registration with the System.

Conditions

1. The SIP Phone not supporting standard SIP protocol is not supported.
2. Support for 3rd-party SIP phones requires a license.
3. The 3rd-party applications supporting standard SIP protocol can be registered.
4. The System checks the SIP status periodically; if the System does not receive the 'REGISTER' message from the SIP Phone in Registration Time, the SIP Phone registration attempt is cancelled.
5. A VOIB channel is needed for the SIP Phone to Place or Receive Calls.

Programming

Related Features

Hardware

9.2 SIP NAME REGISTRATION

Description

If the SIP phone has a Function to register its own Phone Name, when a call is placed, the SIP Phone Name is displayed on the Called Party Station.

Operation

Conditions

Programming

Related Features

Hardware

9.3 SIP PLACING CALLS

Description

The SIP Phone can place Intercom or External Calls. To place a Call, a VOIB channel should be available.

Operation

To place an Intercom Call:

1. Dial station number.
2. Press the **[OUTGOING]** button on the SIP Phone.

To place an External Call:

1. Dial the {CO Access Code} and telephone number.
2. Press the **[OUTGOING]** button on SIP Phone

Conditions

1. If the SIP phone does not have an **[OUTGOING]** button, dial '#' or wait in order to make a Call (refer to the SIP Phone manual for further information).
2. The SIP phone must be registered correctly to be operational.
3. For SIP Phone Features (ex., Redial or Call Log) to work, the phone must be enabled with those features and be able to place a call using its own function.
4. Available features when making calls include:
 - Intercom Call
 - CO Line Call
 - ATD Call
 - Call Pick-Up
 - System Speed Dial
 - Station Group Call

- Interphone Group Call
- Conference Room
- Paging or Meet-me Page
- Announcement
- Voicemail
- Call Park Response

Programming

Related Features

Hardware

9.4 SIP CALL PICK-UP

Description

The SIP phone can Pick-Up Intercom or Incoming CO Line Calls for other Stations.

Operation

To answer an Incoming call:

1. When Ringing is received and caller number is displayed, lift handset or press **[ANSWER]** button on SIP Phone.

Conditions

1. For SIP Phone Features (ex., Receive Calls, etc.) to work, the phone must be enabled with those features and be able to place a call using its own function.

Programming

Related Features

Hardware

9.5 SIP HOLD CALL

Description

The SIP Phone can place an Active Call on Hold.

Operation

To place an Exclusive Hold:

1. While on an Active Call, press the **[HOLD]** button; the Call will be placed on Exclusive Hold, and will not be able to be accessed by other Stations.

Conditions

1. If the SIP phone has a Call Hold Function, it will not operate correctly if the Hold message is not compatible between the SIP Phone and the System.

Programming

Related Features

Hardware

9.6 SIP TRANSFER CALL

Description

The SIP phone can Transfer an Active Call.

Operation

To Transfer an Active Call:

1. While on an Active Call, press the [TRANSFER] button on SIP Phone.
2. Dial Station Number where the call will be transferred.

Conditions

1. If the SIP phone has a Transfer Function, it will not operate correctly if the Transfer message is not compatible between the SIP Phone and the System.

Programming

Related Features

Hardware

9.7 SIP CALL FORWARD

Description

The SIP phone can set-up Call Forward.

Operation

Refer to the SIP Phone User Guide to set-up Call Forward.

Conditions

1. If the SIP phone has a Call Forward Function, it will not operate correctly if the Call Forward message is not compatible between the SIP Phone and the System.

Programming

Related Features

Hardware

9.8 SIP Do NOT DISTURB (DND)

Description

The SIP phone can set Do Not Disturb (DND) when Station User is in need of privacy to complete the current Call (rejects incoming calls until Phone returns to Idle).

Operation

To Activate DND:

1. While on an Active Call, press the **[DND]** button on SIP Phone; incoming calls will be rejected until the phone returns to Idle (cancel DND mode).

Conditions

1. If the SIP phone has a DND Function, it will not operate correctly if the DND message is not compatible between the SIP Phone and the System.

Programming

Related Features

Hardware

9.9 SIP 3-PARTY CONFERENCE

Description

The SIP phone can make a 3-party conference.

Operation

Refer to the SIP Phone User Guide to set up a Conference.

Conditions

1. If the SIP phone has a Conference Function, it will not operate correctly if the Conference message is not compatible between the SIP Phone and the System.

Programming

Related Features

Hardware

9.10 SIP CALL WAIT/BROKER CALL

Description

The SIP phone can initiate Broker Call when the SIP Phone User receives Call Wait indication.

Operation

Refer to the SIP Phone User Guide to initiate Broker Call.

Conditions

1. If the SIP phone has a Call Wait/Broker Call Function, it will not operate correctly if the Call Wait/Broker Call message is not compatible between the SIP Phone and the System.

Programming

Related Features

Hardware

9.11 SIP SMS

Description

The SIP phone can Send and Receive SMS messages with other Stations.

Operation

Refer to the SIP Phone User Guide for SMS.

Conditions

1. If the SIP phone has a SMS Function, it will not operate correctly if the SMS message is not compatible between the SIP Phone and the System.

Programming

Related Features

Hardware

9.12 SIP VOICE MAIL NOTIFICATION

Description

The SIP phone can receive a notification when another Station leaves Voicemail. The SIP phone must be enabled with Voicemail Notification.

Operation

If set, Voicemail Notification is automatic.

Conditions

1. If the SIP phone has a Voicemail Function, it will not operate correctly if the Voicemail is not compatible between the SIP Phone and the System.

Programming

Related Features

Hardware

9.13 SIP VIDEO CALL

Description

The SIP phone can make a Video Call using a Video Phone. To make a Video Call, the Stations must be enabled for Voice Calling.

Operation

Refer to the SIP Phone User Guide for Video Call.

Conditions

1. The System does not transfer the Video Codecs between Video Phones; if Phones have different Video Codecs, an Audio Call will be established instead of a Video Call.
2. To make a Video Conference, it is necessary to have an additional Video Conference System (ex., IP-polycom).

Programming

Related Features

Hardware

10. APPENDIX

10.1 NUMBERING PLAN SET

1) Basic Number

No	Name	NUM SET 1	NUM SET 2	NUM SET 3	Remark
1	Station Number	100 ~ 473	100 ~ 699	1000 ~ 1647	
2	CO Group Access Code	1, 801 ~ 872(MG-300) 801 ~ 824(MG-100)	0, *801 ~ *872(MG-300) *801 ~ *824(MG-100)	9, 801 ~ 872(MG-300) 801 ~ 824(MG-100)	
3	Station Group Number	620 ~ 669(MG-300) 620 ~ 639(MG-100)	*620 ~ *669(MG-300) *620 ~ *639(MG-100)	620 ~ 669(MG-300) 620 ~ 639(MG-100)	

No	Name	NUM SET 4	NUM SET 5	NUM SET 6	Remark
1	Station Number	7000 ~ 7647	2000 ~ 2647	2000 ~ 2647	
2	CO Group Access Code	1, 401 ~ 472(MG-300) 401 ~ 424(MG-100)	0, 801 ~ 872(MG-300) 801 ~ 824(MG-100)	0, 801 ~ 872(MG-300) 801 ~ 824(MG-100)	
3	Station Group Number	620 ~ 669(MG-300) 620 ~ 639(MG-100)	620 ~ 669(MG-300) 620 ~ 639(MG-100)	*620 ~ *669(MG-300) *620 ~ *639(MG-100)	

2) Feature Code

No	Feature Name	NUM SET 1	NUM SET 2	NUM SET 3	Remark
1	Attendant Call	0	*9	0	
2	Conference Room 1	571	*571	571	
3	Conference Room 2	572	*572	572	
4	Conference Room 3	573	*573	573	
5	Conference Room 4	574	*574	574	
6	Conference Room 5	575	*575	575	
7	Conference Room 6	576	*576	576	
8	Conference Room 7	577	*577	577	
9	Conference Room 8	578	*578	578	
10	Conference Room 9	579	*579	579	
11	Internal Page	543	*543	543	543 + 00, xx 00: All Call Page Xx: Page Group #
12	Personal VM Page	544	*544	544	
13	Announcement Page For Attendant	545	*545	545	
14	Page Auto Answer	546	*546	546	
15	Internal Page Answer (Meet-Me Page)	547	*547	547	
16	External Page	548	*548	548	
17	Internal-External Page All	549	*549	549	
18	Call Forward Register	554	*554	554	554 + Type + Destination

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Feature Description & Operation

Issue 1.1

No	Feature Name	NUM SET 1	NUM SET 2	NUM SET 3	Remark
19	Pilot Hunt Call Forward Register	514	*514	514	514 + Type + Destination
20	Pilot Hunt Call Forward Cancel	515	*515	515	
21	DND Status Change	516	*516	516	
22	DND Delete	517	*517	517	
23	Account Code	550	*550	550	
24	CO Flash	551	*551	551	
25	Last Number Redial	552	*552	552	
26	Station Speed PGM	553	*553	553	
27	Speed Dial	555	*555	555	
28	MWI Register	556	*556	556	
29	MWI Answer	557	*557	557	
30	MWI Cancel	559	*559	559	
31	Call Back Register	518	*518	518	
32	Call Back Cancel	519	*519	519	
33	Group Call Pickup	566	*566	566	
34	Direct Call Pickup	7	*7	7	
35	Walking COS	520	*520	520	
36	Call Parking Location	541	*541	541	541 + xx Xx: Parking Location (00 ~ 49)
37	PGM Mode Access	521	*521	521	
38	Two-Way Record	522	*522	522	
39	VMIB Access	523	*523	523	
40	AME Access	524	*524	524	
41	CO Line Access	88	*88	88	88 + xxx Xxx: CO Line # (001 ~ 200 : MG-300 01 ~ 80 : MG-100)
42	VM MWI Enable	*8	*5#8	*8	
43	VM MWI Cancel	*9	*5#9	*9	
44	MCID Request	*0	*5#0	*0	
45	Unsupervised Conf Extend	5##	*5##	5##	
46	PTT Group Access	538	*538	538	524 + (0~9,*) 0 ~ 9: PTT Group # *: Log out
47	Hot Desk Log In/Log out	525	*525	525	
48	Name Register	526	*526	526	
49	Create Conf Room	527	*527	527	527 + Conf. Room #
50	Delete Conf Room	528	*528	528	528 + Conf. Room #
51	Wake Up Register	529	*529	529	529 + HH:MM
52	Wake Up Cancel	530	*530	530	
53	Temporarily COS Down	531	*531	531	
54	Cancel Temp COS Down	532	*532	532	
55	Password Change	533	*533	533	
56	Inter-Phone Group Access	534	*534	534	
57	Call Wait Request	535	*535	535	
58	Preselected MSG PGM	536	*536	536	

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Feature Description & Operation

Issue 1.1

No	Feature Name	NUM SET 1	NUM SET 2	NUM SET 3	Remark
59	Forced Handsfree Call	537	*537	537	
60	Call Based CLIR	582	*582	582	
61	CLIR Access	583	*583	583	
62	COLR Access	584	*584	584	
63	Pilot Hunt Call	585	*585	585	
64	Command Call Oneway	581	*581	581	
65	Command Call Conf	580	*580	580	
66	Intrude Register	589	*589	589	
67	Camp On Register	590	*590	590	
68	OHVO Register	591	*591	591	
69	Mobile Num Register	592	*592	592	
70	Mobile CLI Register	593	*593	593	
71	Mobile Access	594	*594	594	
72	CCR Access	670	*670	670	
73	CCR Access And Drop	671	*671	671	
74	System Hold	560	*560	560	
75	Return Held CO	8**	*8**	8**	
76	Sys Memo	675	*675	675	
77	DISA Tone Service	678	*678	678	
78	All Feature Cancel	679	679	679	
79	Add Conf Member	680	*680	680	
80	System Alarm Reset	565	*565	565	
81	Fault Alarm Reset	564	*564	564	
82	Door Open	#*1	*#1	#*1	
83	Keypad Facility	##*	*##*	##*	
84	T-Net Log-In/Out	586	*586	586	
85	Universal Answer	587	*587	587	
86	USB Call Record	588	*588	588	
87	Delete All VM Message	681	*681	681	
88	VM Page Message Record	682	*682	682	
89	Direct VM Transfer	683	*683	683	
90	Loop Key	684	*684	684	
91	Call Log	685	*685	685	

No	Feature Name	NUM SET 4	NUM SET 5	NUM SET 6	Remark
1	Attendant Call	0	9	#9	
2	Conference Room 1	571	571	*571	
3	Conference Room 2	572	572	*572	
4	Conference Room 3	573	573	*573	
5	Conference Room 4	574	574	*574	
6	Conference Room 5	575	575	*575	
7	Conference Room 6	576	576	*576	
8	Conference Room 7	577	577	*577	
9	Conference Room 8	578	578	*578	
10	Conference Room 9	579	579	*579	

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Feature Description & Operation

Issue 1.1

No	Feature Name	NUM SET 4	NUM SET 5	NUM SET 6	Remark
11	Internal Page	543	543	*543	543 + 00, xx 00: All Call Page Xx: Page Group #
12	Personal VM Page	544	544	*544	
13	Announcement Page For Attendant	545	545	*545	
14	Page Auto Answer	546	546	*546	
15	Internal Page Answer (Meet-Me Page)	547	547	*547	
16	External Page	548	548	*548	
17	Internal-External Page All	549	549	*549	
18	Call Forward Register	554	554	*554	554 + Type + Destination
19	Pilot Hunt Call Forward Register	514	514	*514	514 + Type + Destination
20	Pilot Hunt Call Forward Cancel	515	515	*515	
21	DND Status Change	516	516	*516	
22	DND Delete	517	517	*517	
23	Account Code	550	550	*550	
24	CO Flash	551	551	*551	
25	Last Number Redial	552	552	*552	
26	Station Speed PGM	553	553	*553	
27	Speed Dial	555	555	*555	
28	MWI Register	557	556	*556	
29	MWI Answer	558	557	*557	
30	MWI Cancel	559	559	*559	
31	Call Back Register	518	518	*518	
32	Call Back Cancel	519	519	*519	
33	Group Call Pickup	**	566	*566	
34	Direct Call Pickup	7	7	*7	
35	Walking COS	520	520	*520	
36	Call Parking Location	541	541	*541	541 + xx Xx: Parking Location (00 ~ 49)
37	PGM Mode Access	521	521	*521	
38	Two-Way Record	522	522	*522	
39	VMIB Access	523	523	*523	
40	AME Access	524	524	*524	
41	CO Line Access	88	88	88	88 + xxx Xxx: CO Line # (001 ~ 200 : MG-300 01 ~ 80 : MG-100)
42	VM MWI Enable	*8	*8	*5#8	
43	VM MWI Cancel	*9	*9	*5#9	
44	MCID Request	*0	*0	*5#0	
45	Unsupervised Conf Extend	5##	5##	*5##	
46	PTT Group Access	538	538	*538	524 + (0~9,*) 0 ~ 9: PTT Group # *: Log out

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Feature Description & Operation

Issue 1.1

No	Feature Name	NUM SET 4	NUM SET 5	NUM SET 6	Remark
47	Hot Desk Log In/Log out	525	525	*525	
48	Name Register	526	526	*526	
49	Create Conf Room	527	527	*527	527 + Conf. Room #
50	Delete Conf Room	528	528	*528	528 + Conf. Room #
51	Wake Up Register	529	529	*529	529 + HH:MM
52	Wake Up Cancel	530	530	*530	
53	Temporarily COS Down	531	531	*531	
54	Cancel Temp COS Down	532	532	*532	
55	Password Change	533	533	*533	
56	Inter-Phone Group Access	534	534	*534	
57	Call Wait Request	535	535	*535	
58	Preselected MSG PGM	536	536	*536	
59	Forced Handsfree Call	537	537	*537	
60	Call Based CLIR	582	582	*582	
61	CLIR Access	583	583	*583	
62	COLR Access	584	584	*584	
63	Pilot Hunt Call	585	585	*585	
64	Command Call Oneway	581	581	*581	
65	Command Call Conf	580	580	*580	
66	Intrude Register	589	589	*589	
67	Camp On Register	590	590	*590	
68	OHVO Register	591	591	*591	
69	Mobile Num Register	592	592	*592	
70	Mobile CLI Register	593	593	*593	
71	Mobile Access	594	594	*594	
72	CCR Access	670	670	*670	
73	CCR Access And Drop	671	671	*671	
74	System Hold	560	560	*560	
75	Return Held CO	8**	8**	*8**	
76	Sys Memo	675	675	*675	
77	DISA Tone Service	678	678	*678	
78	All Feature Cancel	679	679	*679	
79	Add Conf Member	680	680	*680	
80	System Alarm Reset	565	565	*565	
81	Fault Alarm Reset	564	564	*564	
82	Door Open	#*1	#*1	#*1	
83	Keypad Facility	##*	##*	##*	
84	T-Net Log-In/Out	586	586	*586	
85	Universal Answer	587	587	*587	
86	USB Call Record	588	588	*588	
87	Delete All VM Message	681	681	*681	
88	VM Page Message Record	682	682	*682	
89	Direct VM Transfer	683	683	*683	
90	Loop Key	684	684	*684	
91	Call Log	685	685	*685	