iPECS-MG

Feature Description & Operation Manual

Please read this manual carefully before operating System.

Retain it for future reference.

iPECS is an Ericsson-LG Brand

Revision History

ISSUE	DATE	DESCRIPTION OF CHANGES
1.0	2009.12	Initial Release
1.5A	2010. 06	Update for iPECS-MG S/W V1.5 Section 3.21.3 Consultation Conference added Section 3.68.2.14 Centralized Voice Mail added Section 3.74.3.5 E-Mail Notification updated Section 4.10.2 Meet Me Page Answer updated Section 4.10.3 VM Paging updated Section 5.6.7 DID Name Service added Section 5.6.8 Incoming CO Line Holiday Service added Section 5.6.9 DID/DISA Restriction added Section 7.3 Greeting/Queuing Tone Service added Section 7.4 CCR Service For Attendant Queuing Announcement added Section 7.9 Day/Night/Time Ring Mode updated Section 10 ACD (Automatic Call Distribution) added Section 11.1 Numbering Plan Set updated
1.5B	2010. 09	Update for iPECS-MG S/W V1.5 (General edits for errata) Section 3.11 Call Forward updated Section 3.16.2 CO/IP Call Transfer updated Section 3.31 Digit Conversion updated Section 3.50 Mobile Extension updated Section 3.53 Music-On-Hold updated Section 3.56.1 Camp-On updated Section 3.56.2 Call Wait updated Section 3.57 Pre-defined & Custom Text Display Messages updated Section 3.62.2 Station Speed Dial updated Section 3.64.2 Greeting/Queuing Tone Service updated Section 3.64.3 CCR Service with Queuing Announcement updated Section 3.64.4 Forward Destination, Overflow Service updated Section 3.68.2.7 Message Waiting Indication (MWI) updated Section 5.2 Automatic Network Dialing updated Section 5.9 CO/IP Ring Assignment updated
1.7A	2011 03	Update for iPECS-MG S/W V 1.7 Table 2.2.4 1 System Capacity Chart updated (Bar Record) Section 3.15.3 Queued Hunt Call Pick-Up added Section 3.65.3 SMDR Option For Second Information added Section 3.51 Multi Language Support updated Section 3.74.3.5 E-Mail Notification updated Section 3.74.5 CCR Without DISA added Section 3.74.6 CCR Reroute Destination added Section 3.74.7 Rerouting From Voice Mail Forwarding added Section 3.76 PSU Fan Alarm added Section 5.18.2 Incoming Calling Line ID (ICLID) Call Routing Exception added Section 5.24 Collect Call Blocking for E1-R2 and in Brazil added Section 5.25 BRIB Clock Port Selection added Chapter 11 Hotel Feature added Section 12.1 Numbering Plan Set updated (Feature Codes)

ISSUE	DATE	DESCRIPTION OF CHANGES
2.0	2011. 10	Update for iPECS-MG S/W P2.0 Section 3.30 Differential Ring updated Section 3.42.4 Direct/Indirect Held CO Retrieve added Section 3.74.3.15 VM Password Input added Section 3.74.3.16 VM Private Message added Section 3.74.3.17 VM Message Delivery Confirmation added Section 3.74.3.18 VM Message Future Delivery added Section 3.74.3.19 VM Message Fast Forward/Rewind added Section 3.74.3.20 VM Message Pause/Start added Section 3.74.8 Voice Mail Service Retrial added Section 3.77 Web Phone added Section 3.78 Emergency Supervisor added Section 3.79 Call Duration Restriction added Section 3.79.1 ICM Duration Restriction added Section 3.79.2 Incoming Call Duration Restriction added Section 3.79.3 Normal Outgoing Call Duration Restriction added Section 3.79.4 Local / Long / International Outgoing Call Duration Restriction added Section 3.79.5 Dedicated Line Call Duration Restrict added Section 3.79.6 Mobile Call Duration Restrict added Section 3.80 Tenant Group Access added Section 3.81 Tone Service For DECT Switch-Off Case added Section 5.26 Fail Over PSTN added Section 5.27 Virtual Subscriber Service added Section 5.28 Private CO Group added Section 6.24 Override (Disconnect) added Section 6.25 USB Auto Call Record added Section 9.14 SIP Phone Voice Mail Notification added Section 9.15 SIP Phone BLF added Section 9.16 SIP Phone Distinctive Ring added Section 9.17 SIP Phone Intercom added Section 9.18 SIP Phone Call-Back added Section 9.19 SIP Phone Call Intrusion added Section 9.20 SIP Phone Call Override added Section 10.11 ACD Group Supervisor Feature added
2.1	2012. 12	Changed CI to Ericsson-LG
2.2	2013. 12	Changed CI to Ericsson-LG Enterprise and update S/W P2.5

Table of Contents

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
- Section 10 – ACD 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.

1.2 System Capacities

The iPECS-MG Series is available in several configurations as listed in Table 1.2-1. Total port capacities range from the 200 channel at iPECS-MG 100 to 414 channel at iPECS-MG 300.

Table 1.2-1 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	
	Feature: 8Digits	
	Trunk: 8 Digits	
Attendant	5/Tenant	
DSS/BLF Console	5	
Member of conference	3 Group/13 Members	
Internal Page zone	15	30
System speed dial	1000	2000
	(32 digits)	
Station Speed Dial	50 (32 digits)	
Call Log (Outgoing/Incoming/Missed Call)	100 (32 digits) (Not protected)	
Save Number Redial(SNR)	1 (32 digits)	
Number of SMDR Records	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)	
Interphone Group	10 (10 member/Group)	
Page Group	15 (50 member/Group)	30 (50 member/Group)
PTT Group	10 (50 member/Group)	
Conference Room	9	

Table 1.2-1 System Capacity Chart

Items	iPECS-MG 100	iPECS-MG 300
Number of Hot Desk Agent	60	324
Station Name Information	16 Characters	
Digit Restriction	COS: 16	
	Allow/Deny Entry per COS: 100	
	Max. Digit: 16	
Digit Translation	Table No: 9	
	Number of Digit: 16	
	300 per 1 table	
No Of Bar Record	3000	7000

2. DIRECTORY NUMBER (DN)

Directory Number (DN) is the telephone number for 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

Description

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

Numbering Plan	1 Flexible Station Number (PGM 112)
Station Data	1 Station DN Type (PGM 130 – Index1)
	2 MADN Member (PGM 130 – Index2)
	3 Prime Number Button (PGM 123 – 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. 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 can be programmed in PGM 234

Default 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

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-Indication only

Description

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.
- Indication only : Indication and mute ring can be provided to station

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.

Operation

Conditions

Programming

Station Number Data 1. Station DN assignment – Ring Option (PGM130)

Related Features

Hardware

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.

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:*

1. 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 **[PGM]** button.
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.

<u>*To enter an Account Code during a call:*</u>

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

Conditions

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

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

Digital Phone

<u>*To assign a Flex button as an **{ALARM STOP}** button:*</u>

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

<u>*To terminate an Alarm Signal while idle:*</u>

1. Dial the **{Sys Alarm Reset Feature Code}**; a confirmation tone is received and the Alarm Signal is terminated. Or, press the programmed **{ALARM STOP}** Flex

button.

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

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. (PGM 121 – 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 Data	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 or System, 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.

There are two types of Authorization Codes, Station and System. A Station Authorization Code is specifically related to a given station and intended for a single user. The System Authorization Codes are intended for use by any station in the system.

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 System Authorization Code consists of 3 parts: "*", the associated ID, and the assigned Password.

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.

Usage

<u>To enter an Authorization Code:</u>

1. Dial the DN number for the Station Authorization code or, for a System Authorization Code, dial '*' and ID for a System Authorization code.
2. Dial the corresponding Authorization Code.(Password)
3. Dial "*" or, '#'

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 '#'.
3. When a user want to enter system authorization code, enter "*", ID, and Password.

Programming

Station Data	1	Password (PGM 131 – Index4)
System Data	1	System Authorization Code (PGM225)

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

<u>Auto Call Release of Intercom calls:</u>

1. 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

Table Data	1	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.

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 change privacy mode in conversation:

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

Conditions

1. With Automatic Privacy disabled, privacy is still assured on all intercom and conference calls.
2. Emergency Supervisor can request OHVO, Override (Hold), Override (Disconnect) regardless of Automatic Privacy.

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 Data

1 System Time (PGM 233 – Index1)
2 System Date (PGM 233 – Index2)
3 DST Enable (PGM 233 – Index3)
4 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 Data

1 Network Time/Date (PGM 223 – Index5)
2 NTP Active (Web Admin.)
3 NTP Sever address (Web Admin.)
4 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.
2. The Lithium battery is not field-replaceable.

Programming

Related Features

Hardware

3.11 Call Forward

3.11.1 Call Forward

Description

User may have selected incoming calls re-route to other stations (local or networked), station groups, the VMIIB, 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. In this case, call forward type is unconditional by default.
- 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, dial CO Group Access code and desired external phone number.
5. Press the **[SAVE]** button to save.
6. 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 the destination station or station group. Or, dial CO Group Access code and desired external phone number.
6. Press **[SAVE]** button to save.
7. 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 the station or station group to receive calls. Or, dial CO Group Access code and 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 the destination station or station group. Or, dial CO Group Access code and desired external phone number..
6. Press hook-switch to save.
7. 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 can be released with release type and time in CO to CO Attribute.
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 143 – 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)
CO Data	1	CO to CO Attribute (PGM 179)
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

3.11.2 Call Forward Option (D/N/T, Int/Ext)

Description

This feature allows options for station forward feature. (Day/Night/Timed and Internal/External)

Operation

Digital Phone and LIP Phone (LDP/LIP Phone).

1. Lift the handset or press the **[SPEAKER]** button to receive a dial tone.
2. Press the **[FWD]** button.
3. Enter Forward Type (0-4).
4. Enter the destination.
5. Press **[Save]** button.
6. Press **[TRANS]** button to enter D/N/T menu.
1. Select Day (1), Night (2), Timed (3), for All, Dial '0', or do not enter any dial.
7. Press **[TRANS]** button to enter INT/EXT menu.
8. Select Internal(1), external(2), all (0).
9. Press **[Save]** button.

SLT

1. Lift the handset to receive Dial tone.
2. Dial **{Call Forward feature code}**.
3. Enter Forward Type (0-4).
4. Enter the destination.
5. Press **[Hook Flash]** button to save.
6. Press **[Hook Flash]** button to enter D/N/T menu.
7. Select Day (1), Night (2), Timed (3), for All, Dial '0' or do not enter any dial.
8. Press **[Hook Flash]** button to enter INT/EXT menu.
9. Select Internal(1), external(2), all (0).
10. Press **[Hook Flash]** button.

Condition

1. If a user does not enter D/N/T, INT/EXT, it can be set as all type,
2. This cannot be applied to SIP terminal.

Programming

NUMBERING DATA 1. Call Forward Register (PGM 113-18)
STATION NUMBER DATA 1. Forward Apply Condition (PGM 143-6)

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 VMIB.

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. Max 24 digits can be assigned to a Pilot Hunt Forward destination.
4. 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.
5. 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 and DND 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.
- Internal DND – Intercom calls that encounter a DND are forwarded immediately.
- 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.
- External DND – external calls that encounter a DND are forwarded immediately.

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.
- DND – if the station is DND, all calls are forwarded to the selected station.

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 143 – 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 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 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 Plan	1	Feature Numbering Plan - Call Park (PGM 113)
Data 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:*

1. 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}** button. Or, 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 Plan	1	Feature Numbering Plan – Direct Call Pick-Up (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:*

1. 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, 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. When several calls are ringing at a station simultaneously, Call Pick-up will connect the first call received.
2. Queue callback calls are not subject to Call Pick-up (receives error tone).
3. Only ringing intercom calls are subject to Call Pick-up; Hands free announced intercom calls cannot be picked up by another station.
4. When a station belongs to multiple pick up groups, system finds the call to pick-up from first pick up group.

Programming

Numbering Data	1	Feature Numbering Plan – Group Call Pick-Up(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

3.15.3 Queued Hunt Call Pick-Up

Description

This feature allows a station to pick up the call in queue of a station hunt group, ACD group or Attendant group.

Operation

To pick up a call in queue of a station hunt group, ACD group or an attendant group:

1. Lift the handset or press the **[MON]** button.
2. Dial **{Direct Call Pick Up}** feature code.
3. Dial the station group number to pick up a queued call.
4. In case of attendant group, dial **{Attendant Call}** feature code.
5. If there's any queued call, the station will be connected to the caller.
6. Otherwise, error tone will be heard.

Conditions

1. This feature is available for normal station group and ACD group.
2. Only queued calls can be picked up by this feature. For the normal station group, the caller should hear first or second queuing announcement in order to be picked up. And for ACD group, the caller should hear one of the 5 queuing announcements (1st to 5th queuing announcement).
3. The first call in the queue will be served first
4. To pick up a queued call, the station doesn't need to belong to the station group.

Programming

Numbering Data	1	Feature Numbering Plan – Direct Call Pick-Up (PGM 113)

Related Feature

- Intercom Signaling Mode
- Directed Call Pick-up
- 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 Transfer 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, press the **{DSS/BLF}** button for the desired station.
5. When answered or when splash tone is heard, announce the call.
6. 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 press **{DSS/BLF}** button for the desired station.
4. 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 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, Transfer 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. The CO-to-CO transfer can be enabled or disabled by using the Transit Option on CO Group basis and also by using offnet forward option on station basis.
4. 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.
5. When CO 1 is transferred to CO 2, if the CO 2 does not answer within the CO-to-CO transfer recall timer, the call is routed to the 'Transfer No answer Destination' of the CO Alternative Destination.

Programming

Station Number Data	1	Offnet Forward Access (PGM 132 – Index3)
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 – index 1)

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

<u>*To place an outgoing CO call:*</u>

1. Lift the handset or press the **[SPEAKER]** button.
2. Press desired **{CO}** line or **{CO ACESS CODE}** button. Or, dial the CO line or CO Access Code.
3. Dial the desired number.

<u>*To place an outgoing IP call:*</u>

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

<u>*To receive an IP call:*</u>

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.

<u>*To answer an incoming CO/IP call:*</u>

1. Lift the handset or press the **[SPEAKER]** button. Or, press **{DN}** button and lift the handset to speak privately.

SLT

<u>*To place an outgoing CO/IP call:*</u>

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

<u>*To answer an incoming CO/IP call:*</u>

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, 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. The Tx path to a station will be muted until the system has verified the Toll Restriction for the CO/IP line.
5. When a CO line is seized, the system will monitor the line for dial tone.
6. 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 (PGM 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}, {Co Line Access Code}** button or dial the CO Group/Line Access Code.
2. Press the **[MSG/CALL BK]** button, a confirmation tone is received.
3. Hang-up; the **[MSG/CALL BK]** LED will flash.

To cancel the queue from the queued station:

1. Press the **[MSG/CALL BK]** button, the **[MSG/CALL BK]** 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 Call Back Indication Ring Timer (PGM 265-9), 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 busy CO line returns to idle. Therefore, Queue Recall ring may be delayed after 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	2	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 establish an ad-hoc conference: ("Conference Member Manual Add" is set OFF)

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, the called party is put into conference and calling party also returns to conference.
5. Repeat steps 3 and 4 above to add additional conference parties (up to 13 total per conference).

To get out from a conference temporarily:

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

To return to a conference call:

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

SLT

To establish an ad-hoc conference: ("Conference Member Manual Add" is set ON)

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. **{Conference Member Add Code}** should be dialed at least once.
6. When completed adding parties, quickly press the hook-switch twice (within 2 seconds), all parties will be connected.

To establish an ad-hoc conference: ("Conference Member Manual Add" is set OFF)

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, the called party is put into conference and calling party also returns to conference.
6. Repeat steps 2–5 above to add additional parties. **{Conference Member Add Code}** should be dialed at least once.

Conditions

1. The **[CONF]** button will remain illuminated at all the phones that are temporarily out of conference 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)
Tenant Data	1	Conference Member Manual Add (PGM 281 – Index1)

Related Features

- Automatic Speaker Select
- Hold Recall
- Unsupervised Conference
- Broker Call
- Conference Room

Hardware

3.21.3 Consultation Conference

Description

A Digital phone user may establish a conference while talking with a party as a screened transfer, which results in a 3-party conference.

Operation

Digital Phone

To set up a Consultation Conference:

1. Press the **[TRANS]** button while talking with an internal or external party.
2. Make a call to another internal or external party.
3. Talking on a second call, press the **[CONF]** button.
   Then, 3-party conference will be established.

Related Features

- Multi-Party Voice Conference

3.21.4 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)

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 30 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. If one member is ring assigned or a delay is 0, the station will receive the ring immediately.
3. If there is no ring assigned station, the call will be routed to busy destination in incoming CO alternative destination.
4. The delay is applied only when ring service type is 'Ring Assign'.

Programming

CO Line Data	1	CO 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 'Ring Assign', 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 ring assigned station immediately and delayed auto attendant is not operated.
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	2	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 DOWN]** 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}**.
3. Enter the name (up to 16 characters); refer to Station Speed Dial for character entry.
4. Press the **[SAVE]** button.

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 "LIST EMPTY" message is displayed.
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

<u>To switch from pulse to DTMF while on a pulse CO line:</u>

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. 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, "*", "#", "X".
- 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 both Allow and Deny entries, the Deny entries are searched. If the dialed number matches a Deny entry, the Allow entries are searched; if no match is found the call is restricted;

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 Data 1 Station COS (PGM 137)

CO Line Data	1	CO COS (PGM 177)
	2	CO-to-CO Attribute; Incoming to Outgoing - Direct (PGM 179 – Index5)
CO Line Data	1	Toll Exception Table (PGM 250)

Related Features

- Temporary Station COS/Lock
- Walking 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/Lock
- Walking COS
- Auto Service Mode Control
- Day/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 (i.e. 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 Down}**.
3. Dial the applicable Authorization Code.
4. Press the **[SAVE]** button.

To retrieve the assigned COS:

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

SLT

To activate Temporary COS:

1. Press the hook-switch.
2. Dial the **{PGM Mode Access Feature Code}**.
3. Dial 31, **{Temporary COS Down Program Code}**.
4. Dial the applicable Authorization Code.
5. Press the hook-switch.

To retrieve the assigned COS:

1. Press the hook-switch.
2. Dial the **{PGM Mode Access Feature Code}**.
3. Dial 32, **{Retrieve COS Program Code}**.
4. Dial Authorization Code.
5. Press the hook-switch.

System Attendant

To activate Temporary COS:

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

To retrieve the assigned COS:

1. Press the **[PGM]** button.
2. Dial 32, **{Retrieve COS}**.

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 Data	1	Station COS Assignment (PGM 137)
CO Line Data	1	CO COS Assignment (PGM 177)
Table Data	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

<u>*To activate Walking COS:*</u>

1. Press the **[PGM]** button.
2. Dial 33, **{Walking COS}**.
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

<u>*To activate Walking COS:*</u>

1. Dial the **{PGM Mode Access Feature Code}**.
2. Dial '33', (Walking COS Program 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 Data	1	Station COS (PGM 137)
CO Line Data	1	CO COS (PGM 177)
System Data	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, 8 or 15 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
   01-15: LDP-6000 series, LDP-7000 series and LDP-9000 series.
   1–4: Digit Phone
   1–8: IP Phone
6. Ring Tone is presented.
7. Press the **[SAVE]** button.

Conditions

Programming

Station Data	1	Intercom/CO Differential Ring ID (PGM 124 – Index3-4)
	2	CO Differential Ring ID (PGM 124-Index4)

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 when a user dials or incoming call occurs from CO line.

- 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): AND is a feature in which AND digits are dialed automatically by the system after a CO Line is seized by means of CO Group Access code.
   − Automatic CO seize
5. In Call log, dialed digit is displayed on the station’s LCD.
6. The Digit Conversion Table can be applied by Apply Option (All/Station/CO Line/ Disable)

Programming

Station Data	1	Digit Conversion Table Index (PGM 131 – Index3)
CO Line 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 State Change Feature Code}**; the **{S-DN}** button LED illuminates.

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

1. Press **{S-DN}** button.
2. Dial the **{DND State Change Feature Code}**; the **{S-DN}** button LED is extinguished.

SLT

To activate DND:

1. Dial the **{DND State Change Feature code}**; a confirmation tone is received.

To remove DND:

1. Dial the **{DND State Change Feature code}**; a confirmation tone is received.

Conditions

1. A station will receive an error tone if not allowed 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. When calling a Station in DND, the Digital Phone display will indicate the DND status.
8. **{S-DN}** button LED in DND state flicks in reference with LED flashing rate of '[DN] DND'.

Programming

Station Data	1	DND Access (PGM 132 – Index4)
System Data	1	LED Flashing Rate (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:*

1. **[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}** or 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 Data	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 **[SPEAKER]** 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 (PGM 123).
4. The new Intercom box (LDP-DPB) has the **[CALL]** button.

Programming

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

Related Features

- Door Open

Hardware

3.35 Emergency Service

3.35.1 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.
3. Assigning emergency code, the emergency code with same dialed digit previous assigned cannot be as assigned.
4. In emergency code table, the field – tenant may be leaved empty. This emergency code with empty tenant will be adapted to all stations of all tenants.

Programming

Table Data	1	Emergency Code Table (PGM 258)

Related Features

Hardware

3.35.2 Forced CO Occupation for Emergency Service

Description

This feature allows the user to seize a CO line for emergency call in case all the CO lines are busy. For emergency call, the emergency code is dialed by the user and it is converted to access a CO line group. If all the CO lines are busy in that group, system searches for a CO line marked for emergency use. If all the CO lines are busy in that group, system check Emergency Force Service option, and if this option set as ON, system drops one of CO line in that group by force. And it makes an emergency call through CO line.

Operation

System

To dial an Emergency Call at the station when all CO lines are busy:

1. Lift handset or press **[SPEAKER]** button.
2. Dial the **{Emergency Code}** (ex., 911) without accessing CO Line.
3. Since all CO lines are busy, the system searches for a CO line marked for “Emergency Force Service”.
4. If there’s a CO group marked for emergency service, the system releases one of CO line in that group by force. And the emergency call will be made through that CO line.
5. Otherwise, system will release the first CO line by force and makes an emergency call.

Conditions

1. If the CO call is released by the system for emergency service, the system will wait for the duration of release guard timer, and then place the emergency call.

Programming

Numbering Plan	1	Emergency Alert, Feature Numbering Plan (PGM 113)
Station Data	1	Flex Button Assign (PGM 126)
CO Line Data	1	Incoming CO Release Guard Timer (PGM 166-Index8)
	2	Outgoing CO Release Guard Timer (PGM 171-Index5)
	3	Emergency Force Service (PGM 180)
Table Data	1	Emergency CO Table (PGM 258)
Tenant Data	1	Emergency CO Usage (PGM 281-Index9)

Related Features

Hardware

3.35.3 Emergency Alert

Description

When there is an emergency call from a station, it can be immediately notified to other stations. Notification involves audible ringing tone, LED flashing, and LCD display. In order to get this alarm information, the station should have a button programmed for emergency alert. After the emergency call is released, the users can retrieve the history for all the emergency calls from its terminal. Emergency call history can be deleted only from the attendant terminal.

Operation

Digital Phone

*To assign a flexible button as **{EMERGENCY ALERT}**:*

1. **[PGM]** + **{FLEX}** + Button Feature Type (1) + **{Emergency Alert Feature Code}** + **[SAVE]**

How the emergency call is alerted:

1. When an emergency call is tried from a station, it is immediately notified to the stations that have **{EMERGENCY ALERT}** button on its flexible buttons.

2. For emergency alert, **{EMERGENCY ALERT}** button will flash first. If the station is idle, it will receive a normal alarm ring and alarm message on LCD display. If the station is busy, it will receive a muted alarm ring only. Once it goes idle, alarm message will be shown on LCD as follows.

```
TOTAL:01 CNT:01 CODE:[777]
S:101      10/01 05:07
```

TOTAL: Total number of emergency logs
CNT: Current index of emergency log
CODE: Emergency code dialed by user
S: Station number having dialed the emergency code
10/01: Month and day (MM/DD)
05:07: Hour and minute (hh/mm)

How to clear Emergency Alarm:

1. Lift handset or press **[SPEAKER]** button.
2. Dial the **{Emergency Alert Feature Code}** or press the **{EMERGENCY ALERT}** button.

How to check the Emergency Call Log:

1. Lift handset or press **[SPEAKER]** button.
2. Dial the **{Emergency Alert Feature Code}** or press the **{EMERGENCY ALERT}** button.
3. Press the **[VOL UP]**/**[VOL DOWN]** buttons to select the Emergency Call Log to display.

How to delete the Emergency Call History (Only from Attendant):

1. Lift handset or press **[SPEAKER]** button.
2. Dial the **{Emergency Alert Feature Code}** or press the **{EMERGENCY ALERT}** button.
3. Press the **[VOL UP]**/**[VOL DOWN]** buttons to select the Call Log to display.
4. Press the **[SPEED]** button or the **[DELETE]** Soft key

Conditions

1. **{EMERGENCY ALERT}** button will flash until all emergency call history is deleted from attendant.
2. If the emergency alert is cleared by a station in the system, the emergency alert will be canceled for all stations.

3. Only SADN number can receive Emergency Alert ring.

Programming

Numbering Plan	1 Emergency Alert, Feature Numbering Plan (PGM 113)
Station Data	1 Flex Button Assign (PGM 126)

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 Directory Number (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 Data	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

<u>*To activate/deactivate Executive/Secretary forward from the Executive Phone*</u>

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).

Programming

Station Data	1	DND Access (PGM 132 – Index4)
Station Group Data	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 SLTM cannot be assigned as member of VM Group.

Programming

Station Group Data	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 Group Data	1	VM Group Assignment (PGM 200)
	2	VM Group Attribute Assignment (PGM 203)
Table Data	1	Voice Mail Dialing Table (PGM 269)
System Data	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 xxxxxxxxx sp yyyyyyyyy sp cr lf^Y
  − Type II message: cr lf **MD** ggg mmmm a xxxxxxxxx sp sp cr lf^Y
  − Type III message: cr lf **MD** ggg mmmm a sp yyyyyyyyy 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
mmmm	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 yyyyy
B	Called Station busy	Busy Mail	P#3P	II	B xxxxx yyyyy
C	Disconnect, connected party	disconnect	*****	II	C xxxxx yyyyy
D	Direct Fwd to VM group	Get Mail	P##	II	D xxxxx yyyyy
E	Error, invalid number	Error	P#*5P	II	E xxxxx yyyyy
H	Two-way Record	record	None	II	H xxxxx yyyyy
I	DND	DND	P#*6P	II	I xxxxx yyyyy
N	No Answer	No Answer	P#*4P	II	N xxxxx yyyyy
R	Direct CO/IP ring to VM group	AA	None	III	R xxxxx yyyyy

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. (PGM 203 – Index7)

Programming

Station Group Data	1	VM Group Assignment (PGM 200)
	2	VM Group Attribute Assignment (PGM 203)
System Data	1	Voice Mail Interface Select (PGM 223 – Index3)

Related Features

- AA/VM Group
- In-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 six 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).
6. When a conflicting Prefix is identified, the existing non-conflicting Numbering Plan is used until correctly updated.
7. If Numbering Plan type 7 is selected, all numbering codes are deleted; (for use when user wants to reconfigure all numbering codes,) first of all, user should assign the Prefix Numbering plan. After configuring the prefix, the user can assign Station Number, CO Group Access Code, Station Group Number, ACD Group Number and Feature code.

Programming

Numbering Plan

1 Numbering Plan Type(PGM 110)
2 System 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)
7 ACD Group Number (PGM 118)

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

Green Mode	1	Green Mode Activation (PGM 492)
	2	Green Mode Time Setting (PGM 493)

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 **{Speaker/Headset Mode}**.
3. Dial the appropriate selection,
   - 0: Speakerphone,
   - 1: Headset,
   - 2: Ear-Microphone.
4. Press the **[SAVE]** button.

To change the device to receive ring signals:

1. Press the **[PGM]** button.
2. Dial 62 **{Headset Ring Mode}**.
3. Dial the appropriate selection,
   - 0: Speakerphone,
   - 1: Headset,
   - 2: Both (Speaker & Headset).
   - 3: Ear-Microphone.
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 Data

1 Headset Ring (PGM 121 – Index3)

2 Headset Mode (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.
2. Dial the **{CO Line Access Code}** and the held CO line number.
3. Station is connected with the Held party.

Conditions

1. After placing CO 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, then the recalled call will be rerouted to Recall No-Answer destination of Incoming/Outgoing CO Alternate.

Operation

Hold Recall operation is automatic.

Conditions

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

Programming

CO Line Data	1	Incoming CO Alternative (PGM 169)
	2	Outgoing CO Alternative (PGM 173)
Table Data	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

<u>*To use Automatic Hold:*</u>

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.
4. Automatic Hold is operated when answering incoming calls. Pressing CO/IP button to make outgoing call is treated as transfer.

Programming

Station Data	1	Automatic Hold Access (PGM 123 – Index3)

Related Features

- Hold Recall

Hardware

3.42.4 Direct/Indirect Held CO Retrieve

Description

Held CO lines can be retrieved by using feature codes. If the user knows the held CO line number, the desired held CO line can be retrieve by using **{CO Line Access}** feature code (ex. “88”). But if they don’t know the number of held CO line, **{Held CO Retrieve}** feature code can be used. In this case, system will retrieve the oldest held CO first among those CO lines that were held by the retrieving station previously. If there’s no CO line held by the retrieving station, the oldest held CO line in the system will be retrieved. But, CO lines held exclusively by other stations cannot be retrieved.

Operation

To retrieve a held CO directly by specifying the desired CO line:

1 Lift the handset or press **[SPEAKER]** button.
2 Dial **{CO Line Access}** feature code. (ex. “88”)
3 Dial the held CO line number (01 ~ 80 for MG-100, 001 ~ 240 for MG-300)
4 If successful, the held CO will be retrieved.
5 Otherwise, the station will hear error tone.

To retrieve a held CO indirectly:

1 Lift the handset or press **[SPEAKER]** button.
2 Dial **{Held CO Retrieve}** feature code.
3 If successful, the held CO will be retrieved.
4 Otherwise, the station will hear error tone.

Conditions

1 The held CO line can be retrieved also by pressing the **{CO}** button or the associated **{LOOP-KEY}** button.
2 If there are multiple CO lines that were held by the retrieving station, the oldest held CO will be retrieved first regardless of the held mode (System Hold or Exclusive Hold).
3 If there’s no CO line that were held by the retrieving station, the oldest held CO among those in system hold mode will be retrieved.
4 The CO lines exclusively held by other station cannot be retrieved.
5 If the station has no access privilege to a CO line group, the held CO lines in the group cannot be retrieved.

Programming

Numbering Plan

1. CO Line Access Feature Code (PGM 113)
2. Held CO Retrieve Feature Code (PGM 113)

Related Features

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 delete existing 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 Agent may only logout from an activated Hot Desk phone.
3. The Flex button map of the Hot Desk station is changed upon log-in procedure and will take on the configuration associated with the Agent's station.
4. 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.
5. Each Hot Desk phone and Hot Desk user (Agent) requires a separate station number (DN) in the system.
6. When a Dummy Terminal seizes a SADN-type number (Sub-DN), the Hot Desk feature cannot be supported.
7. If an agent logs out without registering call forward destination, previous forward destination is automatically registered. if there is no previous forward destination, {Attendant call feature code} is registered as forward destination.

Programming

Numbering Plan	1	Hot Desk Feature Code (PGM 113)

Station Data	1	Dummy Terminal (PGM 121 – Index8)
	2	Station DN Type (PGM 130 – Index1)
	3	Station Password (PGM 131 – Index4)
	4	Hot Desk Agent Number (PGM 131 – Index8)
System Data	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 change the DND status 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 the MADN Flex button Access Option of other Station except the secretary station 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 Data	1	Station DN 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 Feature Code}**.

Conditions

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

Programming

Tenant Data	1	Redial Method (PGM 281 – Index2)

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 Data	1	Prime Number Button (PGM 123 – Index1)
	2	Flex Button Assignment (PGM 126)
	3	Station DN 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)
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 Data	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.

System allows programming the activation button of mobile extension.

A user can activate/deactivate mobile extension feature by pressing the activation button.

Operation

Digital Phone

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

1. Press the **[PGM]** button.
2. Dial 51 **{Mobile Extension Enable}**.
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 Number Program}**.
3. Dial mobile phone index (1 or 2).
4. Dial the mobile phone number with CO access code.
5. Press the **[SAVE]** button.

To register a mobile CLI:

1. Press the **[PGM]** button.
2. Dial 53 {Mobile Extension CLI Program}.
3. Dial mobile phone index (1 or 2).
4. Dial CLI digits.
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}**.
2. 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}**.

To program a Mobile Extension Status Change Button

1. Press **[Trans/PGM]** button and select the flexible button to program
2. Dial '1' to program number
3. Dial 'Mobile Extension Status Change' feature code
4. Dial index of Mobile Extension to program.(1-2)
5. Press **[Hold/Save]** button.

To activate/deactivate Mobile Extension feature.

1. Press **[Mobile Extension Status Change]** button

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. Station Group calls can be routed to the active Mobile Extension.
7. If **[Mobile Extension Status Change]** button is programmed without index of mobile extension, it is operated as first mobile extension index.

Programming

System Data	1	Mobile Attributes (PGM 236)
Station Data	1	Mobile Extension Access (PGM 132 – 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 AAFU or VMIB, the system can support three (3) languages simultaneously. Prompts in the desired languages are loaded into the VMIB/AAFU memory along with the Language Selection announcements. To let the callers select the language as they wish, the Language Selection announcement is played when an incoming call is programmed as DID, DISA, Auto Attendant or Station Hunt group announcement. The Language Selection announcement should be recorded 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

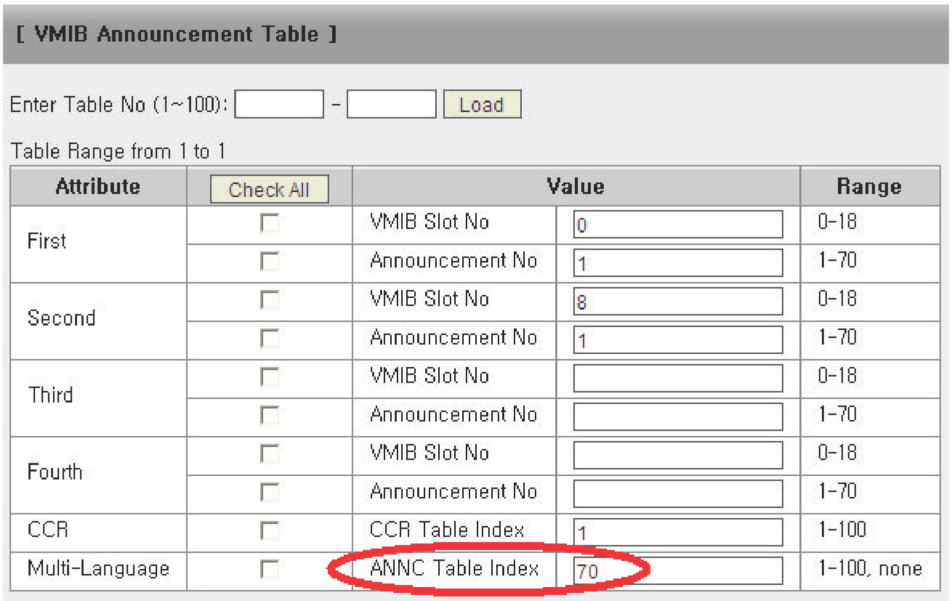
System automatically plays the Language Selection announcement and plays prompts in the selected language.

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

1. Press the **[PGM]** button.
2. Dial 062 **{Record VM Announcement}**.
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.

Programmingexample

Example) CO line 2 set as CCR with announcement 1. Announcement 1 has multi-language announcement 70.


[VMIB Announcement Table]

Enter Table No (1~100): – Load

Table Range from 1 to 1

Attribute	Check All	Value		Range
First		VMIB Slot No	0	0-18
		Announcement No	1	1-70
Second		VMIB Slot No	8	0-18
		Announcement No	1	1-70
Third		VMIB Slot No		0-18
		Announcement No		1-70
Fourth		VMIB Slot No		0-18
		Announcement No		1-70
CCR		CCR Table Index	1	1-100
Multi-Language		ANNC Table Index	70	1-100, none

1. Record the Multi-Language announcement in 70.
2. Program announcement table entry number 70 in PGM 259 to have the Multi-Language announcement 70. This is the default configuration. So, if it is not programmed before, there's no need to change it.
3. Program CO line 2 as CCR feature with announcement number 1.
4. In announcement table (PGM 259), set the Multi-Language as 70 for entry number 1.
5. With all these settings, if there's a call through CO line 2, system plays multi-language announcement 70 before playing announcement number 1. And when announcement number 70 is played, the caller can choose the preferred language by dialing a digit.
6. If the language is chosen, system finally plays the announcement number 1 for CCR in the selected language.

Conditions

1. Multi-language support is available with the VMIB/AAFU.
2. Separate announcements must be recorded by the Attendant for each language supported.
3. Multi-language announcement must be stored in announcement table (PGM 259) first. And then the announcement index can be programmed in other announcement entries for multiple language support.

Programming

Station Data	1	Station VMIB Prompt Language Index (PGM 145)
CO Line Data	1	CO VMIB Prompt Language Index (PGM 161 – Index8)
Table Data	1	Announcement Table (PGM 259)

Related Features

- VMIB Integrated Auto Attd/Voice Mail

Hardware

- VMFU and/or 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]** button.
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:*

1. **[PGM]** + **{FLEX}** + Button Feature Type (1), Number + **{VMIB Access Feature 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]** button.
2. Press the **{VMAILBOX}** Flex button.
3. Dial the Mailbox password.

Conditions

Programming

Related Features

- VMIB Voice Mail

Hardware

- Digital 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 **{Record VM Announcement}**.
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. The external music source should be connected to the BGM RCA jack on the front panel of the MPB.

Programming

Table Data	1	Tone Table (Web Admin. PGM 290 – Index49-61)
System Data	1	Music Assignment (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 Data (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 Ericsson-LG Enterprise 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.
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 (PGM 223 – 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:

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

3.56.1 Camp-On

Description

If a called station is busy, the calling station user may Camp-On until the called station user answers to his or her call. The Camp-On user will be in hold until the call is answered, and the called station user will hear the Camp-On Alarm tone.

Operation

Digital Phone/SLT

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

1. Dial the Camp-On digit, the called stations will receive Camp-On Alarm tone and the calling stations will receive Camp-On Hold 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. Camp-On may be set for any station user who is already being camped on.
3. If a station user is camped on by several other station users, the service will be provided in sequence.
4. If the calling station user deactivates the internal call, Camp-On is cancelled.
5. User can use this service by pressing the **[PGM]** and dialing **{Camp-On Register}** Code instead of one digit service.

Programming

System Data	1	Camp-On Access (PGM 133 – Index8)
Tenant Data	1	Intercom Busy One-Digit Attributes (PGM 237 – Index1)
Numbering Plan	1	Feature Numbering, Camp-On Register (PGM 113)

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 Alarm 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).
- Ignoring the Camp-Wait Alarm tone.

Operation

Digital Phone

To activate a Call Wait while receiving Intercom busy tone:

1. Dial the Call Wait digit, the called station will receive the Call Wait Alarm tone and and the calling station will receive the Call Wait Hold 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.

SLT

To activate a Call Wait while receiving Intercom busy tone:

1. Dial the Call Wait digit, the called station will receive the Call Wait Alarm tone and and the calling station will receive the Call Wait Hold 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. If the waiting extension user deactivates the internal call, Call Wait is cancelled.
3. Call Wait may not be set for any extension user who is already being waited.
4. User can use this service by pressing the **[PGM]** and dialing **{Call Wait Register}** Code instead of one digit service.

Programming

System Data	1	Call Wait Access (PGM 133 – Index7)
Tenant Data	1	Intercom Busy One-digit Attributes (PGM 237)
Numbering Plan	1	Feature Numbering, Camp-Wait Register (PGM 113)

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) and one User-defined Custom message. Several of the ten Pre-defined messages allow for auxiliary information such as a time, date or number.

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
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:

1. 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 the **[PGM]** button.
2. Dial 41 **{Display Message code}**.
3. Press the # key.
4. 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.

Table 3.57-1 Alphanumeric Entry Chart

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

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

Table 3.57-2 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
*	*				*				*
#	#				#				#

SLT

To activate a Display Message:

1. Lift the handset.
2. Dial **{PGM Mode Access}** code. Dial 41 **{display Message code}**, Or dial **{Pre-Selected Message PGM}** code.
3. Dial the Message number (0–9, or * for User's Custom Message).
4. Enter auxiliary input (hh:mm, mm:dd, etc. as needed).
5. Press the hook-switch; confirmation tone is heard.

To cancel an active Display Message:

1. Lift the handset.
2. Dial **{Pre-Selected Message PGM}** code.

To enter the User Custom Message (*):

1. Lift the handset.
2. Dial **{PGM Mode Access}** code.
3. Dial 42 **{Custom Message program 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. Press the # key.
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 enter the User Custom Message for other stations:

1. Press the **[PGM]** button.
2. Dial 052 **{Attendant Custom Message Program code}**.
3. Dial the desired Station range.
4. Enter the Message contents (up to 16 characters, refer to Alphanumeric Entry Chart).
5. 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.

Programming

Related Features

- Do Not Disturb (DND)
- Call Forward
- Speed Dial

Hardware

- Digital Phone required to receive Display Messages

3.58 Registering IP Devices & Registration 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 System Initialization Switch (Dip Switch 1) 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. Max Number of IP Phones (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 System Initialization Switch (Dip Switch 1) position.

Operation

Registration is automatic.

Conditions

Programming

System Info	1 IP Phone/Phontage Registration Table (PGM 106)
	2 Slot Assignment (PGM 104 – Index2)
	3 Logical Slot Assignment (PGM 103)
	4 DECT/IP/SIP Max Port (PGM 104)
SIP Station Data	1 SIP STA Basic Registration

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 System Initialization Switch (Dip Switch 1) position.

Operation

Registration is automatic.

Conditions

1. By default, the IP Phone Registration by STA Number is ON.
2. If some number was already registered to system, ip-phone with same number cannot be registered to avoid duplication.

Programming

System Info	1 Slot Assignment (PGM 101)
	2 Logical Slot Assignment (PGM 103)
	3 DECT/IP/SIP Max Port (PGM 104)
	4 IP Address Plan (PGM 108)
System Data	5 IP Phone Registration by STA Number (PGM 223)

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
- 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.

Programming

Zone Data	1	Zone Attributes (Web Admin)
	2	Zone RTP Relay Group (Web Admin)
	3	Inter-Zone Attributes (Web Admin)
	4	Station Zone Attributes (Web Admin)

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 103-Index2)
4. Max Number of IP Phones (PGM 104 – index 2)

Related Features

Hardware

3.61 Revertible Ring

Description

This feature can be used to identify the station 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 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. It will be operated regardless of DTMF Send Interval admin.

Programming

Numbering Plan	1	Speed Access (PGM 134 – Index1)
Table Data	1	System Speed Dial Table (PGM 257)
System Data	1	Pause Timer (PGM 220)

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 32 digits in length and may include special instruction codes. Range of BINs for station speed is 000 to 049.

Special instruction codes are:

- Flash as 1st digit: Activates dial tone detect.
  Flash **[FLASH]**: Activates CO Hook-Flash.
- 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.

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.

Table 3.62.2-1 Alphanumeric Entry Chart

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

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

Table 3.62.2-2 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 Station Speed Dial number using the 3-soft-key (LCD display phones):

1. Press the **{DIR}** Soft key.

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

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

```
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($1)

   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.

SLT

To dial using Station Speed Dial:

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

To program a Station Speed Dial number:

1. Dial **{Speed Program code}**.
2. Dial the Speed Dial bin number.
3. Dial the **{CO/IP Group Access code}**.
4. Dial the number to be stored.
5. Press for hook-switch.
6. If desired, enter a name (refer to Alphanumeric Entry Chart).
7. 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 32 characters in length and may include special instruction codes. Range of BINs for system speed is 2000 to 2999 (MG100) or 2000 to 3999 (MG300).

Special instruction codes are:

- Flash as 1st digit: Activates dial tone detect.
  Flash **[FLASH]**: Activates CO Hook-Flash.
- 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

Digital Phone

<u>*To dial using a System Speed Dial:*</u>

1. Lift handset or press the **[SPEAKER]** button.
2. Press the **[SPEED]** button.
3. Dial the desired bin number (MG100: 2000~2999, MG300: 2000~3999).

SLT

<u>*To dial using a System Speed Dial:*</u>

1. Lift handset.
2. Dial **{SLT Speed Dial access code}**.
3. Dial the desired bin number (MG100: 2000~2999, MG300: 2000~3999).

Attendant

<u>*To program a System Speed Dial number:*</u>

1. Press the **[PGM]** button.
2. Press the **[SPEED]** button.
3. Dial the Speed Dial bin number (MG100: 2000~2999, MG300: 2000~3999).
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.

Table 3.62.3-1 Alphanumeric Entry Chart

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

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

Table 3.62.3-2 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.

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

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

```
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($1)

   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 Number	1	Station Number Type (PGM 130)
Station Data	1	Flex Button Assign (PGM 126)
	2	DN Flex button Ring Option (PGM 126)
	3	DN Flex button Access (PGM 126)

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:

1. Station Group: Terminal / Circular / Ring / Longest Idle / VM
2. Pick-Up Group
3. Paging Group
4. Command Conference Group
5. PPT Group
6. Interphone Group
7. Pilot Group
8. ACD 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. In case of iPECS-MG300, max 50 groups and 50 members can be assigned. In case of iPECS-MG100, max 20 groups and 50 members can be assigned. There are 5 type station groups, and each group can be defined as Pick-Up group.

- Terminal Group –Call to Terminal Group always is directed to first member in group. If member is unavailable state or unanswered during member no answer timer, the call will be directed to the next member defined in the group. If last defined member is unavailable state or unanswered the call, call will be directed to the first member in group. If any member does not answer during Ring No-Answer Forward timer, call will be routed to defined forward destination. The queued call may be taken out of the group if set to forward to an overflow destination.
- Circular Group – Call to Circular Group is directed to one of member from first defined member to last defined member in order. If member is unavailable state or unanswered during member no answer timer, the call will be directed to the next member defined in the group. If last defined member is unavailable state or unanswered the call, call will be directed to the first member in group. If any member does not answer during Ring No-Answer Forward timer, call will be routed to defined forward destination. The queued call may be taken out of the group if set to forward to an overflow destination.
- Ring Group – Call to Ring Group is directed to all of member at the same time, and any member 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. If any member does not answer during Ring No-Answer Forward timer, call will be routed to defined forward destination. The queued call may be taken out of the group if set to forward to an overflow destination.
- Longest Idle (UCD) Group – Calls to Longest Idle Group is directed to the Station in the group that has been idle for the longest time. If member is unanswered during member no answer timer, the call will be directed to the next longest idle member in the group. If any member does not answer during Ring No-Answer Forward timer,

call will be routed to defined forward destination. 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.

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 Queuing Tone (either a ring-back tone, Voice announcement or MOH)
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 Station Group.
9. CCR feature can be applied during queued announcement service.

Programming

Station	1	DND Access (PGM 132 – Index4)
Station Group	1	Station Group (PGM 200)
	2	Station Group Greeting/Queuing Attribute (PGM 201)
	3	Station Group Attributes (PGM 202)
	4	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

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

Hardware

- Digital Phone

3.64.2 Greeting/Queuing Tone Service

Description

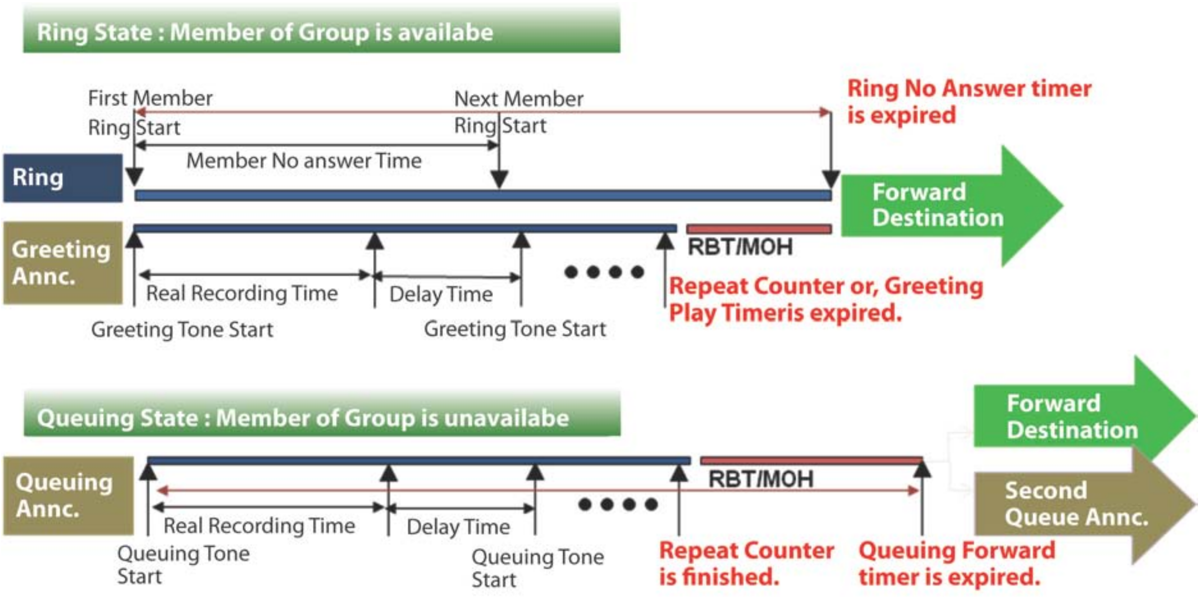
The System can provide a Greeting tone or Queuing tone when a call is routed to Station Group.

When a call is routed to Station 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 7 types of Tones:

- **NORMAL** – System Tone (01–19, Tone Freq. in PGM 264)
- **PROMPT** – VMIB Prompt
- **ANNOUNCEMENT** – VMIB Announcement
- **INT MOH**
- **EXT MOH**
- **VMIB MOH (1–4 for MG-300, 1–3 for MG-100)**
- **SLT MOH (1–5)**

Operation

If set, Greeting/Queuing Tone Service operation is automatic.



Conditions

Programming

Station Group	1	Station Group Attribute (PGM 201–202)
Table Data	1	Tone Frequency/Cadence (PGM 264)

Related Features

3.64.3 CCR Service with Queuing Announcement

Description

The System can provide CCR Service during queuing announcement according to the CCR option.

A CCR Table defines a dialed digit (0–9, #, and *) to a designated route; each individual digit corresponds with a route:

- Station
- Station Group
- ACD Group
- System Speed
- Voice Mail Access Code
- Announcement Table Code + VMIB Announcement
- Announcement Table and Drop Code + VMIB Announcement
- Conference Room
- Net number
- Attendant Call
- Internal Paging
- External Paging
- Internal/external All Paging
- Company Directory
- Record VM Greeting
- Digits

In addition, the System will monitor digits for a system numbering plan (eg. station number). If the User dials a Station number, Group Queuing Service is finished and a call is routed to the dialed destination.

Operation

If set, Greeting/Queuing Tone Service operation is automatic.

Conditions

1. SIP/ISDN Terminal does not support CCR feature**.**
2. If CCR Feature is activated, station group call service will be stop.

Programming

Tenant Data	1	Station Group Attribute (PGM 201-202)
Table Data	1	Tone Frequency/Cadence (PGM 264)
	2	Announcement Table (PGM 259)
	3	CCR Table (PGM 260)

Related Features

3.64.4 Forward Destination, Overflow Service

Description

This can be assigned as Station/Station Group/Telephone Number, covering Station Group Call according to the Forward type of the Station Group. There are 4 kinds of Forward type in a Station Group:

- Unconditional
- Queuing Overflow
- Queuing Timeout
- Queuing Overflow or Queuing Timeout

The Overflow Destination can be programmed as Station/Station Group/ACD Group/NET Destination/External number.

Operation

To use the Unconditional Forward Overflow Destination:

1. Dial the **{Station Group Number}**.
2. The Call is Routed to the Forward Destination.

To use the Queuing Overflow Forward Destination:

1. Dial the **{Station Group Number}**.
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 **{Station Group Number}**.
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 **{Station Group Number}**.
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	Station Group Attribute (PGM 201-202)

Related Features

3.64.5 Pilot Hunt Group

Description

Pilot Hunt group can support Call Cover functions and Call Forwarding function.

1. Call Cover: Pilot hunt group can be assigned as destination of DN Busy Service or destination of Incoming CO Alternative. According to Pilot Hunt call service type and service condition, other member can receive call in pilot hunt group.
2. Group Call Forward: If Pilot hunt group forward is set, all call of all members will be forwarded to defined destination in Day/Night/Timed mode

Operation

If programmed, Pilot Hunt Group is automatic.

To Register a Pilot Hunt Group Call Forward:

1. Lift the Handset or press **[SPEAKER]**.
2. Dial the **{Pilot Hunt Call Forward Register Feature Code}**.
3. Dial 1-4 **{Forward Code}** as appropriate.
4. Dial the station or station group to receive calls. Or, dial CO Group Access code and desired external phone number.
5. Press the **[SAVE]** button to save.

To Cancel registered Pilot Hunt Group Call Forward:

1. Lift the Handset or Press **[SPEAKER]**.
2. Dial the **{Pilot Hunt Call Forward Cancel Feature Code}**.
3. Dial 1-4 **{Forward Code}** as appropriate.
4. Dial the station or station group to receive calls, Or, dial CO Group Access code an desired external phone number.
5. Press the **[SAVE]** button to save.

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

3.64.6 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:

2. Dial **{Direct Call Pick Up}** feature code.
3. 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	1	Feature Numbering, Group Pick-Up (PGM 113)

Related Features
- Group Call Pick-Up

Hardware

3.64.7 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. Speak to all members.
3. 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

3.64.8 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	1	Feature Numbering Plan, Command Group (PGM 113)

Related Features

Hardware

3.64.9 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	1	Feature Numbering, Interphone Group Access (PGM 113)

Related Features

Hardware

3.64.10 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 (MG-300: 01–30, Mg-100: 01-15).

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	1	Feature Numbering, Internal Page Calling Answer (PGM 113)

Related Features

Hardware

3.64.11 Station Group Member Forward

Description

When station group call is routed to a member registered Call Forward, the following operation can be provided as programming data.

- No ring – Call for station group is not routed to the member set to forward.
- Forward – Call is routed to call forward destination.
- Ring – Call is routed to the group member.

Operation

Conditions

Programming

Station Group Data 1. Station Group Attribute (PGM 202-10)

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. 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.
2. 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 – Index9)
	2	SMDR Fraction (PGM 232 – Index10)
	3	Local Call Cost per Minute (PGM 232 - Index 18)

4 Long Call Cost per Minute (PGM 232 - Index 19)
5 International Call Cost per Minute (PGM 232 - Index 20)

Related Features

- Station Message Detail Recording (SMDR)

Hardware

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

1. When sending SMDR data by e-mail, e-mail server can require secure authentication. By adjusting SMTP Security option, secure authentication will be accomplished.

Programming

System	1	SMDR Attributes (PGM 232)
	2	SMDR E-Mail Option (Web Admin)

Related Features

- Call Cost Display

Hardware

3.65.3 SMDR Option For Second Information

Description

By default, the "START TIME" field in SMDR contains the second information. A new option is added to remove the second information in SMDR format. So, the "START TIME" field can be either 14 or 17 character long.

Operation

When "START TIME" field contains second information:

SMDR Data Format					
Field	**Start Column**	**Width**	**Numbers of Character**	**Format**	**Meaning**
NO	1	4	4	0001 (Range: 0001-9999)	Serial number of SMDR record
STA	6	8	2-8	10, 100, 1000, CO001	Station number (2,3 or 4 digits) CO number (CO xxx)
CO	15	3	3	001	CO Line number
TIME	19	8	8	01:00:10 (HH:MM:SS)	Call Line number
START TIME	28	17	17	25/05/10 14:20:10	Call Start Time
DIALED	46	25	1-25	O314504629	Call type and Dialed telephone number
CNT	72	5	1-5		The number of metering count
COST	78	11	1-11		Call Cost
ACCOUNT CODE	90	12	1-12		Account Code entered

<u>When "START TIME" field does not contain second information:</u>

SMDR Data Format					
Field	**Start Column**	**Width**	**Numbers of Character**	**Format**	**Meaning**
NO	1	4	4	0001 (Range: 0001-9999)	Serial number of SMDR record
STA	6	8	2-8	10, 100, 1000, CO001	Station number (2,3 or 4 digits) CO number (CO xxx)
CO	15	3	3	001	CO Line number
TIME	19	8	8	01:00:10 (HH:MM:SS)	Call Line number
START TIME	28	14	14	25/05/10 14:20	Call Start Time
DIALED	43	25	1-25	O314504629	Call type and Dialed telephone number
CNT	69	5	1-5		The number of metering count
COST	75	11	1-11		Call Cost
ACCOUNT CODE	87	12	1-12		Account Code entered

Conditions

1. "Second Info Print" is effective both in On/Off-line SMDR data print and in SMDR view/download feature which is available in the "System Management > SMDR" of Web Admin. But, SMDR interface data is not affected by this admin.

Programming

System Data 1 Second Info Print (PGM 232 – Index 17)

Related Features

- Call Cost Display

Hardware

3.66 System Admin Programming

3.66.1 Keyset 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 Keyset 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

Keyset 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 Data	1	Admin. Access (PGM 121 – Index5)
System Data	1	Admin. Access Authority (Web Admin.)
	2	System Password – Admin Password (PGM 226 – index 2)

Related Features

- Web Administration
- Multi-Level Admin Access

Hardware

3.66.2 Multi-Level Admin Access

Description

System admin data can be protected by password when access the system. According to 3 level ID and password, authority can be defined to access database.

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.

- Maintenance Level

  Only one ID and Password can be supported, The Maintenance level has access to the entire database, all maintenance routines, and defines the Admin.

- Administrator Level

  3 ID and Password can be supported for Administrator. The Admin password has access to specific database items as well as Station Program pages.

- User Level

  3 ID and Password can be supported for USer. A User password can only access specific database items and cannot access Station Program pages.

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 Keyset Admin.
2. Admin. Access Authority applies to all Admin. access whether accessed via a Digital Phone, ISDN or IP channels.

Programming

Station Data	1	Keyset Admin (PGM 121 – Index5)
System Data	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 Ericsson-LG Enterprise 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 Data	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 Ericsson-LG Enterprise R&D STI.

Operation

Operation is detailed in the Admin. Programming Manual.

Conditions

1. Access to the User Guide is not password protected.

Programming

Related Features

- Web Administration

Hardware

3.67 GDC-500H (DECT Protocol Phone) – EU & NA version

Description

The iPECS-MG System supports office building mobility employing Digital Enhanced Cordless Telecommunications (DECT). Ericsson-LG Enterprise's DECT Base stations (GDC-400B/GDC-600B/GDC-600BE) connect to the Wireless Telephone Interface Module (WTIB). The WTIB4/WTIB8 manage up to 4/8 base stations; up to 2/3 WTIBs may be installed in the iPECS-MG100/300 System respectively. DECT handsets (GDC-400H, GDC-450H and GDC-500H) 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. WTIB boards should be installed in only one cabinet
2. GDC-500H has two versions, EU and NA version.

Programming

DECT Data	1	DECT Registration (#0)
	2	DECT Attributes (PGM 491)

Related features

Hardware

- WTIBs (WTIB4, WTIB8)
- GDC-400B/GDC-600B/GDC-600BE Base stations
- GDC-400H/GDC-450H/GDC-500H 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.

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 NAPT, 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.

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 System Numbering Plan.

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 Ericsson-LG Enterprise Distributor); there is a type of software lock key for QSIG/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.

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:

1. 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 Network Numbering Plan 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 Numbering Plan 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.

Programming

Numbering Plan	1	System Numbering Plan (PGM 111)
CO Line Data	1	CO line Attribute (PGM 160, PGM 161)
Station Data	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. System determines the method according to the status of CO line automatically.

Operation

<u>To perform a Screened Transfer to another Station on a different Networked System:</u>

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.

<u>To perform an Unscreened Transfer to another Station on a different Networked Station:</u>

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.

Operation

To perform a CCBS (Call Back):

1. **Dial the Sta**tion of the other System.
2. When a busy tone is received, the User can either,
   1. Press the **[CALL BACK]** button.
   2. Press **[TRANS]** button and dial **{Call Back Register Feature Code}**. Or, 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 Call Back Indication Ring Timer (PGM 265-9), the call will be cleared.
5. There are two modes: Connection Mode and Connectionless Mode; this can be set at PGM 320 – Index 8.
6. CCBS is supported with VoIP CO line.

Programming

Voice Network	1	Network Basic Attribute (PGM 320)
	2	Network Numbering Plan (PGM 321)
Numbering Plan	1	System Numbering Plan (PGM 111)
	2	Feature Numbering Plan (PGM 113)
System Data	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 Call Wait 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, the muted ring is changed to a normal CO ring, the current call is disconnected; the offered call 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	1	Network Basic Attribute (PGM 320)
	2	Network Numbering Plan (PGM 321)
Numbering Plan	1	System Numbering Plan (PGM 111)
	2	Feature Numbering Plan (PGM 113)
System Data	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 cans conference-in a party on any other node.

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

Programming

Voice Network	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 Internal message wait service. 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,
   - Press the **[MSG/CALL BK]** or **{Message Wait Register Feature Code}** button, Or
   - Press the Hook Flash and dial the **{Message Wait Register Feature Code}**-SLT.
3. MWI is left to the called station, and a flashing **[MSG/CALL BK]** button indicates the message waiting.
4. At the called Station, press the flashing **[MSG/CALL BK]** 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	1	Network Basic Attribute (PGM 320)
	2	Network Numbering Plan (PGM 321)
Numbering Plan	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 the status of CO line automatically.

Operation

To activate Net Call Forward:

1. Lift handset, or press the **[SPEAKER]** button.
2. Press **[FWD]** button or Dial **{Call Forward feature code}**.
3. Select forward type (0~4)
4. Dial the Station Number to forward calls (register).
5. Press the **[SAVE]** button.
6. Press * or # key.

To deactivate Net Call Forward:

1. Lift handset, or press the **[SPEAKER]** button.
2. Press **[FWD]** button or Dial **{Call Forward feature code}**.
3. Press # key.

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:

1. 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	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	1	Attendant Attribute (PGM 272 – Index3, Index5)
	2	Night Attendant Group Assign (PGM 275)
	3	Digit Conv. Table (PGM 251, PGM 252)
Voice Network	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 manger 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 Centralized Voice Mail

Description

This function can support that all voice mail occurred in all systems can be recorded in an external VMS.

Operation

Refer to the External Voice Mail function in the system features.

Conditions

1. The number of the centralized VMS should use the representative number of voice mail group created in master system.
2. The numbering plan including the representative of mail access assigned in master system should be included in the numbering plan of QSIG group in slave system.

Programming

Voice Network	1	Network Numbering Plan Table (PGM 321)

Related Features

Hardware

3.68.2.15 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, 7)
	2	Network Numbering Plan Table (PGM 321 – Index8, 9)

3.69 Traffic Analysis

Description

If Traffic Operation option set as ON, 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 Traffic Operation (PGM 223)
2 RS-232 Port Settings (PGM 230)
3 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
- ACD 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 (PGM 223 – Index9)
	2	DB Download Time (PGM 223 – 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

Any user can answer incoming CO by dialing a UA code if CO line is defined as Universal Answer Line. Digital Phones may program a Flex button as a **{UA}** button.

Operation

Digital Phone

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

1. 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, 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
- ACD Group
- System Speed Dial
- Voice Mail Access Code
- CCR Access Code + VMIB Announcement
- CCR Access and Drop Code + VMIB Announcement
- Conference Room
- Net number
- Attendant Cal
- Internal Paging
- External Paging
- Internal/External Paging
- Company Directory
- Record VM Greeting
- Digits

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. To record or delete an Auto-Attendant message, all of the VMIB channels must be in the idle state.
4. 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.
5. 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.
6. If the external caller dials more than a single digit, the call is routed based on the System Numbering Plan.
7. Calls answered by an Auto-Attendant (CCR) Announcement are interactive DISA calls and are subject to conditions of a DISA call. This is valid only when DISA is set ON in CCR Table (PGM 260).
8. A CCR Announcement may be programmed to disconnect the call after playing.
9. The Auto-Attendant Announcement feature is supported for DISA and DID calls.
10. 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.

11. The remote Caller's voice mailbox access can be supported by assigning the **{VMIB Access Code}** to a CCR destination.

Programming

Tables Data	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 Length (PGM 220 – Index8)
Numbering Plan	1	VMIB Access (PGM 113)
Station Data	1	Station VMIB Attributes (PGM 145, 147)

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 **{VIMB Access Feature Code}**, by pressing the [MSG/CALL BK] 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:*

1. 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/CALL BK] button and dial 2 to select VMIB Messages; the Mail Box & Password prompts will be played. Or, press **{VMAILBOX}** button.
3. Enter the Mailbox (Station number) and corresponding password if VM Password Input is "Password"; if entry is valid, the Number of Message prompt will be presented. If VM Password Input is "No Password", the Number of Message prompt will be heard without any user input.
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. 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 “No Message” prompt.
2. If the dialed number is not recognized, the “Invalid Entry” prompt is played; after the fourth 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 Data	1	Station VMIB Attributes (PGM 145, 147)
Numbering Plan	1	VMIB Access (PGM 113)
System Data	1	VM COS Attributes (PGM 243)

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, dial a Station Group number assigned for answer by VIMB Auto-Attendant.
3. Upon answer, dial **{VIMB Access Feature Code}**; the Request for Mail Box Number prompt will be presented.
4. 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 Data	1	Station VMIB Attributes (PGM 145, 147)
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.

Table 3.74.3.4-1 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 Inter-Digit 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.

Programming

Station Data	1	Station VMIB Attributes (PGM 145, 147)
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 can be kept in VMIB after being attached to the e-mail. In this case, the voice message must be deleted from the VMIB even if the e-mail is deleted. Or optionally, voice messages can be deleted automatically from the VMIB after being attached to the e-mail.
2. The e-mail will be sent to the address assigned for the Station with the Sender address defined for the station.

NOTE
The Sender address must be defined, as many e-mail servers will reject anonymous e-mails.

1. The e-mail address for the station can be defined under the Web Admin.
2. The Voice message can be attached to the e-mail notification as a .wav file, if the Attach Message option is set as “Attach only”. If it is set as “Attach & Delete”, the voice message is attached to e-mail and then deleted from VMIB after successful delivery. If disabled, the notification e-mail does not include an attached .wav file.
3. The format of SMTP Mail Server Address can be IPv4 address-form or URL-form. To use the URL-form address, DNS IP Address (PGM 108-5) must be configured beforehand.
4. There are 3 SMTP security options: ‘No security’, ‘SSL’ and ‘TLS’. ‘No security’ is defined initially. If e-mail server require secure authentication, this option must be configured properly according to authentication type.

Programming

Pre-Programmed Data	1	IP Address Plan (PGM 108)
Station Data	1	Station VMIB Attribute (PGM 145, 147)
	2	SMTP Mail Server Address (Web Admin.)
		VM MSG - SMTP Mail Server Address (Web Admin.)
		VM MSG - SMTP Port Number (Web Admin.)
		VM MSG - SMTP Security (Web Admin.)
		VM MSG - SMTP User Mail Address (Web Admin.)
		VM MSG - SMTP Mail Server ID (Web Admin.)
		VM MSG - SMTP Mail Server Password (Web Admin.)
		VM MSG - SMTP Mail Sender Address (Web Admin.)
		VM MSG - Attach Message Option (Web Admin.)

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 you 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 To record a new greeting, press To return to the Main Menu, press 9.").
2. Dial 5 to hear your Greeting. Or, dial 7, the Record Greeting prompt is presented ("At the tone, record your new greeting, press # when done.")
3. After the beep, record Greeting speaking in a normal voice.
4. When finished, press the # key, the Greeting Confirmation prompt is presented ("Your greeting is saved.")

Conditions

1. If the User is external (remote), the User must begin dialing within the Inter-Digit timer, 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 and dial subsequent digits within the Inter-Digit Timer; if not the call is released.
2. Mailbox Set Forward is available for both internal and external users.

Programming

Station Data	1	Station VMIB Attributes (PGM 145, 147)
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 VMIB 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 (PGM 145 – Index7)

Related Features

Hardware

3.74.3.11 Voice Mail COS

Description
Each voice mail can have its own class of service so that it allows for different parameters for each voice mail such as greeting length, message length, number of messages and etc.

Operation

Parameters that can be set for each voice mail class of service

1. Greeting Length (00 - 99 seconds): Maximum user greeting time can be set.
2. Message Length (001 - 999 seconds): Maximum recording time for each voice mail message can be set.
3. Number of messages (001 - 250): Maximum number of voice mail messages.
4. Retention time (00 - 99 days): Voice mail messages will be automatically deleted after this amount of days.
5. E-mail notification (On/Off): E-mail notification can be enabled or disabled.

Conditions

1. There are 5 different classes of service for voice mail.
2. Each directory number is assigned one of the five COS's for voice mail.

Programming

Station Data	1	VM COS (PGM145 - Index20)
System Data	1	VM COS Attributes (PGM 243)

Related Features

Hardware

3.74.3.12 Announce Only Mailbox

Description
This feature provides a method to mark a mailbox as an announce-only type. This type of mailbox plays a greeting only and then returns the caller to the previous menu of CCR feature or hangs up. So, no voice mail message can be saved in this type of mailbox.

Operation
Previous Menu option (CCR Flow)

1. Caller reaches to a menu in CCR table and calls the station in forward to voice mail.
2. When the call is forwarded to voice mail, the caller hears the greeting of mail box.
3. If the greeting is finished, the caller goes back to the CCR menu of step 1. So, the caller will hear the CCR announcement again for another selection.

Previous Menu option (non-CCR flow)

1. Caller calls the station in forward to voice mail.
2. When the call is forwarded to voice mail, the caller hears the greeting of mail box.
3. If the greeting is finished, the outside caller follows the busy destination and the internal caller hears error tone.

Hang-up option

1. Caller calls the station in forward to voice mail.
2. When the call is forwarded to voice mail, the caller hears the greeting of mail box.
3. If the greeting is finished, the call is disconnected and the caller will hear error tone.

Conditions
1. To use previous menu, the caller must call the station by using CCR feature.

Programming
Station Data	1	Announce only Option (PGM145 - Index17)

Related Features

Hardware

3.74.3.13 Voice Mail Message Cascade

Description

Message cascading is a feature that copies voice mail messages left for one mailbox to another mailbox. Once the message is copied into the other mailbox, the other mailbox will receive the notification of message receipt.

Operation

Voice Mail Message Cascade feature is automatic if the feature is enabled:

Conditions

1. The chain of message cascading is not allowed. So, the message cascade happens only one for each new voice mail message.

Programming

Station Data	1	Station VMIB Attribute (PGM145)

Related Features

Hardware

3.74.3.14 VM Password Input

Description

Password input method to access voice mailbox can be configured as follows.

1 DN number + password

2 Password

3 No password

We will call the case 1 as ‘authorization code’.

Operation

DN number + password (authorization code)

Users can access to voice mailbox by dialing an DN number and its password.

With this mode, it is possible to access to other DN’s voice mailbox if you know the password of the DN.

Password

To access to voice mailbox, users should enter only the password of DN which is active currently.

No Password

A user can access to only its own voice mailbox without entering password.

Conditions

1 If a password is not registered, you cannot access to voice mailbox in the configuration requesting authorization code or password.

2 Password should be entered when a voice mailbox of which VM password input is No Password is accessed through remote mode(by dialing ‘*’ during user greeting).

Programming

1. VM Password Input (PGM 147-5)

Related Features

Hardware

3.74.3.15 VM Private Message

Description

When a caller leaves a voice mail message, the message can be marked as a private message. If the voice message is marked as private, the message cannot be transferred to other station.

Operation

To mark a voice message as private:

1 Record the desired message after hearing the user greeting and beep tone.

2 Dial '#' after message recording is finished.

Or

Dial '*' for further options and then dial '#'.

3 The following prompt will be heard.

"For regular delivery, press one. To mark urgent, press two.

To mark private, press three. To mark urgent and private, press four".

4 Dial '3' for a normal delivery in private option

Or

Dial '4' for an urgent delivery in private option

Conditions

1 This feature needs recording of new VMIB prompts. In addition, "Enhanced VM Features" must be enabled in System Attributes (PGM 223 – Index 20).

2 In case the caller is an internal station, "Private Message Mark" attribute must be set ON in VM COS (PGM 243 – Index 8). Otherwise, private message cannot be left.

3 If the caller is an external party, it is always possible to leave a private message.

Programming

Related Features

Hardware

- VMIB

3.74.3.16 VM Message Delivery Confirmation

Description

This feature provides a way to allow a mailbox owner to mark a message for confirmation of delivery. When the user has listened to the sent message, a message is dropped in the sender's mailbox confirming listen receipt.

Operation

To mark a voice message as delivery confirmation:

1 Record the desired message after hearing the user greeting and beep tone.

2 Dial '#' after message recording is finished.

Or

Dial '*' for further options and then dial '#'.

3 The following prompt will be heard.

"For regular delivery, press one. To mark urgent, press two.

To mark private, press three. To mark urgent and private, press four

To request delivery receipt of the message for future, press 5"

4 Dial '5' to set delivery confirmation on the message.

When the voice message set for delivery confirmation is checked by the receiver:

1 A confirmation message is sent back to the sender's mailbox.

2 The sender will see the notification of message through LCD display or LED button.

3 If the sender accesses the mailbox, the following message will be played.

"Message for XXX was listened to on HH:MM MM/DD".

XXX stands for mailbox number or recorded name while HH:MM and MM/DD are the time and date information, respectively.

Conditions

1 This feature is available only if the caller is an internal party.

2 This feature needs recording of new VMIB prompts. In addition, "Enhanced VM Features" must be enabled in System Attributes (PGM 223 – Index 20).

3 "Confirm Message Receipt" attribute must be set ON in VM COS (PGM 243 – Index 7) for the sender's VM COS. Otherwise, it is not available and the corresponding prompt will not be heard.

4 The confirmation message is treated as a normal VM message and all options apply except for attach, pause, resume, rewind and fast forward.

5 The name of mailbox can be recorded by using the **{Record VM Subscriber Name}** feature code.

Programming

Numbering Plan	1. Record VM Subscriber Name Feature Code (PGM 113)
Station Data	1. VM COS (PGM 145 – Index 20)
System	1. Enhanced VM Features (PGM 223 – Index 20)
	2. Confirm Message Receipt (PGM 243 – Index 7)

Related Features

Hardware

- VMIB

3.74.3.17 VM Message Future Delivery

Description

This allows a user to record a message and have it sent to another mailbox at a specific date/time.

Operation

To mark a voice message as future delivery:

1 Record the desired message after hearing the user greeting and beep tone.

2 Dial '#' after message recording is finished.

Or

Dial '*' for further options and then dial '#'.

3 The following prompt will be heard.

"For regular delivery, press one. To mark urgent, press two.

To mark private, press three. To mark urgent and private, press four

To request delivery receipt of the message for future, press 5

For future delivery, press 6"

4 Dial '6' to set future delivery on the message.

5 System will play the prompt like

"Enter date and time and press one of the following options, 1 for AM, 2 for PM".

6 User dials 4 digits MM/DD and 4 digits HH:MM.

Valid input ranges are like the following.

Month: 01 ~ 12, Day: 01 ~31, Hour: 00 ~ 11, Minute: 00 ~ 59.

7 And finally, 1 digit for AM or PM should be dialed.

For example, if user dials 0903 0830 1 for MM/DD, HH:MM, and AM or PM, it means Sep 3rd 8:30 AM.

8 If the user finishes dialing, message will be sent to the destination in future.

Conditions

1 This feature needs recording of new VMIB prompts. In addition, “Enhanced VM Features” must be enabled in System Attributes (PGM 223 – Index 20).

2 “Future Delivery Message” attribute must be set ON in VM COS (PGM 243 – Index 6) for the receiver’s VM COS. Otherwise, it is not available and the corresponding prompt will not be heard.

3 If the call is finished without setting the delivery time properly, the voice message is sent to the called party immediately. But if the delivery time is not set correctly in sending message feature in mailbox, the recorded message is not sent to the destination.

4 If the caller doesn’t dial a valid digit stream, invalid and retry prompt will be heard. And then the caller can try again. But there’s no limitation in retrial count.

Programming

Station Data	1. VM COS (PGM 145 – Index 20)
System	1. Enhanced VM Features (PGM 223 – Index 20)
	2. Future Delivery Message (PGM 243 – Index 6)

Related Features

Hardware

3.74.3.18 VM Message Fast Forward/Rewind

Description

While listening to the voice message left to mailbox, the user can change the position of playback of voice message back and forth as wanted. The playback of the message is relocated as the programmed amount of time.

Operation

To forward the voice message while hearing:

1 Dial ‘#’ while the message is played.

2 System will fast forward the message as the programmed amount of time.

To rewind the voice message while hearing:

1 Dial ‘*’ while the message is played.

2 System will rewind the message as the programmed amount of time.

Conditions

1 This feature is available only for the voice messages left in the mailbox. And only VMIB supports the feature.

2 Delivery confirmation message doesn’t support fast forward and rewind features.

Programming

Related Features

Hardware

- VMIB

3.74.3.19 VM Message Pause/Start

Description

The playback of voice message in a mailbox can be paused and resumed later by dialing pause / start command.

Operation

To pause the playback of a voice message while being played:

1 Dial '8' while the message is played.

To resume the paused playback of a voice message:

1 Dial '8' in a paused status.

Conditions

1 This feature is available only for the voice messages left in the mailbox. And only VMIB supports the feature.

2 Delivery confirmation message doesn't support pause and start feature.

3 When the playback is paused, system does not resume the playback automatically. So, the user must resume or finish the playback manually.

Programming

Related Features

Hardware

- VMIB

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 feature code}**.
2. Select type 1
3. 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 feature code}**.
2. Select type 2
3. The Station number announcement for Station is heard, “This is station 150.”

To hear Station Settings:

1. Dial **{System Voice Memo feature code}**.
2. Select type 3
3. Status for the Station is reported. Items reported are as follows:
   - Station ICM Mode (Handsfree/Tone/Privacy)
   - Station IP Address
   - Station Mac Address
   - Listed message x (x: number of all messages waiting)
   - Wake-Up Time (‘hh:mm’)
   - Do Not Disturb
   - Forward Setting
   - Queued CO/IP xx
   - Locked (temporary COS change)
   - COS x

SLT

To hear Date & Time Prompt:

1. Lift the handset.
2. Dial **{System Voice Memo feature code}**.
3. Select type 1
4. 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 feature code}**.
3. Select type 2
4. Announcement for Station is heard, “This is station 150.”

To hear Station Settings:

1. Lift the handset.
2. Dial **{System Voice Memo feature code}**.
3. Select type 3
4. Status for Station is reported. Items that will be reported are as follows:
   - Station ICM Mode (Handsfree/Tone/Privacy)
   - Station IP Address
   - Station Mac Address
   - Listed message x (the number of all messages waiting)
   - Wake-Up Time (‘hh:mm’)
   - Do Not Disturb
   - Forward Setting
   - Queued CO/IP xx
   - Locked (temporary COS change)
   - COS x
   - Preselected Message

Conditions

1. For Station Status, items from “Listed message x” to “Preselected Message” will be not be announced if not active.

Programming

Station Data	1	VMIB Access (PGM 145 - Index1)
Numbering Plan	1	Feature Numbering Plan, System Memo (PGM 113)

Related Features

Hardware

3.74.5 CCR without DISA

Description

When CCR (Customer Call Routing) feature is used, DISA is also enabled automatically. So, if there's nothing programmed in the CCR table for a digit input or there're multiple digit inputs from the caller, input digit stream is regarded as DISA input by the system and processed as a normal DISA procedure. But with this new option, DISA can be disabled for a specific digit input in CCR table so that only CCR feature is available for a digit input.

Operation

When CCR digit has a value in CCR table and its DISA is disabled

1. When the caller dials CCR digit, the call is routed to the destination immediately.

When CCR digit has a value in CCR table and its DISA is enabled

1. When the caller dials CCR digit, system waits for another digit input since DISA is enabled.
2. If the caller doesn't dial more digits and CCR inter-digit timer expires, the call is routed to the CCR destination.
3. If the caller dials more digits and CCR inter-digit timer expires, the input digits are analyzed and the call is routed according to the result of digit analysis.

When CCR digit has no value in CCR table and its DISA is disabled

1. When the caller dials CCR digit, the caller hears "Invalid" and "Retry" prompt until retry count overflows.

When CCR digit has no value in CCR table and its DISA is enabled

1. When the caller dials CCR digit, system waits for another digit input since DISA is enabled.
2. If CCR inter-digit timer expires, the input digits are analyzed and the call is routed according to the result of digit analysis.

Conditions

1. Each entry in CCR table can be programmed to allow or disallow DISA.
2. If DISA is allowed in CCR table, the operation is just the same as the original CCR feature in which CCR and DISA is enabled at the same time.

Programming

Table Data 1 CCR - Allow DISA (PGM 260 – Index1~12)

Related Features

Hardware

- AAFU and/or VMIB

3.74.6 CCR Reroute Destination

Description

When a call goes to a destination via CCR feature, the call can be rerouted according to the status of the destination. When the original destination is busy, in DND, out of service or doesn't answer the call, the call can be rerouted.

Operation

When CCR table doesn't have an index for System Reroute Table (PGM 244):

1. A call is routed to a destination by using CCR feature.
2. The destination is busy, in DND, out of service or doesn't answer the call so that call rerouting should be performed.
3. Then, the call will be rerouted according to "Incoming CO Alternative Destination (PGM 169)" that can be programmed on a CO line basis.

When CCR table has an index for System Reroute Table (PGM 244):

1. A call is routed to a destination by using CCR feature.
2. The destination is busy, in DND, out of service or doesn't answer the call so that call rerouting should be performed.
3. Then, the call will be rerouted according to "System Reroute Table (PGM 244)" that can be commonly used by each CCR table entry

Conditions

1. By default, each entry in CCR table doesn't have an index for the "System Reroute Table (PGM 244)".
2. This feature is available only for CO line calls.
3. The situations for call rerouting are the same as those in "Incoming CO Line Alternative Destination (PGM 169)".
   - Busy Case
   - No-Answer Case
   - Vacant Number Case
   - Transfer No-Answer Case
   - Recall No-Answer Case
   - DND Case
   - Out Of Service Case
   - Error Case.
4. The programmable destinations for CCR rerouting are the same as those in "Incoming CO Line Alternative Destination (PGM 169)"
   - Disconnect
   - Attendant
   - CO Ring Assign
   - Alternative Ring Table
   - Tone
   - Pilot Hunt
   - Ring or Transfer Station (Transfer Call Only)

Programming

Table Data	1	System Reroute Table Index (PGM 260 – Index 13)
System Data	1	System Reroute Table (PGM 244)

Related Features

Hardware

- AAFU and/or VMIB

3.74.7 Rerouting From Voice Mail Forwarding

Description

This feature allows the caller to select another destination when the call is forwarded to a person's voice mail.

Operation

Digital Phone

To set the rerouting destination from voice mail forwarding:

1. Press **[PGM]** button and Dial 77.
2. Enter the rerouting destination and press **[SAVE]** button.
3. If it is successful, service set tone will be heard.
4. Otherwise, error tone will be provided.

To reset the rerouting destination from voice mail forwarding:

1. Press **[PGM]** button and Dial 77.
2. Press **[SAVE]** button.
3. Service set tone will be heard.

To select another destination

1. A call reaches to a station and it is forwarded to the voice mail.
2. The caller hears the personal greeting.
3. Before greeting is finished and "Begin Recording" prompt is played, the caller dials '0'.
4. If the voice mail owner has a reroute destination for voicemail forward, the call is routed to the destination.
5. Otherwise, the caller can leave the voice mail.

Conditions

1. The rerouting destination can be one of the following.
   - Station Number
   - Station Group Number
   - ACD Group Number
   - Attendant Call
   - Networking Number

2. Internal and external calls can be rerouted by using this feature.

Programming

Station Data	1	VM Forward Reroute Destination (PGM 147 – Index 1)

Related Features

Hardware

- AAFU and/or VMIB

3.74.8 Voice Mail Service Retrial

Description

When there is no channel available in AAFU, AAIB, or VMIB, it is not possible to play the voice prompt, greeting or message. If the call comes through a CO line, the call can be put in the wait queue for the voice mail channel to be idle. Or, optionally, the call can be rerouted immediately to the busy destination. (PGM 169)

Operation

Conditions

1 This feature is available only on CO lines.

And this applies to all CO line types (normal, DID, or DISA)

2 The CO call is put on waiting queue and the caller will hear ring-back tone.

3 System will wait for 3 seconds and try again to start the VM service until the retry count (PGM 161 – Index 13) overflows.

4 The following features will support the retrial of services from CO line.

- CCR (Customer Controlled Routing)
- CCR Drop
- VM Message Retrieve
- VM Message Store
- AME (Answering Machine Emulation)
- System Greeting Recording
- Company Directory

Programming

CO Line Data	1.	VM Service Retry Count (PGM 161 – Index 13)
	2.	Incoming CO Alternate Destination (PGM 169)

Related Features

Hardware

3.74.9 VM Features

3.74.9.1 Temporary Greeting

Description

This feature provides a method for a mailbox owner to record a temporary greeting play once recorded. This will not affect the standard mailbox greeting.

Operation

The user is given an option to record a temporary greeting. Once recorded, the greeting is enabled and played when a caller goes to the mailbox.

Once the greeting is deleted, the standard greeting is resumed.

When the temporary greeting is active, a prompt plays temporary greeting active when a mailbox owner accesses the mailbox.

Press 8 from the main menu and then press 1 to access the greetings.

A new prompt will state, “to listen to your current greeting press 5, to record a new greeting press 7, to access your temporary greeting press 8, to return to the main menu press 9”

Press 8 to record temporary greeting.

- To listen, press 1
- To record, press 2
- To delete, press 3

Once the temporary greeting is recorded, a new prompt will state “your temporary greeting is recorded and active”

When the user accesses his mailbox with a temporary greeting active, the prompt “your temporary greeting is recorded and active” will be played prior to the main menu prompt.

Conditions

1. A mailbox can only have one temporary greeting.
2. Once the temporary greeting is recorded, it becomes active until it is deleted.

Programming

Related Features

Hardware

3.74.9.2 PERSONAL OPERATOR PER MAILBOX ENHANCEMENT

Description

This feature provides more destination choices in PGM 147.

Operation

CCR menus 001-100 and Station/System Speed Dial entries are added for the destination.

Conditions

1. CCR menu is entered by **{Announcement Table}** feature code + (001-100).
2. Station Speed Dial entry is entered by **{Speed Dial}** feature code + (000-049).
3. System Speed Dial entry is entered by **{Speed Dial}** feature code + (2000-2999) for MG-100 and (2000-3999) for MG-300.

Programming

CO Line Data

1. Normal /DISA CO Attributes (PGM 168) - CO Access from DISA
2. CO COS Assignment (PGM 177)
3. CO-to-CO Attributes (PGM 179)

Related Features

Hardware

3.74.9.3 DELETE SENT MESSAGE

Description

This allows a mailbox owner to delete a sent message before the destination party has listened to it.

Operation

Step 1

1. From the main menu press [5].
2. The prompt will state “to edit distribution lists press 1, to delete sent message press 2,

to access your VM name press 3”.

Step 2

1. Press 2.
2. The prompt will state “enter the mailbox number for the message to delete”.
3. Enter the mailbox number.
4. The prompt will state “mailbox XXX message will be deleted, press # to confirm, * to cancel”.

Conditions

1. If there are no messages associated with the mailbox entered, the system will prompt, “there are no messages to delete”.
2. If the mailbox entered is occupied by the owner for retrieval, the system will prompt, “there are no messages to delete”.
3. Messages to be delivered in the future are not deleted by this feature.
4. This feature is applied only to internal VMIB messages. In other words, external VM messages or downloaded voice mail messages are not controlled by this feature.

Programming

Related Features

Hardware

3.74.9.4 LCD VM ENHANCEMETNS

Description

The VM commands in each menu will be displayed on the LCD.

Operation

Conditions

1. VM LCD is displayed only when PGM223-Index 20 Enhanced VM Features is ON.
2. Number display of Voice Messages is changed to VMS(XX/YY/ZZ) from VMS(DD).

   XX : New messages, YY : Saved Messages, ZZ : Urgent Messages

Programming

Related Features

Hardware

3.74.9.5 VM NAME

Description

The VM will be modified so that recording a name for the mailbox is an integral part of the mailbox, not a separate code (542).

Operation

A new option will be created to record the name in a mailbox.

Step 1

1. From the main menu press [5].
2. The prompt will state "to edit distribution lists press 1, to delete sent message press 2, to access your VM name press 3".

Step 2

1. Press 3.
2. The prompt will state "to listen, press 1, to record, press 2, to delete, press 3".

Conditions

Programming

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 # once, then the Daily Alarm will be set and if User dials # once more, then the One-Time Alarm will be set. With the input of @ digit, the alarm mode is toggled.

Programming

Station Data	1	Wake-Up Time (PGM 134 – Index8)
	2	Repeat Wake-up (PGM 134 – Index9)
Table Data	1	Wake Up Answer Tone (Web Admin. PGM 290 – Index65)
	2	Wake-Up Indication Ring (Web Admin. PGM 265 – Index10)
Tenant Data	1	Wake-Up Retry Count (PGM 280 – Index 5)
	2	Wake-Up Retry Time (PGM 280 – Index 6)

Related Features

Hardware

3.76 PSU FAN Alarm

Description

You can configure a fault alarm for PSU FAN device. When a fault is detected, the alarm ring will be given to system attendant.

Operation

Conditions

1. If alarm is active during system attendant is busy, muted ring will be served to the station, and when the station goes to idle, the alarm signal will be given to the station again.
2. When the alarm is ringing, the alarm signal must be reset so that phone operation will be fully functional (fixed or flex buttons do not operate and the user cannot hear the dial tone while alarm is ringing).

Programming

Station Data	1	PSU FAN Alarm Setting (PGM 223 – Index 17)

Related Features

Hardware

- Digital Phone
- PSU FAN, refer to ***iPECS-MG Hardware Description & Installation Manual***.

3.77 Web Phone

Description

The user can make a call without any software in the iPECS-MG web page. Just MS Explore is required. Basically, Web phone has only voice call function and operation is same as softphone.

Operation

Web

1. Click **[Web phone]** Menu in first web page.
2. Type the station login ID/Password and click the **[Login]** button.
3. Install AciveX Control (Click Install AciveX Control).
4. Accept iPECS Web phone as the verified publisher.
5. If installation is finished, web phone will be working.

Condition

1. A lock key is required to use the Web Phone.
2. Currently, the Web phone is designed for the voice call.
3. Only MS Internet explorer is available.
4. If the TLS option is enable in system, user should install the certification file before web access. And TLS applies only when a web phone login.

Programming

Station	1. Logical Slot Assignment (PGM 103)
	2. IP Phone/Phontage Registration Table(PGM 106)

Related Features

Hardware

3.78 Emergency Supervisor

Description

Emergency supervisor can access busy station regardless of privacy authority (Auto Privacy, Voice Over Rejection).

- Voice Over
- Override
- Override & Disconnect

Operation

Same as **{Voice Over}**, **{Override}**, **{Override & Disconnect}**.

Conditions

1. This is not applied to the station setting data line security.

Programming

Station Port Data	1. Emergency supervisor (PGM 123 – Index9)
Station Number Data	1. Voice Over Access (PGM 133-Index9) 2. Rejection of Voice Over(PGM 133-Index10) 3. Auto Privacy(PGM 134-Index11)
System Data	1. Intercom Busy One-digit Attributes (PGM 237)
Numbering Plan	1. Feature Numbering, Override & Disconnect (PGM 113)

Related Features

Hardware

3.79 Call Duration Restriction

The System can be programmed to limit the length of calls at specified stations.

Administrator can make restriction rule in each ICM Call / Incoming Call / Normal Outgoing Call / Outgoing Call in Prefix Table (Local, Long, International Call) / Dedicated CO Line / Mobile Call.

In each case, only single alarm tone can be set as restriction rule after restrict timer. And repeated alarm tone can be set as restriction rule. And also after restriction time, forced release rule can be set, automatically.

If only single alarm tone is assigned as restriction rule, specific station can hear single alarm tone after restriction time.

If repeated alarm tone is assigned as restriction rule, specific station can hear alarm tone periodically in programmed cycle after restriction time.

If forced disconnection rule is assigned as restriction, specific station can hear warning tone and then after timer, call will be released forcibly.

Max 30 rules can be assigned in iPECS-MG 100/300 system. Each station has to refer to one of these rules. And each station will follow one of assigned restriction rule.

3.79.1 ICM Call Duration Restriction

Description

ICM Call Restriction rule can be defined from Call Restriction admin PGM 284, 285.

If call restriction is set, after restriction time, assigned restriction rule will be operated automatically.

Single or repeated alarm tone can be served, or call can be released forcibly.

If only single or repeated alarm tone is set, after restriction time, alarm tone will be served automatically.

If disconnection rule is set, before defined warning tone time, released warning tone will be served to user.

Operation

System

Operation of this feature is automatic when assigned:

Conditions

1. ICM Call Duration Time Display to station’s LCD option is added for ICM call conversation in PGM 123.
2. Call Duration Restriction Admin is moved from DN base admin PGM134 to Station Base admin PGM 121.

Programming

Station 1. Call Duration Restriction Table (PGM 121-

Index14)
2. ICM Call Duration Time Display (PGM 123-Index13)

Table Data 1. Call Duration Restriction Table (PGM 284-285)

Related Features

Hardware

3.79.2 Incoming Call Duration Restriction

Description

Incoming Call Restriction rule can be defined from Call Restriction admin PGM 284, 285.

If call restriction is set, after restriction time, assigned restriction rule will be operated automatically.

Single or repeated alarm tone can be served, or call can be released forcibly.

If only single or repeated alarm tone is set, after restriction time, alarm tone will be served automatically.

If disconnection rule is set, before defined warning tone time, released warning tone will be served to user.

Operation

Operation of this feature is automatic when assigned:

Conditions

1. Call Duration Restriction

Programming

Station 1. Call Duration Restriction Table (PGM 121-Index14)

Table Data 1. Call Duration Restriction Table (PGM 284-285)

Related Features

Hardware

3.79.3 Normal Outgoing Call Duration Restriction

Description

Normal Outgoing Call Restriction rule can be defined from Call Restriction admin PGM 284, 285.

Normal Outgoing Call means unmatched all of outgoing call from Local / Long / International Prefix table.

If call restriction is set, after restriction time, assigned restriction rule will be operated automatically.

Single or repeated alarm tone can be served, or call can be released forcibly.

If only single or repeated alarm tone is set, after restriction time, alarm tone will be served automatically.

If disconnection rule is set, before defined warning tone time, released warning tone will be served to user.

Operation

Operation of this feature is automatic when assigned:

Conditions

1. Call Duration Restriction

Programming

Station	1. Call Duration Restriction Table (PGM 121-Index14)
Table Data	1. Call Duration Restriction Table (PGM 284-285)

Related Features

Hardware

3.79.4 Local / Long / International Outgoing Call Duration Restriction

Description

Outgoing Call can be divided Local / Log / International Call with comparison between dialed digit from user and Local / Long / International Prefix table.

Each Local / Long / International Call can be set restriction rule from Call Restriction admin PGM 284, 285. If call restriction is set, after restriction time, assigned restriction rule will be operated automatically. Single or repeated alarm tone can be served, or call can be released forcibly.

If only single or repeated alarm tone is set, after restriction time, alarm tone will be served automatically.

If disconnection rule is set, before defined warning tone time, released warning tone will be served to user.

Operation

Operation of this feature is automatic when assigned:

Conditions

1. Call Duration Restriction.

Programming

Station	1.	Call Duration Restriction Table (PGM 121-Index14)
Table Data	1.	Call Duration Restriction Table (PGM 284-285)
Tenant Data	1.	Local Call Prefix Table (PGM 286)
	2.	Long Call Prefix Table (PGM 287)
	3.	International Call Prefix Table (PGM 288)

Related Features

Hardware

3.79.5 Dedicated Line Call Duration Restrict

Description

CO line can be defined as Normal CO line or Dedicated Line in CO Access Mode admin in PGM162-Index1.

If CO line is dedicated line, all of outgoing call will be handled as dedicated line. And Outgoing Call through dedicated line can be set restriction rule from Call Restriction admin PGM 284, 285. If call restriction is set, after restriction time, assigned restriction rule will be operated automatically. Single or repeated alarm tone can be served, or call can be released forcibly.

If only single or repeated alarm tone is set, after restriction time, alarm tone will be served automatically.

If disconnection rule is set, before defined warning tone time, released warning tone will be served to user.

Operation

Operation of this feature is automatic when assigned:

Conditions

Call Duration Restriction.

Programming

Station	1. Call Duration Restriction Table (PGM 121-Index14)
CO	1. CO Access Mode (PGM 162-Index1)
Table Data	1. Call Duration Restriction Table (PGM 284-285)

Related Features

Hardware

3.79.6 Mobile Call Duration Restrict

Description

In case of Mobile Call, restriction rule can be defined from Call Restriction admin PGM 284, 285 in Tenant base admin. Administrator can define mobile call prefix digits in PGM289 Mobile Prefix Table.

If user seizes the CO line and dial defined mobile number in Mobile Prefix Table. Mobile Call Restriction rule will be applied. If call restriction is set, after restriction time, assigned restriction rule will be operated automatically. Single or repeated alarm tone can be served, or call can be released forcibly.

If only single or repeated alarm tone is set, after restriction time, alarm tone will be served automatically.

If disconnection rule is set, before defined warning tone time, released warning tone will be served to user.

Operation

Operation of this feature is automatic when assigned:

Conditions

1. Call Duration Restriction

Programming

Station	1. Call Duration Restriction Table (PGM 121-Index14)
Table Data	1. Call Duration Restriction Table (PGM 284-285)
Tenant Data	1. Mobile Call Prefix Table (PGM 289)

Related Features

Hardware

3.80 Tenant Group Access

Description

Stations in a group are allowed or denied the ability to place intercom calls to Stations and CO calls in other groups on a Group-by-group basis.

There are four tables for Tenant group access.

1 CO tenant to CO tenant
2 CO tenant to Other types tenant
3 Other types tenant to Other types tenant
4 Other types tenant to CO tenant

Operation

Allow/Deny access to other groups:

1 Press the **[PGM]** button and dial 283.
2 Dial the desired tenant number(1-5 for the iPECS-MG 100, 1~9 for the iPECS-MG 300)
3 Select access tenant type (1: CO, 2: Others) of selected tenant.
4 Select the tenant type (1: CO, 2: Others) to be accessed.
5 Press flex button to access or deny (toggle) the tenant.

Conditions

1 The Tenant Group Access is one way access. To allow access two-way access between group 1 and group 2, it should be allowed 1→2 and 2→1.

Programming

Tenant Data	1. Tenant Group Access (PGM 283)
Station Data	1. Station Tenant No. (PGM 131 – Index2)
CO Line Data	1. CO Tenant No. (PGM 160 – Index5)

Related Features

Hardware

3.81 Tone Service For DECT Switched-Off Case

Description

When DECT terminal is switched off, system does not know the situation until it tries to call the terminal. So, if there's an incoming call to a switched-off DECT station, the system makes a call first and then waits for the response for about ten seconds. Meanwhile, the caller will hear a new tone called “Wireless Station Searching Tone”. If the DECT station is found, the caller will hear ring back tone immediately. Otherwise, the call can be rerouted to another destination according to the setting. If the caller is an internal station, the caller will hear “Internal No Answer Tone” and the call will be released. If the call is an incoming CO call, the call will be rerouted to the “Error Destination” of the CO line specified in Incoming CO Alternative Destination (PGM 169).

Operation

When DECT terminal is switched on:

1 The DECT terminal will receive ring immediately and the caller will hear ring-back tone.

When DECT terminal is switched off:

1 The caller will hear “Wireless Station Searching Tone” first and the system will search for the called DECT station.

2 In about ten seconds, the search will fail and the call will be routed to “Error Destination” in case it is a CO call. If the caller is an internal station, “Internal No Answer Tone” will be heard and the call will be disconnected.

When DECT terminal is turned on in the middle of phone searching:

1 The caller will hear “Wireless Station Searching Tone” at first. But ring back tone will be heard immediately after the called station is found.

Conditions

1 This feature is not available when an internal SIP terminal is calling a DECT station.

2 Internal call will not be rerouted even if DECT station is not found.

3 If the CO ring is assigned only to one DECT station, the CO call will be rerouted in case that DECT terminal is switched off. But if there're multiple ring-assigned stations, the call will not be rerouted.

4 If the DECT station is the only member of a MADN, the caller will hear “Wireless Station Searching Tone” first. And if the call is from a CO line, it will be rerouted. But if the DECT station is not the only member of a MADN, the caller will hear ring-back tone and CO call rerouting will not be supported.

Programming

Tenant Data	1. Wireless Station Searching Tone (PGM 290–Index78)
CO Line Data	1. Incoming CO Alternate Destination (PGM 169)

Related Features

Hardware

3.82 IPCR

Description

System can record a voice automatically or manually using IPCR server. IPCR (IP call recording) Server can be registered as an SIP extension to an iPECS-MG system. A station's number which is same with an agent ID number of an IPCR server is automatically or on-demand recorded about internal call, and external call.

Operation

System

IPCR Equipment Registration:

Set IPCR Server IP address and SIP extension number for IPCR SIP User ID in System Attribute (PGM223).

Configure [SIP Station Basic Registration Table] in Web Admin for SIP extension number for IPCR SIP User ID.

IPCR Server

Before registration, you should install the IPCR server in PC based on Linux (OS: Fedora 12) using install CD or downloading from our BCS web site.

Set IPCR before registration to system.

PBX registration (system IP, SIP ID, SIP Password)

IPCR Server registration

User registration

Channel registration (Agent ID Registration)

Agent ID Registration

Match the Agent ID number to the station number which has to be programmed to record voice of its call.

You can use ACR (Auto-call recording) by configuring Auto-Record Service (PGM145-3).

Conditions

1. This feature requires a license key for IPCR server and IPCR agents based on the number of agent to be recorded.
2. You can search the recorded voices using Web Admin of IPCR.
3. If recording type on a station is ODR (On Demand Recording, i.e. IPCR Auto Record is OFF), the station user has to press the [Two Way Record] button to start recording.
4. If a user press [HOLD]/[TRANS] button while recording, the recording will be stopped.

Programming

Station VMIB Attribute PGM145-3 Auto-Record Service

Station VMIB Attribute PGM145-4 Two-Way Record Access

Station VMIB Attribute PGM145-5 Two-Way Recording Destination

System Attribute PGM 223-21- IPCR Server IP address

System Attribute PGM 223-22- SIP Extension Number for IPCR

SIP Station Basic Registration Table (380)

SIP Station Additional Registration Attributes (381) - SIP Phone Type

Related Features

Hardware

IPCR Server

3.83 Mobile Extension Callback

Description

To use mobile extension, 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.

According to the option,

1) The system can provides a system dial tone to the mobile user and the user can dial **[Mobile Extension Callback]** code and disconnect the call. Then, the system will make outgoing call to the registered number in PGM 146.

2) The system disconnects the call and makes outgoing call to the registered number in PGM 146 automatically.

The mobile user is able to save phone bill.

Operation

By Feature Code

1. Set PGM 146 Mobile Enable to ON.
2. A mobile user calls to his extension of iPECS-MG system.
3. If CLI is matched, the dial tone is heard.
4. Dial **[Mobile Extension Callback]** feature code
5. Confirm tone is heard.
6. Disconnect the call by the system or the mobile user.
7. The system makes new outgoing call to the mobile user automatically. After answering the call, the mobile user can use the features for mobile extension.

Automatic Callback

1. Set PGM 146 Mobile Enable to CALLBACK.
2. A mobile user calls to his extension of iPECS-MG system.
3. If CLI is matched, the system disconnects the call.
4. The system makes new outgoing call to the mobile user automatically. After answering the call, the mobile user can use the features for mobile extension.

Conditions

1. The Mobile Extension Callback feature is supported via system digital lines only.
2. When the mobile phone places an external call through the system, the CLI of the corresponding extension is used.
3. When the system makes a callback call to the mobile user, the CLI of the corresponding station is used.

Programming

Numbering Plan	1	Feature Numbering Plan(PGM 113)
System Data	1	Attributes (PGM 236)
Station Data	1	Mobile Extension Access (PGM 132 – Index6)

2 Mobile Extension Number Attributes (PGM 146)

Related Features

Hardware

3.84 Station/Attendant Group Forward Program

Description

A member of station /attendant group can set forward for station group/attendant group

Operation

Digital Phone and LIP Phone (LDP/LIP Phone).

1. Dial {Group Forward Register} feature code.
2. Dial station group number or attendant call code.
3. Enter Forward Type (1-4).
4. Enter the destination.
5. Press [Save] button.

SLT

1. Dial {Group Forward Register} feature code.
2. Dial station group number or attendant call code.
3. Enter Forward Type (1-4).
4. Enter the destination.
5. Press [Hook Flash] button.

Programming

Station Group Data	1. Forward destination (PGM 202-5)
Tenant Data	1 Forward destination (PGM 272-5)

3.85 WAKE-UP FAILURE RING TO ATD

Description

When a wake-up ring fails; that is, a wake-up ring is not answered until the retry counter, and it is notified to the tenant attendant by wake-up failure ring.

Operation

1. A station's wake-up ring fails.
2. It is notified to the tenant attendant through wake-up failure ring.
3. When the wake-up failure ring is answered, the tenant attendant will hear the wake-up time.

Conditions

1. When an attendant is busy, wake-up failure rings are queued.
2. Queued wake-up failure rings are handled when busy attendants goes idle.
3. When wake-up failure ring is not answered until the ring time (PGM265-Index 23), it is cleared.

Programming

Table Data 1. Ring Table (PGM265)-Index 23 Wake-up Failure Ring

Related Features

Hardware

3.86 Authorization usage

Description

Each DN may enter authorization code according to option when he makes a call.

- Disable.
- CO Access: Only when a user accesses CO line, system requests authorization code (station number + password, or * + ID + Password).
- Authorization Table – User dials digits in authorization table, system requests authorization code (station number + password, or * + ID + Password).
- CO Access, Authorization Table - When a user accesses CO line or user dials digits in.
- authorization table, system requests authorization code.
  (station number + password, or * + ID + Password)

Operation

Conditions

1. In case of System Speed Dial, authorization usage feature can be enabled according to Authorization for system speed option.

Programming

Station Number Data	1. Station DN attribute – Authorization usage (PGM132-9)
Tenant Data	1. Tenant attribute – Authorization for system speed (PGM280-12)

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.

[DN] button:

- In use at Station
- In use by another Station
- DND
- Receive Incoming Call
- Hold
- Call Forward
- Conference at Station
- Conference by another Station
- Conference Supervisor

[DSS] button:

- Incoming Call
- Busy
- DND
- Call Forward
- Handset Lifted
- Pre-selected Message
- Hold

Operation

Digital Phone

*To assign a **{DSS}** button to a Flex button on a Digital Phone or a DSS Console:*

1. 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

<u>*To place an Intercom call:*</u>

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, await answer. Or, for Intercom handsfree-tone, speak and await answer.

SLT

<u>*To place an intercom call:*</u>

1. Lift the handset to receive ICM dial tone.
2. Dial Station number.
3. For Ring-Back tone, await answer. Or, for Intercom splash-tone, speak and wait 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, 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 Plan	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)
Tenant Data	1	Tone Table (PGM 290)

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 System Hold and will Recalls 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 **[DSS]** button LED associated with the Held station will flash at the designated Hold rate.

To retrieve the held ICM call:

1. Press the **{DSS}** button associated with the Held station or **[HOLD]** button, 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
2. Dial the Held station number and 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 Plan	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

<u>*To change the ICM Signaling mode:*</u>

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.
5. This Feature is available only for digital phone.

Programming

Station Data	1	Forced Hands free Access (PGM 132 – Index1)
Numbering Plan	1	Forced Hands free Call Code (PGM 113 – Index 59)

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 Dial 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. 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.
2. 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 and dial the Station to receive the call. Or, 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 **[TRANS]** button and dial the Station to receive the call. Or, 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. If the Receiving Station is busy, the Transferring Station may Camp-On the call at the Busy Station (refer to Camp-On).
2. 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

<u>*To perform an Intrusion:*</u>

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}** soft button.
4. To delete, Press **{DELETE}** soft button.

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 Keyset 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/CALL BK] 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/CALL BK] button LED will flash; when MWI is received at a SLT, the MW lamp will flash.

Operation

Digital Phone

<u>*To leave a Message Wait:*</u>

1. While receiving a Ring Back tone or No answer on a Call Announce (H or P mode).
2. Press the **[MSG/CALL BK]** button.
3. On Hang-up, the MWI is activated.
4. The **[MSG/CALLBK]** button of the receiver will flash as programmed.

<u>*To leave a Message Wait:*</u>

1. While receiving a DND tone, press the **[MSG/CALL BK]** button.
2. On Hang-Up, the MWI is activated.
3. The **[MSG/CALLBK]** button of the receiver will flash as programmed.

<u>*To leave a Call Back (queue for a station):*</u>

1. While receiving a Busy Signal on an attempted call, press the [MSG/CALL BK] button; the confirmation tone will be heard.
2. Hang-up, and return to Idle.

<u>*To respond to a Call Back Recall Request:*</u>

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/CALL BK]** 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),
2. Press the Hook-Switch.
3. Dial the **{Message Wait Register feature code}**.
4. 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 Register feature code }**.
3. Hang-Up, the MWI is activated.

To retrieve a Station Message Wait:

1. Dial the **{Message Wait Answer feature code}**.

To leave a Call Back (queue for a station):

1. While receiving a Busy Signal, press the hook switch.
2. Dial the **{Call Back Register feature 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.
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/CALL BK]** 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 250 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)
Station Port Data	1	Message-Wait Indication (PGM 124-Index1)

Related Features

Hardware

4.9.3 Message Wait Reminder Tone

Description

In addition to the **[MSG/CALL BK]** 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

<u>When programmed, the MWI Reminder Tone is automatically sent to Stations in Idle as appropriate.</u>

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.

In addition, one external page zone is supported.

The default Page Zone Dial codes are:

Table 4.10.1-1 Paging Dial Codes

Page Type	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	548	548
All Call Page	549	549

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:*

1. 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 CODE}** button.
3. Dial page group number.
4. If assigned, after the Page Ring, make the desired announcement.
5. When finished, replace the handset and go On-Hook.

SLT

<u>*To initiate a Page:*</u>

1. Lift the handset.
2. Dial the desired Paging code.
3. Dial page group number.
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. Due to the restriction of paging feature, it is not possible to use paging when the station is temporarily out of the conference or conference room. If the station tries to use paging in this case, the error tone will be heard.
5. If a station pages to the page zone 00, all the stations in each page zone will receive the paging. In this case, paging feature code is not saved as a call log in PC Phone (UCS client or Phontage).
6. Stations, that are not included in any Internal Page Zone, will not receive any page, including All Call.
7. For External Paging, an External Amplifier and Speaker(s) are required.
8. 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 **{Page Auto Answer Feature Code}** or **{Internal Page Answer Feature Code}**. In addition, **{Meet Me}** button can be assigned for paging answer in digital phones.

Table 4.10.2-1 Page Answer Dial Codes

Page Type	iPECS-MG 100	iPECS-MG 300
Page Auto Answer	546	546
Internal Page Answer	547	547

Two kinds of Page Answer codes are supported:

- {Page Auto 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
- {Internal 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 **{PAGE AUTO ANSWER CODE}** OR **{INTERNAL PAGE ANSWER CODE}** button:*

1. Press the **[PGM]** + **{FLEX}** + Button Feature Type (1) + **{PAGE AUTO ANSWER CODE}** or **{INTERNAL PAGE ANSWER CODE}** + **[SAVE]**.

To answer a Page by the Paging Party:

1. Lift the handset and press the **[HOLD]** button. Or, press the **[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, and dial the Group number. Or, 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. Users may answer a Meet Me Page from any Station regardless of Pick-Up/Paging Group assignments and Page access permission. But the station should have the Meet-Me access permission.
3. The Paging Party must remain Off-Hook until the Paged Party answers the Meet Me request.
4. The Initiator/Paging Party may press the Mute button to eliminate transmitting over the Page circuit while waiting for the Paged Party to answer.
5. A SLT user can answer a paging by using a Meet Me Page feature code in dialing state.
6. If the user doesn't know the zone number of the current paging, zone number 00 can be dialed so that zones from 01 to the end can be searched for paging answer.

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 page to a preferred zone by using the **{Personal VM Page}** feature code. Before using the VM Paging, users have to record a voice message for paging by using **{VM Page Message Record}** feature code. Then, the recorded voice message can be paged to a zone. In case of Attendants, they can also use system announcement for paging with **{Announcement Page for Attendant}** feature code. This feature allows the user to record a voice message as a provision against an emergency situation and page to a zone with the recorded message when an emergency occurs.

Table 4.10.3-2 VM Paging Dial Codes

Page Type	iPECS-MG 100	iPECS-MG 300
Personal VM Page	544 + Group Number	544 + Group Number
Announcement Page for Attendant	545	545

Operation

Digital Phone

To record a VM Message:

1. Lift the handset or press the **[MON]** button.
2. Dial the **{VM PAGE MESSAGE RECORD}** feature code.
3. Record VM Message.

To delete a recorded VM Message:

1. Lift the handset or press the **[MON]** button.
2. Dial the **{VM PAGE MESSAGE RECORD}** feature code.
3. Dial **[SPEED]** Button for deletion.

To make a VM Page:

1. Lift the handset or press the **[MON]** button.
2. Dial **{PERSONAL VM PAGE}** feature code or press a **{PERSONAL VM PAGE}** button.
3. Dial page group number.
4. If assigned, after the Page Ring, VM message will be played.
5. Replace the handset, go on-hook.

To make a Announcement Page by Attendant:

1. Lift the handset or press the **[MON]** button.
2. Dial **{ANNOUNCEMENT PAGE FOR ATTENDANT}** feature code or press a **{ANNOUNCEMENT PAGE FOR ATTENDANT}** button.
3. Dial page group number.
4. Dial desired Announcement number.
5. If assigned, after the Page Ring, VM message will be played.
6. Replace the handset, go on-hook.

SLT

To make a page:

1. Lift the handset.
2. Dial **{PERSONAL VM PAGE}** feature code.
3. Dial page group number.
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 in which 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.

Operation

Digital Phone

*To assign a **{PTT}** Flex button:*

1. Press **[PGM]** + **{FLEX}** + Button Feature Type (2) + **{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 the **{PTT}** Flex button.
2. Make the desired Page announcement after hearing confirmation tone.
3. When paging is finished, press the **{PTT}** Flex button once more or replace the handset.

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 feature code (PGM 113 – Index 46)
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 paths 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.
5. If ARS is operated with Digit Conversion Table, Dialed or Converted Digit will be provided to CO line after ARS service.

Programming

Numbering Plan	1	CO Group Access Code (PGM 114)
CO Line Data	1	CO Group Access Code Attributes (PGM 180) − ARS Service (Index 6) − ARS Digit (Index 7, 9) − ARS Digit Usage (Index 8, 10) − AND Digit (Index4)

Related Features

Hardware

5.2 Automatic Network Dialing

Description

AND is a feature in which AND digits are dialed automatically by the system after a CO Line is seized by means of CO Group Access code. When a Station User dials a CO Group Access Code and a CO Line is seized, Automatic Network Dialing (AND) digits are dialed automatically by the system. 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 / Last / First).
- **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 Group 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

<u>While connected to an Analog CO line:</u>

1. Press the **[CO FLASH]**.button, the System automatically will generate a Flash on the CO Line.

SLT

<u>While connected to an Analog CO line:</u>

1. Press the Hook-switch.
2. Dial **{CO Flash Feature Code}**.

Conditions

1. During Flash, the LED for the CO Line button will remain lit.
2. A Flash may be stored as a part of a Station or System Speed Dial number.
3. While connected to an Internal Call or Dial Tone, pressing the **[CO FLASH]** button will re-access the ICM Dial tone.
4. 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)** - When an incoming CO call from ISDN CO line makes an outgoing call through non-ISDN line, ISDN progress indicator Information element can be sent to PX.
- **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 ingoing 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:

1. **Screen Indicator** - Determines if Screen Indicator information is included when it consists of ISDN CLI information.
2. **Sending Caller Number** - Determines if Calling Party Number information is included when it consists of ISDN CLI information.
3. **Calling Type** - Defines the "Type of Number Plan" provided in Calling Party Information Element of the ISDN call SETUP message.
4. **Calling Party Numbering Plan** - Determines Calling Party Numbering Plan information if it consists of ISDN CLI information on Outgoing Transit Call.
5. **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.
6. **Sending Complete IE for Information Message** - Determines if Digit Sending Complete IE message is included on an ISDN information message.
7. **CLI Information when Transit** - Determines if CLI Information is included on Outgoing Transit Calls when no information is provided on Incoming Transit Calls.
8. **CLI Information to Transmit Toward PX on Outgoing Transit Calls** - Creates new CLI by adding or deleting the received CLI Information.
9. **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.
10. **CLI Conversion Table Index** - Defines the table index in which CLI conversion rule is programmed for each outgoing 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, Ring, 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, Ring, 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.
- **Out Of Service**– If an Incoming CO Call is received at a Station (Which is Out of Service), the call can be set for routing destination, Disconnect, ATD, CO Ring Assign, Alt Ring table, Tone, or Pilot Hunt.
- **Error**– If an Incoming Call is placed to error case, 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 Line.
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.
4. If select prompt usage, the prompts will be played and follows routing destination.

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, Pilot Hunt, or Ring.
- **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, Ring, 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, Pilot Hunt, or Ring.

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 Enblock or Overlap.

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 external caller 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

5.6.7 DID Name Service

Description

When there's DID call, if name is programmed in Digit Conversion Table and dialed digit is matched, the name is displayed on ringing Station's LCD.

Operation

If programmed, did name is displayed to DID destination station's LCD.

Conditions

Programming

Table Data	1	DID Name (PGM 251)

Related Features

Hardware

5.6.8 Incoming CO Line Holiday Service

Description

When Ring Mode is Holiday mode, incoming CO call can be routed to Alternative Ring destination by a programmed holiday ring table index.

The destination of alternative ring table can be station or feature code.

Operation

If set, CO Line Holiday Service operation is automatic.

Conditions

Programming

CO Line Data	1	Incoming CO Line Attributes (PGM 165–166)
	2	Alternative CO Ring Table (PGM 181)

Related Features

Hardware

5.6.9 DID/DISA Restriction

Description

DID or DISA calls to some stations can be restricted according to Admin Program.

If some station number are set to DID/DISA restriction, the information us displayed on the station's LCD.

Operation

If programmed, DID/DISA Restriction status is displayed to station's LCD.

To change DID/DISA restriction status from each station terminal(toggle):

1. Dial the **{DID/DISA Restriction Feature Code}**.

Conditions

1. When the external caller dials the station's DID or DISA number, the call is restricted.
2. Other calls (ring assigned CO calls or forwarded or transferred calls, etc) are not restricted.
3. This feature is not applied to SIP phones.

Programming

Station Data	1	DID/DISA Restriction (PGM 134 – Index 11-12)
Numbering Plan	1	DID/DISA Restriction (PGM 113 – Index108)

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

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.

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

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 – Index13)
	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

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 Assignment (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

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

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, it is possible to 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 COS Assignment (PGM 177)
	2	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

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 not a member of Pilot 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 Assignment (PGM 130)
CO Line Data	1	CO Line Attribute (PGM 160)
	2	Incoming CO Alternative (PGM 169)
Numbering Plan	1	Feature Numbering Plan (PGM 113)
Table Data	1	Digit Conversion Table (PGM 251/252)

Related Features

Hardware

- 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. If password input prompt is heard, dial the required password.

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. 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.

The duration of DTMF signals can be adjusted.

It is also possible to change the signal OFF time between DTMF digits when DTMF signals are sent to PX.

Operation

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

1 H.323 Route Attribute (PGM 360)

2 H.323 Call Attributes (PGM 361)

3 H.323 Incoming Attributes (PGM 362)

Related Features

Hardware

5.18 Incoming Calling Line Id (ICLID) Call Routing

5.18.1 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 Data	1	ICLID Table (PGM 262)

Related Features

Hardware

5.18.2 Incoming Calling Line ID (ICLID) Call Routing Exception

Description

This feature provides a way to route a call through DID CO line to an alternative destination even if ICLID is matched. Originally, if the CLI is matched in ICLID table, there is no way to perform a different call routing. But, with this feature, the call can be routed after analyzing both the CLI and the dialed digits from the network.

Operation

When an entry in ICLID table has no exception table index (ICLID service is enabled for CO)

1. If there's an incoming call and CLI is matched in ICLID table, the call will be routed immediately to the programmed destination.
2. Otherwise, the call will be routed as programmed.

When an entry in ICLID table has an exception table index (ICLID service is enabled for CO)

1. If there's an incoming call and CLI is matched in ICLID table, the call is not routed since there is an exception table index.
2. System waits for the digit stream coming from the network.
3. If the digit stream is matched in the programmed ICLID exception table, ICLID routing is canceled and normal DID procedure will be applied (digit conversion process).
4. If the match failed in the relevant ICLID exception table, ICLID routing is performed immediately.

Conditions

1. This feature is available only for DID CO lines.
2. Each entry in ICLID table can have an exception table index
3. By default, the exception table index is empty.
4. There are 5 different ICLID exception tables in system. And each exception table has maximum 50 entries.

Programming

CO Line Data	1	ICLID Service (PGM 165 – Index7)
Table Data	1	ICLID Exception Index (PGM 262 – Index 8)
	2	ICLID Exception Table (PGM 267)

Related Features

Hardware

5.19 IP Trunking

5.19.1 H323 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

Operation of H.323 Service is automatic.

Conditions

Programming

H.323	1	H.323 Call Attribute (PGM 361)

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 supports 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 CO Basic Registration (Web Admin. PGM 370)
	2	SIP CO Additional Registration Attributes (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 sent through inband, RFC2833 or outband (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. When system gets call from the assigned IP address in PGM 362; the system routes the call to one of 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 Attribute (PGM 361)
	3	H.323 Incoming Attributes (PGM 362)

Related Features

Hardware

5.21 Integrated Service Digital Network (ISDN)

Description

According the board type, PRIB supports 23B+1D(T1/PRI) channels and 30B+2D(E1/PRI) channels.

T1/PRI supports North America standards and PRI supports ETSI specifications.

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)
	2	Metering Usage (PGM 160 - Index12)

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 identification, which is also included in call records and the Digital/IP Phone can display it on the LCD. The received identification may be sent to a selected serial port.

Digital/IP 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

Station Number Data	1	CLIR When Outgoing (PGM 135 – Index4)
	2	COLR When Incoming Answer (PGM 135 – Index5)

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

<u>*To program a **{KEYPAD FACILITY}** button:*</u>

1. Press **[PGM]** + **{FLEX}** + Button Feature Type (1) + **{Keypad Facility}** feature code + **[SAVE]**.

<u>*To activate the keypad facility after seizing an ISDN line:*</u>

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 Port Data	1	Flex Button Assignment (PGM 126)
	2	Keypad Facility usage (PGM 133 – Index12)

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 assign multiple subscriber numbers to one ISDN CO line. Additionally, it enables a station to make an outgoing call using a specific CLI.

Operation

If set, MSN operation is automatic.

Conditions

1. If a CO line uses Representative CLI, that Representative CLI is sent prior to 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.
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 Number Data	1	Station CLI Attributes (PGM 135)
CO Line Data	1	Own Code (Incoming) (PGM 165 – Index9)
	2	Own Code (Outgoing) (PGM 170 – Index13)
	3	Representative CLI Usage (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 Offnet Call Forward Usage (PGM 166 – Index16)

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 – Index11)
	2	Representative CLI (PGM 170 – Index12)

Related Features

Hardware

5.24 Collect Call Blocking For E1-R2 In Brazil

Description

When the E1-R2 receives the incoming collect call, it could reject to accept the call by operating collect call blocking procedure.

There are two kinds of collect call service type. One is 'Without Indicator' and the other is 'With Indication'. For E1-R2, 'Double Answer' is operated with E1 line signaling and 'With Indication' is operated with R2MFC signaling.

Operation

To deactivate collect call blocking feature

1. Set Admin Program '0 (disable)' to deactivate collect call blocking.
2. It does not operate anything.

To activate collect call blocking feature as 'Double Answer'

1. Set Admin Program '1 (Double Answer)' to activate collect call blocking enabled.
2. When a incoming call is received, the call is answered.
3. If it is activated as 'Double Answer', as soon as answering, it sends the collect call blocking signal towards PX.

To activate collect call blocking feature as 'With Indication'

1. Set Admin Program '2 (With Indication)' to activate collect call blocking enabled.
2. When a incoming call is received, R2 signaling is processed.
3. If it is activated as 'With Indication', when a caller's category is 'Collect Call', it sends Unallocated Number signal towards PX.

Conditions

1. When the MG system sends the collect call blocking signal towards PX, if it is a collect call, it disconnects the call. If it is a normal call, it continues to communication.
2. To block collect call as Double Answer, Double Answer signaling is made for 'Collect Call Break Timer' after answering during 'Collect Call Make Timer'

Programming

CO Line Data	1	Collect Call Blocking (PGM 166-Index20)
	2	Collect Call Answer Timer (PGM 166-Btn 21)
	3	Collect Call Idle Timer (PGM 166-Btn 22)

Related Features

Hardware

5.25 BRIB Clock Port Selection

Description

System can use the clock from BRIB as main system clock when select the ports on BRIB to use clock. PGM 300 – F18, F19, F20 and F21 represent each port on BRIB.

Operation

1. Set the PGM 300 – F18, F19, F20 and F21 for each port on BRIB.

Conditions

1. If BRIB isn’t set on PGM 301(no priority), this function is ignored.(i.e. system can’t select the clock from BRIB as master clock)
2. If no BRIB ports are selected for this function on PGM 300, system can’t select the clock from BRIB as master clock even though BRIB is set on PGM 301.
3. In case of BRIB2, F20 and F21 are ignored.

Programming

1 ISDN/Digital Board Attribute – PGM 300 Index 18(port 1), Index 19(port 2), Index 20(port 3), Index 21(port 4)

2 ISDN Clock Priority – PGM 301

Related Features

1. ISDN Clock Priority (PGM 301)

Hardware

5.26 Fail Over PSTN

Description

iPECS-MG supports fail over PSTN rouging. When extension go to out of service, make call to forward destination number on failure. This service can be enabled or disabled by system base.

Operation

To Register Fail over PSTN:

1 Select DN number at PGM 311

2 Set destination number on Failure.

Conditions

Fail over PSTN only used for extension.

In case of CO lines fail over PSTN, can be supported by ARS services.

Programming

1. Outgoing CO Line Attributes (PGM 171 – Index 11)
2. Incoming Attributes (PGM 362 – Index 4)
3. H.323 Check Message Information (PGM 364)

Related Features

Hardware

5.27 Virtual Subscriber Service

Description

This feature allows considering CO incoming call with CLI (Calling Line Identification) as a virtual subscriber. The virtual subscriber is processed with day/night/timed class and tenant. The virtual subscriber can have specific destination. This feature is for transit exchange or intermediate exchange. This service is implemented for any trunk line types which can identify CLI number.

Virtual Subscriber

A subscriber which is not IPECS-MG extension subscriber, but which can be identified by received CLI number and/or Called Number

Virtual Subscriber Table

This table contains incoming CLI, called number, incoming CO group, day/night/timed class, tenant, maximum virtual calls, Virtual CLI table index and destination. The incoming CLI can be assigned up to 24 digits. The table can be assigned up to 300(MG-300)/100(MG-100) tables.

Virtual CLI Table

This table contains numbers used for CLI when a virtual subscriber makes outgoing call. The table can be assigned up to 300(MG-300)/100(MG-100) tables.

Virtual Subscriber Service Option

This option contains whether to apply virtual subscriber service or not and how to apply virtual subscriber service.

Operation

System

The System will implement routing automatically based on database entries and the received CLI.

1. System receives a call from CO.
2. System processes a virtual subscriber with temporary day/night/timed class and tenant if CLI of incoming call fulfills the condition of virtual subscriber, and system sends the call to the specific destination of virtual subscriber.
3. If the destination of virtual subscriber is CO access code, system processes the CO access code and sends the called party number to the outgoing CO.
4. If there is no destination of virtual subscriber, system just sends called party number incoming through CO.

1 Virtual Subscriber Service Option

- **NO**: Not to apply virtual subscriber service to incoming calls. It means normal CO incoming call process.
- **ALLOW**: If received CLI fulfills digits condition of virtual subscriber and real incoming CO group is same with pre-assigned incoming CO group of virtual subscriber, virtual subscriber service would be applied to the incoming call. But if received CLI doesn't fulfill digits condition of virtual subscriber or real incoming CO group is not same with pre-assigned incoming CO group of virtual subscriber, the incoming call would be released.
- **DENY**: If received CLI fulfills digits condition of virtual subscriber and real incoming CO group is same with pre-assigned incoming CO group of virtual subscriber, the incoming call would be released. But if received CLI doesn't fulfill digits condition of virtual subscriber or real incoming CO group is not same with pre-assigned incoming CO group of virtual subscriber, the incoming call would be processed by normal CO incoming call.
- **MATCH**: If received CLI fulfills digits condition of virtual subscriber and real incoming CO group is same with pre-assigned incoming CO group of virtual subscriber, virtual subscriber service would be applied to the incoming call. But if received CLI doesn't fulfill digits condition of virtual subscriber or real incoming CO group is not same with pre-assigned incoming CO group of virtual subscriber, the incoming call would be processed by normal CO incoming call.

2 Virtual Subscriber's CLI

Incoming CLI through CO should fulfill one of condition below and real incoming CO group should be same with pre-assigned incoming CO group of virtual subscriber.

- **Whole Numbers**: Whole received CLI should be same with pre-assigned digits of virtual subscriber. It has the highest priority.
  Ex) The case that assigned digits of virtual subscriber are '4504875' and received CLI is '4504875'.
- **Prefix Masked Numbers**: Length of CLI should be same with pre-assigned digits of virtual subscriber and fixed length end digits of CLI should be same with one of pre-assigned digits of virtual subscriber.
  Ex) The case that pre-assigned digits of virtual subscriber are 'XXX4875' and received CLI is '4504875'. 'X' means any one digit. Length of incoming CLI is 7. The last 4 digits of CLI are same with one of virtual subscriber and any beginning 3 digits are available.
- **Postfix Masked Numbers**: Length of CLI should be same with pre-assigned digits of virtual subscriber and fixed length beginning digits of CLI should be same with one of pre-assigned digits of virtual subscriber.
  Ex) The case that pre-assigned digits of virtual subscriber are '450XXXX' and received CLI is '4504875'. Length of incoming CLI is 7. The first 3 digits of CLI are same with one of virtual subscriber and any last 4 digits are available.
- **Length Matching**: Length of CLI should be same with one of virtual subscriber and any digits are available.
  Ex) The case that pre-assigned digits of virtual subscriber are 'XXXXXXX' and received CLI is '4504875'. Length of incoming CLI is 7.
- **Beginning Masked Numbers**: The last part of digits of CLI should be same with one of virtual subscriber and length of CLI should be same with or longer than virtual subscriber.
  Ex) The case that pre-assigned digits of virtual subscriber are 'C4875' and received CLI is '4504875'. 'C' means any one and more digits. Length of incoming CLI is more than 4. The last 4 digits are same with one of virtual subscriber.

- **End Masked Numbers**: The first part of CLI should be same with one of virtual subscriber and length of CLI should be same with or longer than virtual subscriber.
  Ex) The case that pre-assigned digits of virtual subscriber are ‘450C’ and received CLI is ‘4504875’. Length of incoming CLI is more than 3. The first 3 digits are same with one of virtual subscriber.
- **No Number**: There is no CLI in incoming CO call. The case that pre-assigned digit of virtual subscriber is ‘N’. The priority is same with ‘Whole Numbers’.
- Incoming CLI has numbers from 0 to 9 and *, #.
- In Admin programming, ‘N’, ‘X’ and ‘C’ can’t be used in one CLI type at the same time. And ‘N’ and ‘C’ can’t be assigned more than one in one CLI type.
- In Admin programming, ‘C’ can be only the first or last digit.
- If incoming CLI fulfills one more conditions above, selection of condition follows above priority. The order of priority is as next: Whole Numbers and No Numbers > Prefix Masked Numbers > Postfix Masked Numbers > Beginning Masked Numbers > End Masked Numbers > Length Matching.

3 Virtual Subscriber’s Called Party Number

The Called Party Number from incoming CO call should fulfill one of condition below and real incoming CO group should be same with pre-assigned incoming CO group of virtual subscriber.

- **Whole Numbers**: Whole received Called Party Number (CPN) should be same with pre-assigned digits of virtual subscriber. It has the highest priority.
  Ex) The case that assigned digits of virtual subscriber are ‘4504875’ and received CPN is ‘4504875’.
- **Prefix Masked Numbers**: Length of CPN should be same with pre-assigned digits of virtual subscriber and fixed length end digits of CPN should be same with one of pre-assigned digits of virtual subscriber.
  Ex) The case that pre-assigned digits of virtual subscriber are ‘XXX4875’ and received CPN is ‘4504875’. ‘X’ means any one digit. Length of incoming CPN is 7. The last 4 digits of CPN are same with one of virtual subscriber and any beginning 3 digits are available.
- **Postfix Masked Numbers**: Length of CPN should be same with pre-assigned digits of virtual subscriber and fixed length beginning digits of CPN should be same with one of pre-assigned digits of virtual subscriber.
  Ex) The case that pre-assigned digits of virtual subscriber are ‘450XXXX’ and received CPN is ‘4504875’. Length of incoming CPN is 7. The first 4 digits of CPN are same with one of virtual subscriber and any last 3 digits are available.
- **Length Matching**: Length of CPN should be same with one of virtual subscriber and any digits are available.
  Ex) The case that pre-assigned digits of virtual subscriber are ‘XXXXXXX’ and received CPN is ‘4504875’. Length of incoming CPN is 7.
- **Beginning Masked Numbers**: The last part of digits of CPN should be same with one of virtual subscriber and length of CPN should be same with or longer than virtual subscriber.
  Ex) The case that pre-assigned digits of virtual subscriber are ‘C4875’ and received CPN is ‘4504875’. ‘C’ means any one and more digits. Length of incoming CPN is more than 4. The last 4 digits are same with one of virtual subscriber.

- **End Masked Numbers**: The first part of CPN should be same with one of virtual subscriber and length of CPN should be same with or longer than virtual subscriber.
  Ex) The case that pre-assigned digits of virtual subscriber are '450C' and received CPN is '4504875'. Length of incoming CPN is more than 3. The first 3 digits are same with one of virtual subscriber.
- **No Number**: There is no CPN in incoming CO call. The case that pre-assigned digit of virtual subscriber is N. The priority is same with 'Whole Numbers'.
- Incoming CPN only has numbers from 0 to 9 and *, #.
- In Admin programming, 'N', 'X' and 'C' can't be used in one CPN type at the same time. And 'N' and 'C' can't be assigned more than one in one CPN.
- In Admin programming, 'C' can be only the first or last digit.
- If incoming CPN fulfills one more conditions above, selection of condition follows above priority. The order of priority is as next: Whole Numbers and No Numbers > Prefix Masked Numbers > Postfix Masked Numbers > Beginning Masked Numbers > End Masked Numbers > Length Matching.

4 Incoming CO Group

This is a incoming CO Group for Virtual Subscriber Service. If CO Group is equal with current call group, Virtual Subscriber Service will be checked.

5 Virtual Subscriber's Day/Night/Timed Class

It is used for regular class services. For example, class base O/G digits allow/deny service, CO outgoing service and so on.

6 Virtual Subscriber's Tenant

It is used for regular tenant services. For example, inter tenant connection allow/deny service, class base O/G digits allow/deny service and so on.

7 Virtual Subscriber's Maximum virtual calls

Maximum number of CO incoming calls that virtual subscriber service is applied to simultaneously. If the number isn't assigned, there is no limit of incoming calls. Otherwise, the number can be assigned from 0 to 254. If this value is 0 and Virtual Subscriber Service Option value is ALLOW or MATCH, the matched virtual subscriber's call should be released.

8 Digit Conversion Table

After checking the condition for virtual subscriber service, this Digit Conversion Table is used for toll restriction with temporary Virtual Subscriber's COS.

9 Virtual CLI

- Type: There are two types to make Virtual CLI.

+ ALL - The assigned Virtual CLI is applied for all CO groups and extension.
+ IND(Individual) - The system can make CLI for extensions, CO groups(max. 6) and the others.

- O/G CO Group Number: If the type ALL is selected, the assigned Virtual CLI is used for all CO groups. In case of IND, user is able to assign the Virtual CLI for each O/G CO groups up to 6. And "station" is used only for extension display and "others" is used for all other O/G CO groups which are not assigned in O/G CO groups.
- Table Index: Index of virtual CLI table for the outgoing CO groups and extension. When virtual subscriber makes a call, this Virtual CLI is used. But real incoming CLI is be used if virtual CLI table index is not assigned.

10 Virtual Subscriber's Destination

If virtual subscriber has specific destination number, received called party number (or incoming number) is ignored. The call goes to the destination. Destination number can be extension number, ATD code, group number, outside subscriber number and CO access code. Length of number is 32 digits. If destination is CO access code, system processes the CO access code and called party number. For example, if there is CO access code '9', destination number is '9' and called party number is '8451274', '98451274' will be processed.

11 Virtual CLI Table

Table used for CLI when a virtual subscriber makes outgoing call. Length of number is 24 digits. The maximum of index is 300(MG-300)/100(MG-100).

- Whole CLI: Assigned virtual CLI is sent whatever CLI comes in.
  Ex) The case that assigned virtual CLI is '2793914' and received CLI is '4504875'. CLI for O/G call is virtual CLI '2793914'.
- Begin Copied CLI: Virtual CLI includes some beginning digits of received CLI.
  Ex) The case that assigned virtual CLI is '12BXXXX0' and received CLI is '4504875'. CLI for O/G call is '1245040'. B means to copy digits from the beginning of CLI.
- End Copied CLI: Virtual CLI includes some end digits of received CLI.
  Ex) The case that assigned virtual CLI is 'EXXX1234' and received CLI is '4504875'. CLI for O/G call is '8751234'. E means to copy digits from the end of CLI.
- Combined Copied CLI: Virtual CLI includes some beginning digits and some end digits of received CLI.
  Ex) The case that assigned virtual CLI is 'BXXXEX123' and received CLI is '4504875'. CLI for O/G call is '4505123'.
- In Admin programming, Virtual CLI is assigned up to 24 digits in case of not including 'B' or 'E', 25 digits in case of including 'B' or 'E' and 26 digits in case of including both 'B' and 'E'.
- In Admin programming, 'X' should be after the 'B' or 'E'. And, 'B', 'E' and 'X' don't stand alone.
- If the incoming call's CLI is shorter than masking length of Virtual CLI, only existed digits are used for new CLI. The virtual CLI is 30BXXXXX and received CLI is 100. CLI for O/G call is 30100. The last XX are omitted.
- If there is no incoming call's CLI, only digits in virtual CLI are used for new CLI. The virtual CLI is BXXXXX100 and no received CLI. CLI for O/G call is 100. The masking (BXXXXX) is ignored.

Conditions

1 Virtual Subscriber Service should be processed before Digit Conversion process and ICLID process.

2 Duplicated CLI can be entered in Virtual CLI Table.

3 It is not allowed to assign duplicated Virtual CLI table index in Virtual Subscriber's Table. If a user wants to make same CLI in different O/G CO groups, duplicate the CLI in Virtual CLI Table and assign different index which has same CLI to Virtual Subscriber's Table.

4 The Virtual Subscriber Table and Virtual CLI Table can be assigned up to **300(MG300)/100(MG100)** tables.

5 The CLIP Number should be assigned at Virtual Subscriber Table. If only Called Number is assigned without CLIP number, this call is ignored. But, when CLIP is assigned, the Called Number can be omitted.

Programming

CO Line Data	1. Incoming CO Attributes (PGM 165 – Index17)
System Data	1. Virtual CLI Table(PGM 750)
	2. Virtual Subscriber Table (PGM 751)

Related Features

Hardware

5.28 Private CO Group

Description

One or more users can be assigned exclusive use of a CO line or lines. These CO lines are programmed as Private line for access mode.

Each station can assign CO group access code and CO lines for Private CO.

Operation

Private CO usage: Private CO Only

If all private co line is busy, a user hears busy tone.

Private CO usage: Private & Normal CO

If all private co line is busy, a user can seize normal CO line related to CO group access code.

Private CO usage: Normal CO &Private

If all co lines for CO group access code is busy, a user can seize a CO line in private CO group.

Conditions

1 A private CO group access code should be one of CO group access codes.
2 2 Private CO group can be assigned in each station number.
3 Max 5 CO line can be assigned in each CO group access code.
4 When a user makes a private CO group call, it can be operated with CO Group Access code attributes (PGM180).

Programming

Station Number Data	1. Private CO Attribute (PGM136)
CO Line Data	1. CO Group Access Code (PGM180)

5.29 CO Features

5.29.1 Digit Map Table (Enblock Prefix)

Description

A system can make an outgoing call without inter digit timer even if a user does not dial end mark “#”.

It can analyze dialed digits with Digit Map Table when a user dials the number.

If it matched with Digit Map Digit Table, the call can be made by attributes correspond to Digit Map Digit Table.

The “Digit Map Table” consists of “Digit Map Option table” and “Digit Map Digit Table”.

There are 20 “Digit Map Option Tables” and each NET Numbering Plan Table / CO line / Tenant can have “Digit Map Option table”.

(Priority: Net Numbering Plan Table → CO line → Tenant)

It consists of ‘station/CO line apply type’, ‘Duplication digit timer’ and 20 ‘Digit Map Digit Table index’.

There are 100 Digit Map Digit Tables in a system and each Table has 100 entries.

Each entry of “Digit Map Digit Table” has ‘Digit’, ‘Number of type’, ‘Numbering plan’, ‘Sending Complete’ and ‘Usage’.

- Database of “Digit Map Option/Digit table”

Digit Map Option	Sta Apply Type	NOT USE/Enbloc STA/ ALL-> including CO Access code.								
	CO Apply Type	NOT USE/ Enbloc CO(User Dialed Digits)/ ALL (User Dialed Digits) -> Except CO Access code. Enbloc CO (real PSTN digits) / All (real PSTN digits) -> Except CO Access code.								
	Inter Digit Timer	10-200 (100msec)								
	Digit Map Index (1~100)									
	1	2	3	4	5	6	7	8	9	10
	11	12	13	14	15	16	17	18	19	20

Digit Map Digit 1	Table Comment	Max 32				
	Implicit_digit	Max 32				
	Index	Digit	Usage	Number of T	Numbering P	Sending IE
	1	T010XXXXXXXX	On	Unknown	ISDN	ON / OFF
	2	T01[1-9][2-8]XXXXXX	Off	Unknown	ISDN	ON / OFF
	3	T01[1-9]9XXXXXXX	On	Unknown	ISDN	ON / OFF
	..	94XXX	On	Unknown	ISDN	ON / OFF
	..	9(112|113|119).	On	Unknown	Unknown	ON / OFF
	..					
	..					
	98					
	99					
	100					

A digit of Digit Map Digit Table can be number such as (0~9, *, #) and Option character

- Digit Map Option Character

Option List	Meaning	Example
(Digit A | Digit B)	'OR' option, Digit A or B can be located	(9|801)012 : 9012, 801012
[Digit A - Digit B]	'Range' option, all digit between A and B can be located	[2-4]012 : 2012, 3012, 4012
X	'MASK' option, All digit can be located	9012XXXXXXXX : 9012 + 8 digits
?	'Duplication Digit Timer' option, Duplication digit timer(PGM291) can be applied	9012XXXXXX?? : 9012 + 7~8 digits
T	'Implicit Digit' option, Implicit Digit(PGM292) can be located	T012XXXXXXXX : 9012 + 8 digits (T:9)
.	'End' option, Digit Map analysis can be finished	9010. : 9010 (without duplication digit timer)

- Digit Map Option Rules

	Option	Rules
1	X	The next digit of option 'X' should be 'X','?' or '.'.
2	?	The next digit of option '?' should be '?'.
3	T	The option 'T' can be used only one time.
4	.	The option '.' should be used as last digit.
5	X, ?, .	These options cannot be located in a first digit.

- Example

Digit Map Digit Table Example						
Digit Map Digit Table 1						
Attribute		**Value**				**Explanation**
Comment		Digit Map for Telecomm. Service Provider A				The comment for a table
Implicit Digit		801				Implicit Digit for Digits
Index	**Digit**	**Number of type**	**Numbering Plan**	**Sending Complete**	**Usage**	**Explanation**
1	**(9|801)010**	Unknown	Unknown	With IE	On	9010, 801010
2	**[8-9]010XXXXXXX**	National	ISDN/Telephony Numbering Plan	With IE	On	8010 and 7 digits, 9010 and 7 digits.
3	**9031450XXXX**	Subscriber	Unknown	With IE	On	9031450 and 4 digits.
4	**902350XX??**	Subscriber	Unknown	With IE	On	9031350 and 3~4 digits, the duplication digit timer can be applied to '?' digit.

..	**9031.**	Subscriber	Unknown	Without IE	On	9031 without duplication digit timer
99	**T8210XXXXXXX**	International	ISDN/Telephony Numbering Plan	With IE	On	8018210 and 7digits. (T = 801)
100		Unknown	Unknown	With IE	Off	

Operation

1. System compares digits with Digit Map Option/Digit Table when a user dials digits
2. It matched with the condition of Digit Map Option/Digit Table, system make a call regardless of end mark (‘#’)

Conditions

1. Numbering Plan/Type of Number/Sending Complete IE can be applied about only ISDN/H.323 line.

Programming

CO Line Data 1. Digit Map Option Table Index (PGM170-20)

Voice Network 1. Digit Map Option Table Index (PGM321-11)

Tenant Data 1. Digit Map Option Table Index (PGM280-12)

2. Digit Map Option Table (PGM 291)

3. Digit Map Digit Table (PGM 292)

Related Features

Hardware

5.29.2 Multiple SIP UID selection

Description

SIP UID can be selected by CLI type. This CLI type is based on DN or CO Transit number.

Compatibility

Operation

Operation of this feature is automatic when assigned.

Conditions

1. The station CLI number Long CLI1, Long CLI2, Long CLI3 corresponding to SIP UID1, SIP UID2, SIP UID3 in Station DN Attribute(PGM 135).
2. The CO transit CLI number Long CLI1, Long CLI2, Long CLI3 corresponding to SIP UID1, SIP UID2, SIP UID3 in Incoming CO Attribute(PGM 165).
3. This SIP UID1, 2, 3 have high priority from DN SIP UID and CO group SIP UID.

Programming

Station	1. Station DN Attribute (PGM 131-Index11) 2. Station DN Attribute (PGM 135-Index16, 17, 18)
CO Line Data	1. Incoming CO Attribute (PGM 165-Index19, 20, 21) 2. SIP CO Additional Registration (PGM 371-Index1)

Related Features

Hardware

5.29.3 Multiple SIP UID 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 configured, Alternative Route Selection operation works automatically.

Conditions

1. ARS is optional and must be programmed to be operational.
2. Up to 2 ARS paths can be assigned for each CO Group Access code.
3. Alternative route can be assigned as a user assigns CO Group Access Code in ARS CO group access code.
4. ARS can be used in coordination with Last Number Redial, Station Speed Dial, and System Speed Dial.
5. When ARS is provided, differential digits can be sent according to ARS digit usage (converted digits, original digits, converted digits with digit conversion table or original digits with digit conversion table).
6. If ARS digit usage is set to "converted/original digits with digit conversion table", then, digits to send PSTN can be converted one more according to ARS digit conversion table.

Programming

Numbering Plan	1 CO Group Access Code (PGM 114)
CO Line Data	1 CO Group Access Code Attributes (PGM 180)
	− ARS Service (Index 6)
	− ARS CO Group Access code (Index 7, 10)
	− ARS Digit Usage (Index 8, 11)
	− ARS digit conversion table (Index9, 12)

Related Features

Hardware

5.30 T1PRI Caller Name Service

Description

The PRI(T1) lines support the caller name service in the call message. According to the network provider, the caller name may be included in Display Information Element or Facility Information Element.

Operation

Conditions

1. To make caller name for outgoing calls, the caller name in PGM 171-Index 14 should be set.
2. If the network provider type is NI-1, the caller name for outgoing calls will be made into Display Information Element.
3. If the network provider type is NI-2, the caller name for outgoing calls will be made into Facility Information Element.
4. If the network provider type is NI-1, the system will check only Display Information Element for the caller name of incoming calls.
5. If the network provider type is NI-2, the system will check only Facility Information Element for the caller name of incoming calls.

Programming

CO Line Data	1. Outgoing CO Line Attributes (PGM 171–Index14) : Caller Name (New)
Board Data	1. ISDN/Digital Board Attribute (PGM 300) : Caller Name Type (New)

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 & LIP-8000E Series (8040E/8040L, 8024E/8024D, 8012E/8012D, 8004D)
  - LDP-6000 (6030D, …)
  - LDP-7000 & LDP-9000 Series (7024LD, 7024D, 7016D, 7008D, 7004D, 7004N. 9008D, 9030D)
  - 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/CALL BK]** button. Or, press the **{ACNR}** soft button.
2. Hang-up handset, or press **[SPEAKER]**.

<u>To cancel ACNR while phone is in Idle state:</u>

1. Press the flashing **[MSG/CALL BK]** button.

<u>To cancel ACNR during an ACNR attempt:</u>

1. Lift the handset, press the **[MUTE]** button and go on-Hook.

Conditions

1. The applicable Timers and Retry Counter must be programmed,
   − ACNR Pause Timer: Determines the time allowed between ACNR retires.
   − 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. 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.
2. Dial 24 for LIP-series terminals or dial 23 for non LIP-series.
3. Select the desired **{BGM Source}**.
4. 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	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 16 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 are 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 people in the same place to listen the conversation to both parties.

Operation

Digital Phone

<u>*While on a call using the Handset:*</u>

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 called user can hear the ICM caller's voice after splash tone. The called user is not needed to lift the Handset or press the **[SPEAKER]** button to answer the call.
- **Call Announce with Privacy (P)** – When an ICM call is received, the called user can hear the ICM caller's voice after splash tone. The called 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

<u>*To change ICM Signaling Mode:*</u>

1. Press the **[PGM]** button; the **[SPEAKER]** button LED will light steady.
2. Dial 11 **{Station User Program code}**.
3. Dial the desired ICM Signaling Mode code (1: H, 2: T, 3: IP).
4. Press the **[SAVE]** button.

Conditions

1. Callback call and forward call 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

<u>If set, Off-Hook Ring signal operation is automatic.</u>

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

<u>*To access the Station Auto Digit Dial:*</u>

1. Lift the Handset or press the **[SPEAKER]** button; take no action (wait), Auto Digit Dial will be accessed as designated.

Conditions

1. Auto Digit Dial 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 Auto Digit Dial 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 Auto Digit Dial 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

<u>*To download a Ring Tone from System memory:*</u>

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.

<u>*To select the downloaded Ring Tone:*</u>

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

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 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 dial the desired number to create the Telephone number button.
5. 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. The 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	Numbering Plan (PGM 110–118)
Station Port Data	2	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 Flashing Late (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 **[VOLUMN 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.

Table 6.20-1 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	CO Ring Type	
23	Ring Download	LIP-Series Only
24	Back Ground Music	LIP-Series Only PGM 23 for other terminal
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	Activate a mobile service by CLI	
55 + x	Register CLI serve the mobile extension	X=1-5
56+ Room No. & Auth Code	Start a Conf Room	

Table 6.20-1 User Program Code Chart

User PGM Code	Description	Remark
57 + Room No. & Auth Code	Close a Conf Room	
61	Speaker/Headset Mode	Speak/Headset/E-MIC or BTU
62	Headset Ring Mode	Speaker/Headset/ Speaker, Headset/Ear-Mic
71	Register Station ICLID	
72	View Station ICLID	
73	Activation Out Call Notification	
74	Set Out Call Notification Attempts	
75	Set Out Call Notification Interval	
76	Register Out Call Notification Number	
77	VM Forward Reroute Destination	
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	

Table 6.20-2 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 + Room No. & Auth Code	Start a Conf Room	
55 + Room No. & Auth Code	Close a Conf Room	
71	Register Station ICLID	

User PGM Code	Description	Remark
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, 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:*

1. Press **[PGM]** + **{FLEX}** + Button Feature Type(1) + **{Two-Way Record Feature Code}** + **[SAVE].**

To activate Two-Way Record:

1. While on an active internal or 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.*

To stop Two-Way Record:

1. While on an active internal or CO/IP Call that is being recorded by the Station, press the **{RECORD}** button again. Or, 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, 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 VMIB, Internal Calls can be recorded as well as External Calls.
5. Conference Call cannot be recorded.
6. 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 Number Data	1	Auto Record Service (PGM 145 – Index 3)
	2	Two-way Record Access (PGM 145 – Index 4)
	3	Two-way Record Destination (PGM 145 – Index 5)
Tenant Data	1	Tone Table (PGM 290)

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 the voice message being recorded and also can answer the call while recording 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 voice conversations being saved 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 any more. If user presses the **{Mute}** button, the conversation will start and recording will be continued at the same time.

Operation

Digital Phone

*To assign a flexible button as **{AME}**:*

1. **[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, press **{Mute}** button to communicate with saving conversation.

<u>To Answer a call in Speaker Hearing Mode:</u>

1. When a call is forwarded to VMIB, press the **{AME}** button to communicate without saving conversation. Or, press **{Mute}** button to communicate while saving conversation.

Conditions

1. VMIB Access feature has to be ON (enabled)
2. AME Function will be started when VMIB Forward feature is set.

Programming

Station Data	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}** that is selected in Tenant Intercom Busy One-Digit service. Or, press a pre-programmed **{OHVO}** button.
2. 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

6.24 Override (Hold, Disconnect)

Description

When a user calls to a station and receives a busy signal, the user can request Override(Hold or Disconnect).

Override - Hold

A user makes a conversation with busy station directly and the party talking with busy station goes to hold state.

Override - Disconnect

A user makes a conversation with busy station directly and the party talking with busy station is disconnected.

Operation

With One Digit Service

To activate a 'Override–Hold' while receiving Intercom busy tone:

1. Dial the digit programmed as 'Override-Hold', the called station will receive the Call Wait Alarm tone
2. The party talking with busy station goes to hold state and a conversation with overriding station will be established automatically.

To activate a 'Override-Disconnect' while receiving Intercom busy tone:

1. Dial the digit programmed as 'Override-Disconnect', the called station will receive the Call Wait Alarm tone
2. The existing conversation is terminated and a conversation with overriding station will be established automatically.

With Feature Code

To activate a 'Override-Hold' while receiving Intercom busy tone:

1. Press Hook Flash or **{TRANS}** button
2. Dial the feature code for 'Override-Hold', the called station will receive the Call Wait Alarm tone
3. The party talking with busy station goes to hold state and a conversation with overriding station will be established automatically.

To activate a 'Override-Disconnect' while receiving Intercom busy tone:

1. Press Hook Flash or **{TRANS}** button
2. Dial the feature code for 'Override-Disconnect', the called station will receive the Call Wait Alarm tone
3. The existing conversation is terminated and a conversation with overriding station will be established automatically.

Conditions

1. Override -Disconnect cannot be supported to SIP terminal.

Programming

Station Number Data	1. Voice Over Access (PGM 133-Index9)
	2. Rejection of Voice Over(PGM 133-Index10)
System Data	1. Intercom Busy One-digit Attributes (PGM 237)
Numbering Plan	1. Feature Numbering, Override & Disconnect (PGM 113)

Related Feature

- Voice Over

Hardware

6.25 USB Auto Call Record

Description

The conversation of a station can be recorded automatically by an optional USB module installed in LDP 7000 keysets. Recording is started immediately when the conversation begins, and finished once the station goes on hook.

Operation

To enable "USB Auto Call Record":

1. Lift the handset or press **[SPEAKER]** button.
2. Dial **{Automatic Call Record Mode}** feature code.
3. Dial 2 to enable "USB Always Call Record"
4. Press **[SAVE]** button.

To disable "USB Auto Call Record":

1. Lift the handset or press **[SPEAKER]** button.
2. Dial **{Automatic Call Record Mode}** feature code.
3. Dial 0 to disable automatic recording.
4. Press **[SAVE]** button.

Conditions

This feature is available only from LDP-7000 series. And USB module must be installed.

Programming

Numbering Plan	1. Auto Call Record Mode Feature Code (PGM 113)
Station Data	1. Auto Record Service (PGM 145 – Index 3)

Related Features

Hardware

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 Attributes (PGM 271 - 272)
	3	Night Group Assignment (PGM 275)
	4	Night Group Attributes (PGM 276 - 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 Group Attributes (PGM 276 - 277)

Related Features

Hardware

7.3 Greeting/Queuing Tone Service

Description

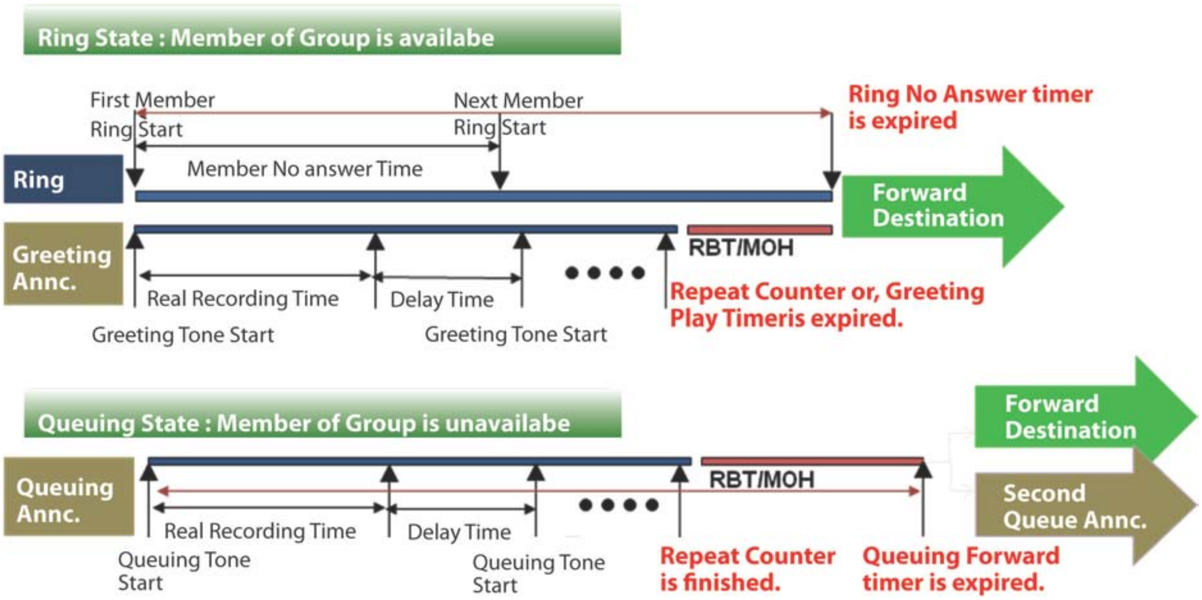
The System can provide a Greeting tone or Queuing tone when a call is routed to an Attendant Group or Night Attendant Group.

The tone will be provided according to the Tone Time/Delay Time/Repeat counters. There are 7 types of Tones:

- **NORMAL** – System Tone (01–19, Tone Freq. in PGM 264)
- **PROMPT** – VMIB Prompt
- **ANNOUNCEMENT** – VMIB Announcement
- **INT MOH**
- **EXT MOH**
- **VMIB MOH (1–4 for MG-300, 1–3 for MG-100)**
- **SLT MOH (1–5)**

Operation

If set, Greeting/Queuing Tone Service operation is automatic.



Conditions

1. If Greeting Play timer is expired, RBT is provided.
2. During Announcement delay time, MOH or RBT can be provided by PGM.
3. If queuing announcement forward timer is expired, second queuing announcement can be provided.
4. If Second queuing announcement forward timer is not assigned, a call is routed to the forward destination after first queuing announcement forward timer is expired.

Programming

Tenant Data	1	Attendant Group Attribute (PGM 271–272)
Table Data	1	Tone Frequency/Cadence (PGM 264)

Related Features

Hardware

7.4 CCR Service For Attendant queuing announcement

Description

The System can provide CCR Service during queuing announcement according to the CCR option.

A CCR Table defines a dialed digit (0–9, #, and *) to a designated route; each individual digit corresponds with a route:

- Station Number
- Station Group
- ACD Group
- Announcement Table
- Announcement Table And Drop
- System Speed
- Conference Room
- Attendant Call
- VMIB Access
- Networking Number
- Internal Paging
- External Paging
- Internal/External All Paging
- Company Directory
- Record VM Greeting
- Direct VM Transfer
- Digits

When the User dials a Station number, Group Queuing Service is finished and a call is routed to the dialed destination.

Operation

<u>If set, Greeting/Queuing Tone Service operation is automatic.</u>

Conditions

1. SIP/ISDN Terminal does not support CCR feature.

Programming

Tenant Data	1	Station Group Attribute (PGM 201–202)
Table Data	1	Tone Frequency/Cadence (PGM 264)
	2	Announcement Table (PGM 259)
	3	CCR Table (PGM 260)

Related Features

7.5 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

7.6 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

7.7 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.

Table 7.7-1 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 NON-STA SMDR	
014	DELETE NON-STA SMDR	
015	PRINT ALL SMDR	
016	DELETE 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

Table 7.7-1 Example of User Program Code Chart

User PGM Code	Item Description	Remark
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
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 FROM 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, use the **[VOL UP]**/**[VOL DOWN]** button to display the desired menu item and enter the desired code.
4. 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

7.8 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

7.9 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

<u>*To modify Day/Night/Timed Ring Mode manually (Attendant Only):*</u>

1. Press the **[PGM]** button and dial 081 Or, dial the **{Day/ Night Program Feature Code}**.
2. Select Tenant Number (0=All, or 1–9).
3. Select Auto Ring mode (0=AUTO, 1=DAY, 2=NIGHT, 3=TIMED).

<u>*To set Day/Night/Timed Ring Mode automatically (Auto Service Mode Control):*</u>

1. Press the **[PGM]** button and dial 081. Or, dial the **{Day/ Night Program Feature Code}**.
2. Select Tenant Number (0=All, or 1–9).
3. 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

Numbering Plan	1	Feature Numbering Plan (PGM 113)
Table Data	1	System Time Table (PGM 253/254)
Tenant Data	1	Attendant Group Assign (PGM 270)
	2	Tenant Attribute 1 (PGM 280)

Related Features

- System Clock Set
- Loud Bell Control (LBC)
- Dialing Restrictions
- Auto Service Mode Control

Hardware

7.10 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.

In case of LIP-8000 Phones, only serial DSSs can be connected with them initially. Serial DSS doesn't have its station number and connect physically with LIP-8000 phone. To distinguish general DSS/DLS consoles with serial DSS, general DSS/DLS consoles are named as station DSS. Station DSS has a station number and occupies one station port. To connect station DSS with LIP-8000 phone, DSS Mode for LIP-8000 phone must be configured as station DSS.

Operation

To connect Station DSS to Digital Phone (except for LIP-8000 Phone):

1. Check station number of Station DSS which will be connected.
2. Check DSS map of Digital Phone in Station Type (Web Admin PGM120).
3. Enter DSS's number at DSS map of Digital Phone.
4. Press the **[SAVE]** button.

To connect Serial DSS to LIP-8000 Phone:

1. Check DSS map of LIP-8000 Phone (Web Admin PGM120).
2. Select serial DSS type.
3. Press the **[SAVE]** button.

To connect Station DSS to LIP-8000 Phone:

1. Set DSS Mode of LIP-8000 as station DSS.
2. Check station number of Station DSS which will be connected.
3. Check DSS map of LIP-8000 Phone in Station Type (Web Admin PGM120).
4. Enter DSS's number at DSS map of LIP-8000 Phone.
5. Press the [SAVE] button.

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)
	2	LIP-8000 Phone DSS Mode (PGM122-Index9)
Tenant Data	1	Attendant Group Assign (PGM 270)

Related Features

- Station Flexible Buttons
- DSS/BLF

Hardware

7.11 The Usage of LIP-7048DSS

Description

LIP-7048DSS can be registered with remote mode.

Operation

Using Individual Web Admin of LIP-7048DSS, register mode can be modified to remote mode.

7.12 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

7.13 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	1	System Date & Time (PGM 233)

Related Features

- Least Cost Routing (LCR)
- Station Message Detail Recording (SMDR)
- Auto Service Mode Control
- Day/Night/Timed Ring Mode

Hardware

7.14 USB Upgrade

Description

The Attendant can upgrade the System via USB memory. USB upgrade could be executed using the Attendant Keyset. 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 Keyset:

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 Keyset will return to Idle state.

```
SOFTWARE UPGRADE
USB UPGRADE SUCCESS
```

9. Restart the System.

Conditions

1. USB Upgrade in Attendant Keyset can support up to 10 ROM image files.

Programming

Tenant Data	1	Attendant Group Assign (PGM 270)

Related Features

Hardware

7.15 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 Keyset:

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 Keyset will return to Idle.

```
DB DOWNLOAD TO USB
DOWNLOAD SUCCESS
```

To upload the System Database using the Attendant Keyset:

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 Keyset 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 Keyset.
2. Download can be serviced by TELNET ('usb down' in maintenance mode).

Programming

Tenant Data	1 Attendant Group Assign (PGM 270)

Related Features

Hardware

7.16 IP Attendant

Description

IP 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. IP Attendant works independently from digital phone (i.e. hardware terminal) while ez-Attendant operates in conjunction with the Attendant Digital Phone. IP Attendant has its own station number and is registered to system like phontage or UCS client.

Operation

Attendant

Operation of IP Attendant is described in detail in the IP Attendant Installation and User Guide

Conditions

Programming

Tenant Data	1 Attendant Group Assign (PGM 270)

Related Features

Hardware

7.16.1 IP Attendant Lock-key

Description

IP Attendant can't use Lock-key of ezAttendant any more. To use IP Attendant, IP Attendant dedicated lock-key should be existed. There are 2 types of IP Attendant Lock-key. One is for Office Version. The other is for Hotel Version.

Operation

[Lock Key Install]

Key : 000DA2D407000001		
Index	Value	State
Network(Q-Sig)		Installed
ez Attendant		Installed
CTI(TAPI 3rd)		Installed/Disconnect
PHONTAGE		Basic : 5 Deluxe : 5 (2 copies free)
UCS Client		5 Copy
Web Phone		0 Copy
TNET(Local-Survival)		Installed
NMS		Installed
SIP Phone		5 Copy
ez Phone		Basic : 4 Deluxe : 0
FIAS		
iPECS IPCR		Server : 1 Client : 10
iPECS Communicator		10 Copy
IP Attendant		Office : 5 Hotel : 5
Demo Key		expired
	Save	

Conditions

Programming

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 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 operated, dropped or ignored.

- Flash-Normal – When Hook-Flash detected, the previous Line will be continued and it 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 Data	1	Howling Tone (PGM 121 – Index7)

Related Features

- Intercom Lock-Out

Hardware

8.4 SLT Message Wait Indication

Description

SLT can receive a Alert Message Wait 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 F1 or Sub Type in Web)

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.

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 Entry Chart).

Table 8.5-1 Alphanumeric Entry Chart

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

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 Ericsson-LG Enterprise SIP-based phone and 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 SIP Station Basic Registration Table.
2. Enter the User ID Auth ID and Password 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. Reboot 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

Pre-Programmed Data	1	Logical Slot Assignment (PGM103)
	2	DECT/IP/IP Max Port (PGM104)
SIP Station Data	1	SIP STA Basic Registration (PGM380)
	2	SIP STA Additional Registration (PGM 381)

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. For this feature, SIP phone should have Call Hold Function.

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. Only Transfer Hold is supported for SIP phone. The counterpart will hear Transfer Hold tone when it is held by a SIP phone.
2. For this feature, SIP phone should have Call Hold Function.
3. Though the SIP phone has a Call Hold Function, it could 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. For this feature, SIP phone should have Call Transfer Function.

Operation

To Transfer an Active Call:

1. While on an Active Call, press the **[TRANSFER]** button on SIP Phone.
2. Dial a station Number or an external telephone number where the call will be transferred.

Conditions

1. For this feature, SIP phone should have Call Transfer Function.
2. For the exact operation, refer to your SIP phone manual.
3. 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

1. Refer to the SIP Phone User Guide to set-up Call Forward.
2. Refer to [3.11 CALL FORWARD] if you want to register Call Forward by the system feature code.

Conditions

1. SIP phone can use the system feature code for Call Forward function.
2. 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 be placed in DND to block incoming ringing for CO/IP and Intercom calls, transfers, and paging announcements

Operation

To Activate DND:

1. Refer to the SIP Phone User Guide to set up Call Forward.
2. Refer to [3.11 CALL FORWARD] if you want to register Call Forward by the system feature code.

Conditions

1. SIP phone can use the system feature code for DND function.
2. 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

- 3.32 DO NOT DISTURB (DND)

Hardware

9.9 SIP 3-Party Conference

Description

The SIP phone can make a 3-party conference.

Operation

1. 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

1. SIP phone user A in conversation with user B is called from user C.
2. User C is placed on Call Wait state automatically.
3. User A can alternate calls between user C and user B.

Conditions

1. To use this feature, the SIP phone must be enabled with Broker Call function.
2. 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.

To use this feature, the SIP phone must be enabled with SMS function.

Operation

1. 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

SIP phones such as IP-8800/6800 can receive a notification when another Station leaves Voicemail.

To use this feature, 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 such as IP-8850 can make a Video Call using a Video Phone. To make a Video Call, the Stations must be enabled for Voice Calling.

Operation

1. 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.

Programming

Related Features

Hardware

9.14 SIP Phone Voice Mail Notification

Description

When a user has left a voice message on an SIP terminal, some SIP terminals including IP-8800 can notify the user of voice messages. This function requires the VM notification function to be supported in SIP terminals.

The [MSG] LED will flicker if there is a voicemail saved for the user.

Operation

SIP Phone

<u>*To retrieve voice mail:*</u>

1. When **[MSG]** notification is present, press the **[MSG]** button; the number of new messages and saved messages are displayed.
2. Dial the **{Voice Mail Access}** feature code.

Conditions

1. To use this function, the SIP terminals must support VM notification.

Programming

Related Features

Hardware

- SIP phone supporting extended “Alter-Info” function (IP-8800)

9.15 SIP Phone BLF

Description

The BLF function is available in the terminals with BLF device (IP-8800 series). This function enables users to check the status of other extension, and to perform functions like pick-up and call transfer.

Operation

SIP Phone

To register BLF:

1 On the terminal program, assign the BLF function to the FLEX button.

2 Check the status of other users with the registered button.

To use BLF:

1 Access the SIP terminal web setting screen.

2 Register an extension in the **[BLF]** button of the SIP terminal.

3 The status of the registered extension is displayed on the **[BLF]** button.

- Busy: LED on
- Incoming ring: LED flickers
- Not used: LED off

To perform Call Pick-up:

1 The **[BLF]** button flickers due to an incoming ring to the registered BLF user.

2 By pressing the flashing button, you can pick up the incoming call.

Conditions

1 This function is only supported in the SIP terminals that comply with the Ericsson-LG Enterprise SIP-extended I/F spec (IP88xx).

Programming

Related Features

Hardware

- Ericsson-LG Enterprise SIP phone (IP88xx)

9.16 SIP Phone Distinctive Ring

Description

This function enables the system to provide different rings depending on whether the call is from an extension user or a trunk user.

Operation

SIP Phone

To program distinct ring based on the call type:

1 Access the SIP phone web setting window.
2 Set different rings for internal calls and trunk calls.
3 When a call is incoming, the ring is provided depending on whether the call is from internal or trunk user.

Conditions

1 This function is only supported in the SIP terminals that comply with the Ericsson-LG Enterprise SIP-extended I/F spec (IP88xx).

Programming

Related Features

Hardware

- Ericsson-LG Enterprise SIP phone (IP88xx)

9.17 SIP Phone Intercom

Description

This function automatically opens the speaker for incoming calls, so that the users can speak without lifting the handset.

Operation

SIP Phone

If the terminal receiving mode is Handsfree:

1 On Web-ADMIN, change the internal call receiving mode to “H”.

2 For incoming call, the terminal automatically opens the speaker and answers the call.

If “forced change of extension call answer mode” is activated for the extension with the terminal receiving mode “T”:

1 The calling party presses the **{Forced Hands Free Call}** feature code to place a call to an extension.

2 If a call is incoming, the terminal automatically opens the speaker to answer the call.

Conditions

1 This function is only supported in the SIP terminals that comply with the “Alert-Info” function. (IP-68xx, IP-88xx)

Programming

Related Features

Hardware

- SIP phone supporting extended “Alter-Info” function (IP-8800)

9.18 SIP Phone Call-Back

Description

If the dialed number is on busy, the caller may register “call-back” to the called party. If call-back is assigned, the system sends the ring to the calling party if the call is terminated, and the called party returns to idle.

Operation

SIP Phone

To register call-back:

1 Place a call to a busy extension user.

2 Receive a busy tone.

3 On-hook, and dial the **{Call Back/Queuing Register}** feature code + Extension Number.

4 You will hear the confirmation tone.

To answer call-back:

1 The busy extension returns to idle.

2 The call from the extension is incoming automatically.

3 Answer the call.

Conditions

1 This function is only supported in the SIP terminals that comply with the “Alert-Info” function (IP-68xx, IP-88xx)

Programming

Related Features

Hardware

- SIP Phone

9.19 SIP Phone Call Intrusion

Description

Users can attempt intrusion into the bus user. If intrusion is made successfully, a 3-party conference is made including the extension that has attempted an intrusion.

Operation

To perform call intrusion:

1 Place a call to a busy extension.

2 Receive the busy tone.

3 On-hook, and dial the **{Intrude Request}** feature code + Extension number.

4 A 3-party conference is made between the parties that have been in the call, and the new user who intruded into the call.

Conditions

1 Allowing intrusion into busy SIP phone is only supported in the terminals that comply with Ericsson-LG Enterprise SIP extended I/F spec. (LIP-88xx)

2 Call intrusion is supported in any SIP terminal.

Programming

Related Features

Hardware

- Ericsson-LG Enterprise SIP phone which is supporting SIP extension I/F (LIP-88xx)

9.20 SIP Phone Call Override

Description

If a user attempts call override, the user in the current call is put on hold, and a connection is made between the called party and the user that attempted call override.

Operation

SIP Phone

To connect a call override

1. Place a call to a busy extension.
2. Receive the busy tone.
3. On-hook, and dial the **{Call Override}** feature code + Extension number.
4. The current call of the called party is put on hold, and the connection is made with the extension that attempted call override.

Conditions

1. Allowing override into busy SIP phone is only supported in the terminals that comply with Ericsson-LG Enterprise SIP extended I/F spec. (LIP-88xx)
2. Call override can be attempted in any SIP terminal.

Programming

Related Features

Hardware

- Ericsson-LG Enterprise SIP phone which is supporting SIP extension I/F (LIP-88xx)

9.21 iPECS-Communicator:Android & iOS Version (Phase 2)

Description

iPECS-MG supports the iPECS Communicator (Mobile SIP Client) iOS Version as well as Android Version. 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

Conditions

1. Lock key for Android / iOS Communicator is needed.
2. This lock key is based on the number of copies.

Programming

Pre-Programmed Data	1 Logical Slot Assignment (PGM 103) 2 DECT/IP/SIP Max Port (PGM104)
SIP Station Data	1. SIP Station Basic Registration (PGM 380) 2. SIP Station Additional Registration (PGM 381)

Related Features

Hardware

10. ACD (AUTOMATIC CALL DISTRIBUTION)

Description

ACD (Automatic Call Distribution) feature provides the service to distribute calls to agents in an efficient way. Each agent can set or change own specific state and get ready to receive the ACD calls. And supervisor can be assigned to each group and they can change the ACD group status.

Table 10-1 ACD Group Capacity

Items	iPECS-MG 100	iPECS-MG 300
Number of ACD Group	20	50
Number of Supervisor	1	1
Number of Sub-Supervisor	3	3
Number of Agents	50	50
Max Queue Count	99	99
Max Steps for Queue Announcement	5	5
ACD Agent Priority	20 (1 ~ 20)	20 (1 ~ 20)

10.1 ACD Basic Feature

Description

ACD calls are analyzed first by the system to find an appropriate agent who will receive the call. System selects one of the free agents and then directs the ACD call to the agent selected.

Each agent registered in admin has to log in first to receive an ACD call. If all the agents are in busy status or in logout status, the next ACD calls will be queued. When one of the agents goes to ready state or idle state, the queued ACD call is routed to the agent in ready state. While an ACD call is being queued, queuing announcement will be served.

- ACD Group has 5 types of status (Normal / Forward / Overflow / Night / Holiday Status). And in each status ACD calls can be handled.
- Each ACD group can have one Supervisor and three Sub-Supervisors. Supervisor or Sub-Supervisor can monitor the state of agents and can also check the group's call traffic.
- Each ACD group can have max. 50 agents. To answer the ACD call, agents have to log-in.
- Administrator or Supervisor can assign priority to each agent. So when ACD call arrived, at first higher priority group's agents will be received the ACD call, and then all of agents of higher priority group are busy, next priority group's agent will be received next ACD call.
- Each ACD group can have max. 5 Queuing ACD announcements. And also they have Night, Holiday and Overflow announcement.

Operation

Agent Log-In

1. Dial **{ACD Agent Log-In/Out}** feature code. Or, press flex button registered as **{ACD Agent Log-In/Out}** feature code.
2. If agent is in log-out state, agent is put in log-in state. The flex button registered as **{ACD Agent Log-In/Out}** feature code will be turned on.
3. If [Password Check When Agent Login] is set in admin [PGM 214 – Flex8], agent has to enter password on log-in.

Agent Log-Out

1. Dial **{ACD Agent Log-In/Out}** feature code. Or, press flex button registered as **{ACD Agent Log-In/Out}** feature code.
2. If agent is in log-in state, agent is put into log-out state. The flex button registered as **{ACD Agent Log-In/Out}** feature code will be turned off.

ACD Call Operation

1. If an ACD call arrives, system automatically finds an idle agent (not in DND and not in Work Mode).
2. If there are more than one idle agents, system check the priority of each agent and then one of longest idle agents will be chosen from the highest priority group.
3. If there is no available agent, the ACD call will be queued and ACD announcement service is started.

Conditions

1. Only My-DN number can be assigned as Supervisor or Sub-Supervisor or Agent Members.
2. (LIP-7004/LIP-8004/LDP-6004/LDP-7004/LDP-7008)These kinds of Keysets cannot be assigned as Supervisor or Sub-Supervisor.
3. SIP Phone keyset cannot be assigned as Supervisor or Agent.
4. If one My-DN number is assigned for Supervisor, it cannot be a Sub-Supervisor or an Agent member. If one My-DN number is assigned for Sub-Supervisor, it cannot be a Supervisor or an Agent. And also if one My-DN number is assigned for an Agent member, then it cannot be a Supervisor or a Sub-Supervisor.
5. If [Announcement Use When Incoming CO Call] is set in admin [PGM 214 – Flex5], when agent answer the ACD call form CO line has, at first Answer Announcement will be played before conversation.
6. If [Agent-Agent Call Restriction] is set as Direct Call Restriction in admin [PGM 214 – Flex2], agents cannot make a direct call to other agent.
7. If [Call Restriction When Agent Logout] field is set as All-Call in admin [PGM 214 – Flex14], no calls are allowed in an agent's terminal in logout status.
8. If [Call Restriction When Agent Logout] filed is set as CO Outgoing Call in admin, CO Outgoing calls are not allowed in an agent's terminal in logout status, but internal calls are not restricted.
9. Administrator can assign priority to each agent in admin [PGM 124 – Flex11]. Range of priority is from 1 to 20. Lower numbers have higher priority.
10. If a user wants to transfer a call to an ACD group, this call will be handled as only unscreened transfer feature. So, when user want to make transfer to ACD group, the call will be directly transferred to ACD group and the transfer initiator goes to idle state automatically.

Programming

Numbering Plan	1	Feature Numbering Plan (PGM 113)
	2	ACD Numbering Plan (PGM 118)
Station Port	1	Station Port Attribute (PGM 124)
Station Group	1	ACD Group Assignment (PGM 212)
	2	ACD Group Attribute1 (PGM 213)
	3	ACD Group Attribute2 (PGM 214)

Related Features

Hardware

10.2 ACD Group Service Status

Description

ACD group has 4 statuses (Normal / Forward / Night / Holiday).

1. Normal status is a general service status.
2. Forward status, ACD call will be forcibly forwarded to registered destination.
3. Night a Holiday status can be changed automatically by system time-table or can be changed manually by Supervisor. In each status Administrator or Supervisor can make rule how to handle the ACD call. ACD calls can be released immediately, or can be served with Night or Holiday Announcement, or can be forwarded to the registered destination.
4. Overflow status will be changed automatically from Normal status when all agents are busy and also ACD queue is full. Administrator or Supervisor can make rule release, overflow announcement or forward.

In case of Forward, Night, or Holiday status, these kinds of status can be set manually by system Administrator or group Supervisor. And also in case of Night, or Holiday status, if [Auto service change] admin is set PGM 212 – Flex5], group status can be changed automatically according to the system time table.

Operation

Group Forward Status Manually by Supervisor

1. Dial **{ACD Supervisor Group Call Forward}** feature code. Or, press flex button registered as **{ACD Supervisor Group Call Forward}** feature code.
2. **{ACD Supervisor Group Call Forward}** flex button is turned on steadily.

Night Status Manually by Supervisor

1. Dial **{ACD Supervisor Group Night Mode}** feature code. Or, press flex button registered as **{ACD Supervisor Group Night Mode}** feature code.
2. **{ACD Supervisor Group Night Mode}** flex button color is turned on steadily.

Holiday Status Manually by Supervisor

1. Dial **{ACD Supervisor Group Holiday Mode}** feature code. Or, press flex button registered as **{ACD Supervisor Group Holiday Mode}** feature code.
2. **{ACD Supervisor Group Holiday Mode}** flex button color is turned on steadily.

ACD Call Operation when Group Forward Status

1. ACD call arrives and ACD group is in forward status.
2. System finds the registered destination and the call is forwarded.

ACD Call Operation when Group Night or Holiday Status

1. ACD call arrives and group status is in overflow, night, or holiday status.
2. System checks the registered rule.
3. If the defined rule is to release the call, ACD call will be released immediately.
4. If the defined rule is to play announcement, prerecorded announcement is played first and then ACD call will be released after announcement.
5. If the defined rule is to forward the call, ACD call will be forwarded to the registered destination.

Conditions

1. Group status can be changed only by supervisor or administrator.
2. Supervisor can change the rule for ACD group in its Station Web Program.
3. If [Password Check When Service Mode Changed] is set in admin [PGM 214 – Flex1], supervisor has to enter the correct password when supervisor wants to change the group status (Group Forward / Night / Holiday Status).
4. If [Auto Service Status Change] is set in admin [PGM 212 – Flex5], group status will be changed automatically from Normal to Night or Holiday status according to system time table.
5. If [Auto Service Status Change] is set in admin [PGM 212 – Flex5] and the System-Time is in Timed mode, ACD will follow the settings for Night Service.
6. If [Auto Service Status Change] is set as Manual Change [PGM 212 – Flex5], supervisor can change group status manually.

Programming

Station Port	1	Station Port Attribute (PGM 124)
Station Group	2	ACD Group Assignment (PGM 212)
	3	ACD Group Attribute1 (PGM 213)
	4	ACD Group Attribute2 (PGM 214)

Related Features

Hardware

10.3 ACD Call Distribution By Priority

Description

Administrator or Supervisor or Sub-Supervisor can assign or change priority of agents. There are 20 levels as priories; low level value is means higher priority. Administrator or Supervisor or Sub-Supervisor can make sub-group of agents with priority level In a ACD Group.
When ACD call arrived, at first higher priority group's agents will be received the ACD call, and then all of agents of higher priority group are busy, next priority group's agent will be received next ACD call. Administrator can make priority of agent in [ACD Agent Priority] admin PGM124 – Flex11. And Supervisor or Sub-Supervisor can make priority in their Station Web Program page in Web admin.

Operation

Change Priority of agents by Supervisor or Sub-Supervisor

1. Enter Station Program page in Web-Admin of site with password of Supervisor or Sub-Supervisor.
2. Go to ACD Group Management field in Station Program.
3. Only Supervisor or Sub-Supervisor can access the ACD Group Management field.
4. Supervisor or Sub-Supervisor can change priority of Agents.

Condition

1. Agent's Priority can be applied when agent goes to log-in and ready state.
2. Supervisor and Sub-Supervisor can change the priority at their Station Program in Web Program.

Programming

Station Group	1	ACD Group Assignment (PGM 212)
	2	ACD Group Attribute1 (PGM 213)
	3	ACD Group Attribute2 (PGM 214)

Related Features

Hardware

10.4 ACD Call Queuing Service

Description

If all of agents are busy or all of agents are not log-in, next ACD call will be queued. Max queue count can be changed by Administrator or Supervisor. If there are over the max queued count, next call will be released or listen overflow-announcement or routed destination by Overflow Service Rule.

- Overflow status will be changed automatically from Normal status when all agents are busy and also ACD queue is full. Administrator or Supervisor can make rule release, overflow announcement or forward.

When ACD call is queued, system gives Queuing announcement. Each group can serve max. 5 queuing announcements and can define how many announcements can be served [Queuing Announcement Service Step PGM 213 – Flex10]. And also each group can have the rule how to handle the queued call after finish the Queuing Announcement. In that case ACD call can be released or forwarded to registered destination.

- If [queued count display option] admin is set [PGM 214 Flex6], queued count will be displayed to Supervisor and Sub-Supervisor and all of agent keyset.
  And Administrator can make time interval for Queued call count display [PGM 214 Flex6].
- If **{ACD Supervisor Queued Call Answer}** feature code is saved at Supervisor's or Sub-Supervisor's flex button, this button color will be steady-on when there are queued ACD call. At that time, Supervisor or Sub-Supervisor can answer the first queued ACD call or can forward to registered destination.
- Administrator can make Queuing Announcement Scenario with [Repeat Announcements Count] and [Repeat Announcements Start Position] in PGM 213 Flex11, 12. After [Queuing Announcement Service Step], if [Repeat Announcements Count] is set. Queuing announcement will be restarted from [Repeat Announcements Start Position].

Operation

<u>*ACD Call Operation when All of agents are not ready*</u>

1. ACD call arrives and all the agents are not ready to receive the call.
2. ACD call will be queued and then 1st queuing announcement will be served.
3. After 1st queuing announcement timer expired, system checks there are next announcement step admin is set or not.
4. If there is more next announcement step, 2nd announcement will be served.
5. And after 2nd announcement, system checks next announcement usage.
6. If all of announcement is expired, system checks the forward option.
7. If ACD call release option is set, this call will be released. Or, if ACD call forward option is set, this call will be forwarded to registered destination.

Queuing Announcement Repeat Scenario.

1. If [Queuing Announcement Service Step] is set 5^{th} Announcement.
2. And If [Repeat Announcements Count] admin is set 3 times.
3. And if [Repeat Announcements Start Position] is set 3^{rd}. In that case;
4. ACD call arrives and all the agents are not ready to receive the call.
5. ACD call will be queued and then queuing announcements will be served from 1st to 5th.
6. After 5^{th} queued announcement service, those repeat announcements will be served 3 times.
   1^{st} announcement service => 2^{nd} announcement service => 3^{rd} announcement service => 4^{th} announcement service => 5th announcement service
   => 3^{rd} announcement service => 4^{th} announcement service => 5^{th} announcement service
   => 3^{rd} announcement service => 4^{th} announcement service => 5th announcement service.
   => 3^{rd} announcement service => 4^{th} announcement service => 5th announcement service.
   => release or forwarded to registered destination.

First Queued Call Answer by Supervisor or Sub-Supervisor

1. If there are queued ACD call,
2. Dial **{ACD Supervisor Queued Call Answer}** feature code. Or, press flex button registered as **{ACD Supervisor Queued Call Answer}** feature code.
3. Supervisor or Sub-Supervisor can check CLI of first queued and sub-menu.
4. Press 1 is Answer the call. Or, press 2 is Forward queued call to some of destination.
5. If dial some of Tel-Number and then press hold or '#' button, first queued call will be rerouted.

Conditions

1. Each Queuing Announcement playtime is max. 600 sec.
2. If there is not registered forward destination, after queuing announcement, ACD call will be released.
3. Supervisor, Sub-Supervisor and all of Agents can check and listen all of ACD Group Announcement with **{ACD Announce Play}** feature code.

Programming

Numbering Plan	1	Feature Numbering Plan (PGM 113)
Station Group	1	ACD Group Assignment (PGM 212)
	2	ACD Group Attribute1 (PGM 213)
	3	ACD Group Attribute2 (PGM 214)
	4	ACD Group Announcement (PGM 215)

Related Features

Hardware

10.5 CCR Service During ACD Announcement

Description

Administrator can set about CCR service during each Announcement service. In each ACD Group has 8 Announcement for CCR (5 Queuing Announcements, Night, Holiday, and Overflow Announcements).

Administrator can make rule CCR usage during each kinds of Announcement. If each kinds of announcement's Tone type is set as announcement and then CCR usage is set. During announcement service if some of number is detected, CCR rule in Announcement table will be operated.

The System can provide CCR Service during queuing announcement according to the CCR option.

A CCR Table defines a dialed digit (0–9, #, and *) to a designated route; each individual digit corresponds with a route:

- Station
- Station Group
- ACO Group
- Announcement Table
- Announcement Table And Drop
- System Speed Dial
- Conference Room
- Attendant Call
- VMIB Access
- Networking Number
- Internal Paging
- External Paging
- Internal/External All Paging
- Company Directory (New Prompts Needed)
- Record VM Greeting (New Prompts Needed)
- Direct VM Transfer
- Digits

In addition, the System will monitor digits for a system numbering plan(eg station number). If the User dials a Station number, Group Queuing Service is finished and a call is routed to the dialed destination.

Operation

CCR Operation During Queuing Announcement

1. ACD call arrives and all the agents are not ready to receive the call.
2. ACD call will be queued and then Queuing Announcement will be served.
3. During Announcement caller dial number.
4. Defined CCR table in Announcement table will be referenced.
5. Defined rule will be operated.

Conditions

1. SIP/ISDN Terminal does not support CCR feature.
2. In case of Agent Answer Announcement, CCR option is not applied.

Programming

Station Group	1	ACD Group Assignment (PGM 212)
	2	ACD Group Attribute1 (PGM 213)
	3	ACD Group Attribute2 (PGM 214)
	4	ACD Group Announcement (PGM 215)

Related Features

Hardware

10.6 ACD Agent State

Description

When administrator assigns the agents, all of agents are log-out state. If agents want to receive the ACD call, agents have to go to log-in state. Agents can go to log-in state with [ACD Agent Login/Logout] feature code. And also log-in agents can change their state from Ready state to DND state and Work state. So each agent can have 4 types of state (Log-Out / Log-In Ready / Log-In DND / Log-In Work).

1. **Log-Out State:** Agent is only assigned for ACD member. Agents cannot receive ACD call.
   If [Call Restriction When Agent Logout] admin is set as All Call Restriction, agents cannot make call.
2. **Log-In Ready State:** Agent is log-in and ready to receive the ACD call.
3. **Log-In Work State:** Agent is log-in state but agent can work without any ACD call during Work-Mode Expired Timer. Work-Mode Expired Timer can be changed at admin [PGM 214 – Flex3]. After this timer agent's state is changed to ready state automatically.
   - Agents can set own state to Work state with **{ACD Agent Work Mode}** feature code.
   - If agent dial **{ACD Agent Work Mode}** feature code or press flex button registered as **{ACD Agent Work Mode}** feature code, agent state goes to Work State.
   - Also agents can set Auto Work State Changing Function with **{ACD Agent Auto Work}** feature code. If Auto Work State Changing Function is set, after conversation or after ringing or after outgoing call, agent state automatically changes to Work State. Changing condition can be set at Agent Auto Work Mode admin [PGM 214 Flex4].
4. **Log-In DND State:** Agent is log-in but agent is set as DND. DND agent does not receive ACD call. DND is not changed automatically, agent has to change own state to Ready state or Work state. Supervisor or Sub-Supervisor can monitor DND agent and then can change their DND state to Ready or Work State.
   - Agent can set own state to DND State with **{ACD Agent DND}** feature code.
   - If agent dial **{ACD Agent DND}** feature code or press flex button registered as **{ACD Agent DND}** feature code, agent state goes to DND State.

If [Agent No-Answer Service] admin is set as DND State Change [PGM 213 – Flex15], when agent does not answer the ACD call, this agent state will be changed to DND state. Administrator can also set no-answer ACD call to forward to registered destination [PGM 213 – Flex16].

Operation

Agent Work State

1. Dial **{ACD Agent Work Mode}** feature code. Or, press flex button registered as **{ACD Agent Work Mode}** feature code.
2. If agent is log-in, agent goes to Work mode state.
3. After [Agent Work-Mode Expired Time], agent goes to Ready State.

Agent Auto-Work Mode

1. Dial **{ACD Agent Auto Work}** feature code. Or, press flex button registered as **{ACD Agent Auto Work}** feature code.
2. Agent is set as Auto-Work Mode.

3. After Call or Ring, agent state is automatically changed to Work State, and then when Work Mode Expired timer is over, agent state will be back to Ready State.

Agent DND State

1. Dial **{ACD Agent DND}** feature code. Or, press flex button registered as **{ACD Agent DND}** feature code.
2. If agent is log-in, agent goes to DND state.

ACD Call Operation when agents no-answer

1. ACD call arrives and agent does not answer the call.
2. When No-Answer timer is expired, system checks the Agent No-Answer Service option.
3. If Option is Forward to Forward destination, ACD call is rerouted to forward destination [PGM 214 – Flex6]. Or, if Option is Agent DND State Change. No-answer agent state will be change to DND, and ACD call will be released. Or, if Option is Agent DND State Change and Forward. No-answer agent state will be change to DND, and ACD call will be rerouted to forward destination.

Programming

Numbering Plan	1	Feature Numbering Plan (PGM 113)
Station Group	1	ACD Group Assignment (PGM 212)
	2	ACD Group Attribute1 (PGM 213)
	3	ACD Group Attribute2 (PGM 214)

Related Features

Hardware

10.7 ACD Agent Log-In / Log-Out Default Setting

Description

When agents go to log-in, some of default functions can be set according to admin values. And these kinds of values can be defined by administrator.

1. Administrator can define Auto-Answer mode when agent goes to log-in [PGM 214 – Flex10]. If [Auto Answer Use When Agent Login] option is set, when agent goes to log-in state, this agent's answer mode is set as Auto-Answer. So when this agent receives the ACD call, after first ring, agent keyset automatically answer the ACD call. Agent cans turn-on or turn-off Auto-Answer function with **{ACD Agent Auto Answer}** feature code.
2. Administrator can define Auto-Work mode when agent goes to log-in [PGM 214 – Flex11]. If [Auto Work-Mode Use When Agent Login] option is set, when agent goes to log-in state, this agent's auto work-mode is set.
   Agent cans turn-on or turn-off Auto-Answer function with **{ACD Agent Auto Work}** feature code.
3. Administrator can define log-in agent's headset mode when agent goes to log-in by [PGM 214 – Flex12] which can be Headset, Handset, Ear-Mic or Bluetooth.
4. Administrator can define log-out agent's headset mode when agent goes to log-out by [PGM 214 – Flex13] which can be Headset, Handset, Ear-Mic or Bluetooth.
5. Administrator can define log-out agent's call restriction by [PGM 214 – Flex14]. Log-in agent's call also can be restricted by administration.

Operation

Conditions

Programming

Numbering Plan	1	Feature Numbering Plan (PGM 116)
Station Group	1	ACD Group Assignment (PGM 212)
	2	ACD Group Attribute1 (PGM 213)
	3	ACD Group Attribute2 (PGM 214)

Related Features

Hardware

10.8 ACD Call Indication

Description

Agents can recognize ACD call or the other common call with **{ACD Call Indication}** and **{NON ACD Call Indication}** feature codes. Agent assign ACD call indication and Non-ACD call Indication in a flex button with **{ACD Call Indication}** feature code and **{NON ACD Call Indication}** feature code. Then if an ACD call is routed to an agent, (ACD Call Indication) flex button will be turned on. And when a non-ACD call is routed to an agent, (Non ACD Call Indication) flex button will be turned on.

Operation

*Agent Assign **{ACD Call Indication}** feature code*

1. Press **[Trans/PGM]** button.
2. Choose Flex button Press flex button and then Dial 1 for Tel-Number.
3. Dial **{ACD Call Indication}** feature code.

*Agent Assign **{NON-ACD Call Indication}** feature code*

1. Press **[Trans/PGM]** button.
2. Choose Flex button Press flex button and then Dial 1 for Tel-Number.
3. Dial **{NON-ACD Call Indication}** feature code

Programming

Numbering Plan	1	Feature Numbering Plan (PGM 113)
	2	ACD Numbering Plan (PGM 118)
Station Port	1	Station Port Attribute (PGM 124)
Station Group	1	ACD Group Assignment (PGM 212)
	2	ACD Group Attribute1 (PGM 213)
	3	ACD Group Attribute2 (PGM 214)

Related Features

Hardware

10.9 ACD Group Supervisor Functions

Description

ACD group can have one Supervisor and three Sub-Supervisors.

Supervisor can change group service status from Normal Status to Group forward Status and Overflow, Night, Holiday Status. Also Supervisor can register each cases forward destination.

Supervisor and Sub-Supervisor can check queued ACD call count and can answer or reroute first queued ACD call with **{ACD Supervisor Queued Call Answer}** feature code. Also Supervisor and Sub-Supervisor can monitor Agent's state and can overhear agent's conversation with caller. And also Supervisor and Sub-Supervisor can check ACD group call traffic.

1. **Supervisor Group Management.**
   - Supervisor can change Group Service Status with feature code.
     - **Group Forward Status**: Supervisor can change from other status to Group Call Forward Status with **{ACD Supervisor Group Call Forward}** feature code.
     - **Group Night Status:** Supervisor can change from other status to Group Night Status with **{ACD Supervisor Group Night Mode}** feature code.
     - **Group Holiday Status:** Supervisor can change from other status to Group Holiday Status with **{ACD Supervisor Group Holiday Mode}** feature code.
   - Supervisor can define how to handle ACD call when not Normal Group Status. Supervisor can access **ACD Group Management web page in Station Program of iPECS-MG Web-Admin.**
     - Supervisor can assign Group Forward Destination.
     - Supervisor can make rule how to handle ACD call when group status is Night Status.
       And Supervisor can assign Night Forward Destination.
     - Supervisor can make rule how to handle ACD call when group status is Holiday Status. And Supervisor can assign Holiday Forward Destination.
     - Supervisor can change max. Queuing Count and also can change Queuing Service Announcement Step. Supervisor also can make rule how to handle ACD call when all of queuing announcement service is over. And After Queuing Forward Destination can be changed.

2. **Supervisor and Sub-Supervisor Answer the Queued ACD call.**
Supervisor and Sub-Supervisor can find queued ACD call count on the LCD. And if **{ACD Supervisor Queued Call Answer}** feature code is saved at flex button, this flex button's color will be changed to steady-on.

− First Queued ACD call Answer by Supervisor

- When there are queued ACD call, if Supervisor or Sub-Supervisor press **{ACD Supervisor Queued Call Answer}** feature codes, Total queued ACD call count and sub-menu will be displayed on LCD.
- If supervisor choose Check First Queued Call menu, CLI of first Queued call will be display. At that time, if supervisor choose Answer option, queued ACD call will route to Supervisor.

− First Queued ACD call Reroute

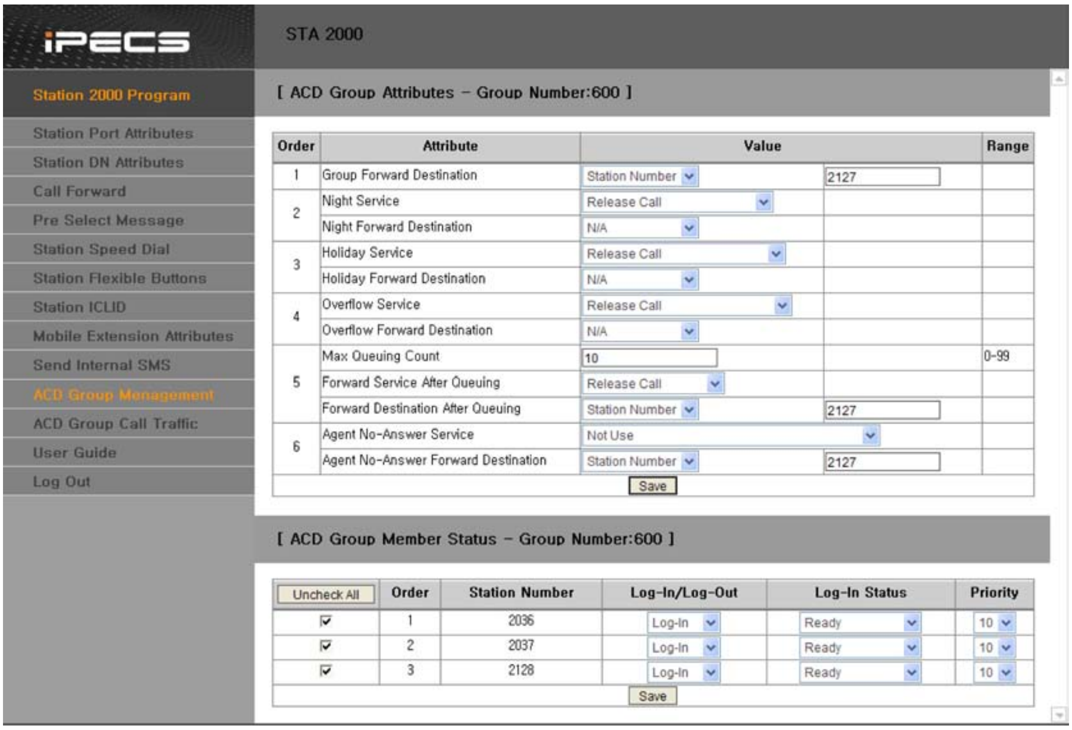
- When there are queued ACD call, if Supervisor or Sub-Supervisor press **{ACD Supervisor Queued Call Answer}** feature codes, Total queued ACD call count and sub-menu will be displayed on LCD.
- If supervisor choose Check First Queued Call menu, CLI of first Queued call will be display. At that time, if supervisor choose Reroute option, current registered Forward Destination after Queuing Announcement will be displayed.
- If press **[HOLD]** or "#" button, queued ACD call will be rerouted to registered destination. Or, if dial other Tel-Number and then press **[HOLD]** or "#" button, queued ACD call will be rerouted to new dialed Tel-Number.

3. **Supervisor and Sub-Supervisor Monitoring Agent.**
Supervisor and Sub-Supervisor can check and monitor agent's status. And also Supervisor and Sub-Supervisor can overhear agent's conversation.

− Agent Status Monitor by Supervisor on **Digital Phone**

- Supervisor can check agent's state with **{ACD Supervisor Agent State Check}** feature code. If supervisor dial or press flex button registered as **{ACD Supervisor Agent State Check}** feature code, first agent state will be displayed on LCD.
- Agent Log-In / Log-Out information is displayed and if agent is Log-In state, Ready or DND or Work sub-state also displayed. At that time, Supervisor can choose sub-option (Forced-Log In/Out, Forced Ready, Forced DND, and Forced Work).
- Next agent's status information can be changed when Supervisor press Volume-Up/Down button or Up/Down in Navigation Key.

− Agent Status Monitor on **ACD Group Management web page in Station Program in iPECS-MG Web-Admin.** And in this web-page Supervisor or Sub-Supervisor can check an change All of Agent's Log-In / Out, Ready, DND and Work State. Also agent's priority can be changed in this web-page.



− Silent-Monitor of Agent's Conversation
Supervisor can monitor agent's conversation with **{ACD Supervisor Silent Monitor}** feature code. If Supervisor dial **{ACD Supervisor Silent Monitor}** feature code and dial agent number. Supervisor can overheard this agent's conversation.

4. **Supervisor and Sub-Supervisor ACD Group Call Traffic Check**
   Supervisor and Sub-Supervisor can check ACD Group Call or Agent Call Traffic with **{ACD Supervisor Traffic Check}** feature code.

Operation

Group Forward / Night / Holiday Status Change by Supervisor

1. Dial **{ACD Supervisor Group Call Forward}**, **{ACD Supervisor Group Holiday Mode} or {ACD Supervisor Group Night Mode}** feature code. Or, Press flex button registered as **{ACD Supervisor Group Call Forward}**, **{ACD Supervisor Group Holiday Mode}** or **{ACD Supervisor Group Night Mode}** feature code.
2. Group Status will be changed.

ACD Group Management or ACD Agent State Check With Web-Admin

1. Connect Web-Admin Page of iPECS-MG System.
2. Choose Station Program Menu.
3. Enter Supervisor Number and Password.
4. **ACD Group Management** Menu will be displayed on the Left Menu List.
5. Supervisor can change each Group Status rules and Destination.
6. Supervisor and Sub-Supervisor can check Agent Log-In/Out, Ready, DND, Work State. And also Agent's Priority can be displayed.

7. Supervisor and Sub-Supervisor can change Agent's State and Priority.

<u>*ACD Group Call or Agent Traffic Check with Web-Admin*</u>

1. Connect Web-Admin Page of iPECS-MG System.
2. Choose Station Program Menu.
3. Enter Supervisor Number and Password.
4. **ACD Group Traffic** Menu will be displayed on the Left Menu List.
5. Supervisor and Sub-Supervisor can check Group Call Traffic and Agent Traffic Data.
6. Supervisor and Sub-Supervisor can Clear Group Call Traffic and Agent Traffic Data.

<u>*First Queued Call Answer or Reroute by Supervisor*</u>

1. If there are queued calls, **{ACD Supervisor Queued Call Answer}** feature code is saved at flex button.
2. Flex button's color will be changed to steady-on.
3. Dial **{ACD Supervisor Queued Call Answer}** feature code. Or, press flex button registered as **{ACD S Supervisor Queued Call Answer}** feature code.
4. Choose sub-menu then First Queued Call CLI will be display
5. Press "1" then queued call will be routed to Supervisor. Or, press "2" and Dial Tel-Number.
6. Press "#" or **[HOLD]** button, then queued call will be routed to Dialed tel-Number.

<u>*Silent Monitor by Supervisor*</u>

1. Dial **{ACD Supervisor Silent Monitor}** feature code. Or, press flex button registered as **{ACD Supervisor Silent Monitor}** feature code.
2. Dial desired Agent number.
3. Agent's conversation will be overhead.

Conditions

1. If **{ACD Supervisor Group Call Forward}**, **{ACD Supervisor Group Holiday Mode}** or **{ACD Supervisor Group Night Mode}** feature code is saved at a flex button, when group status is some Status, this flex button's color will be changed to steady-on.
2. **{ACD Supervisor Group Call Forward}**, **{ACD Supervisor Group Holiday Mode}** or **{ACD Supervisor Group Night Mode}** feature code works in toggle mode.
3. If **{Auto Record Service}** admin is set at agent, Supervisor and Sub-Supervisor cannot overhear agent's conversation.

Programming

Numbering Plan	1	Feature Numbering Plan (PGM 116)
Station Group	1	ACD Group Assignment (PGM 212)
	2	ACD Group Attribute1 (PGM 213)
	3	ACD Group Attribute2 (PGM 214)

Related Features

Hardware

10.10 ACD Group Call Traffic

Description

ACD Group Call Traffic and Agent Call Traffic are automatically saved in the System.

Supervisor and Sub-Supervisor can check these kinds of Traffic data on their Digital-Phone. And also Supervisor and Sub-Supervisor can access **ACD Group Call Traffic web page in Station Program of Web-Admin** and then check and clear Traffic data.

ACD Group Call Traffic can be printed periodically as Information-Data format. And also Supervisor and Sub-Supervisor can print these kinds of data manually.

1. **Supervisor and Sub-Supervisor can check ACD Group Call or Agent Call Traffic with {ACD Supervisor Traffic Check} feature code.**

− **ACD Group Call Traffic**

All of ACD call data will be saved and Traffic Data format is just like bellow lists.

- Total Calls Count
- Unanswered Call Count
- Average Call Time 00:00 (minute: second)
- Average Ring Time 00:00 (minute: second)
- Busy Count and Time 00:00:00 (hour: minute: second)
- Number of calls count in Current Queue
- Average Queued Time 00:00 (minute: second) and Longest Queued Time 00:00 (minute: second)

− Supervisor can check all of data with Volume Up/Down Key or Up / Down in Navigation Key.
− During checking the Group Call Traffic information, Supervisor can clear all of Group Call Traffic with **[SPEED]** button. If Supervisor uses 3-soft keyset, Delete menu will be displayed at 3-soft menu.
− Average Call time means average conversation time of all of agent ACD call.
− Busy count means how many times all of agents are busy. And Busy time means total accrued Times of Agent's busy state.
− Information about Queued data will be always computed when there are queued ACD calls in Queue.

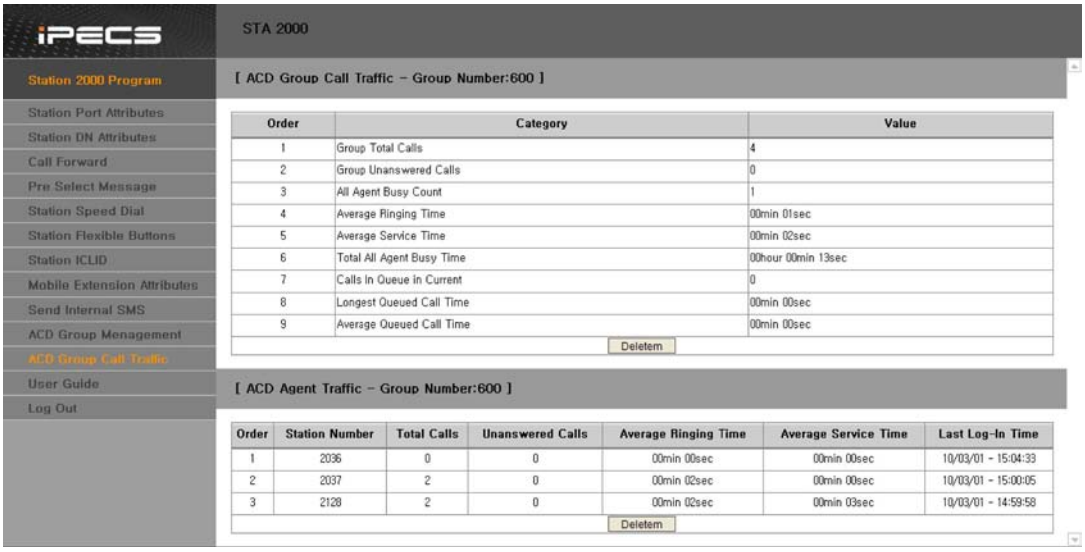
− **Agent Call Traffic**

Agent's ACD call data will be saved and Traffic data format is just like bellow lists.

- Total Calls Count
- Unanswered Call Count
- Average Call Time 00:00 (minute: second)
- Average Ring Time 00:00 (minute: second)
- Last Log-In Time 00:00:00 (hour: minute: second)

− When Supervisor enter Agent Call Traffic feature, first agent data will be displayed. And Supervisor can check all of agent data with Volume-Up/Down button or Up/Down in Navigation Key.
− During checking the Group Call Traffic information, Supervisor can clear all of Group Call Traffic with **[SPEED]** button. If Supervisor uses 3-soft keyset, Delete menu will be displayed at 3-soft menu.
− Average Call time means average conversation time.
− In case of Last Log-In time, if check at **ACD Group Call Traffic web page in Station Program of Web-Admin**, Log-In date also can be checked.

2. **ACD Group Call Traffic web page in Station Program**



Order	Category	Value
1	Group Total Calls	4
2	Group Unanswered Calls	0
3	All Agent Busy Count	1
4	Average Ringing Time	00min 01sec
5	Average Service Time	00min 02sec
6	Total All Agent Busy Time	00hour 00min 13sec
7	Calls In Queue in Current	0
8	Longest Queued Call Time	00min 00sec
9	Average Queued Call Time	00min 00sec

Order	Station Number	Total Calls	Unanswered Calls	Average Ringing Time	Average Service Time	Last Log-In Time
1	2036	0	0	00min 00sec	00min 00sec	10/03/01 - 15:04:33
2	2037	2	0	00min 02sec	00min 00sec	10/03/01 - 15:00:05
3	2128	2	0	00min 02sec	00min 03sec	10/03/01 - 14:59:58

3. **Traffic data Print to Information Print Port**

If Traffic information data print admin is set [PGM 214 – Flex18], ACD Group Call traffic data will be printed at Information-Print Port. And also [Information Data Print Interval] admin can set periodic print interval time.

Periodic Print ACD Group Call Traffic Format

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

Field(s)	DESCRIPTION
~(tilt)	Means start of ACD statistics and is always located at first column
=(equal)	Delimiter between each meaningful data
1	ACD Group number
2	Total call counter
3	Unanswered call counter
4	All busy counter
5	Average ringing time (ex., 96=1 min 36 sec)
6	Average call service time (ex., 25=0 min 25 sec)
7	Total busy time (ex., 64=1 min 04 sec)
8	Number of current queued calls
9	Longest queued time
0	Average queued time
lf	Line Feed (0x0A)
cr	Carriage Return (0x0D)

Supervisor or Sub-Supervisor can print Group Call Traffic data at Information-Print port manually.

1. Press **{ACD Supervisor Traffic Check}** feature code.
2. Choose Group Traffic data.
3. Press **[HOLD/SAVE]** button for printing data to Information-Print Port.

Supervisor or Sub-Supervisor can print Agent Call Traffic data at Information-Print port.

1. Press **{ACD Supervisor Traffic Check}** feature code.
2. Choose Agent Traffic data.
3. Find desired Agent number with "*" or "#" button, Left/Right in Navigation Key
4. Press **[HOLD/SAVE]** button for printing data to Information-Print Port.

Agent Call Traffic Format

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

Field(s)	Description
~(tilt)	Means start of ACD statistics and is always located at first column
=(equal)	Delimiter between each meaningful data
1	Each Agent number
2	Total call counter
3	Unanswered call counter
4	Average ringing time (ex., 96=1 min 36 sec)
5	Average service time (ex., 96=1 min 36 sec)
lf	Line Feed (0x0A)
cr	Carriage Return (0x0D)

Conditions

1. Information Print Port can be changed at Serial Selection Admin [PGM 231].

Programming

Numbering Plan	1	Feature Numbering Plan (PGM 116)
Station Group	1	ACD Group Assignment (PGM 212)
	2	ACD Group Attribute1 (PGM 213)
	3	ACD Group Attribute2 (PGM 214)

Related Features

Hardware

10.11 ACD Group Supervisor Feature

Description

Supervisor and Sub-Supervisor can check and monitor agent's status. And Supervisor and Sub-Supervisor can overhear agent's conversation.

Supervisor and Sub-Supervisor also can record agent's conversation during monitoring with two-way recording feature code or Record menu on 3 Soft-Button. When supervisor try to record agent's conversation, S-Monitor alert tone will be served to agent and called-party. And then agent's conversation will be recorded.

Supervisor and Sub-Supervisor can make conference with agent's conversation. And also during conference supervisor can release one of member forcibly.

Operation

Agent's call Recording:

1 Dial **{ACD Supervisor Silent Monitor}** feature code.
 Or,
1 Press flex button registered as **{ACD Supervisor Silent Monitor}** feature code.
2 Dial desired Agent number
3 Press Two-way recording feature flex button during conversation monitoring.
 Or,
1 Press Record menu on 3 Soft-Button.
2 If supervisor want to stop recording,
3 Press again Two-way recording feature flex button
 Or,
1 Press Record-Stop menu on 3 Soft-Button

Agent's call Conference:

1 Dial **{ACD Supervisor Silent Monitor}** feature code.
 Or,
1 Press flex button registered as **{ACD Supervisor Silent Monitor}** feature code.
2 Dial desired Agent number
3 Press Conf menu on 3-soft button.

Conditions

1 Supervisor has to have Two-way recording authority.
2 If supervisor make conference during recording, recording feature will be stopped.
3 If agent's conversation is already recorded, supervisor cannot record conversation.
4 While agent's conversation is recorded, even if agent's conversation is over, recording feature will not be stopped. Supervisor has to stop recording feature.

Programming

Station Group	1. Main Supervisor (PGM212)
	2. Sub Supervisor (PGM212)

Related Features

Hardware

11. HOTEL FEATURE

iPECS-MG system supports hospitality features like check-in/check-out, management of guest room information (Guest name, Guest Class of Service, Maid Status, Mini-bar, etc.), convenient features for hotel guests (Wake-up, DND, Message Wait, Bath Alarm, etc.), and efficient hotel Operation and services through interworking with PMS (Property Management System) system.

In iPECS-MG system, hotel features and office features are available in one software package. Basically, the hotel features are available only for hotel stations like guest station, front desk, and service station. But other normal call features like call transfer, call forward, camp-on, message wait and etc are common to both hotel and office stations. It is also possible to allow or deny calls between hotel and office stations. This configuration would be very useful in case there's a client who wants to have an office solution and hotel solution at the same time with only one physical telephony system.

11.1 HOTEL SERVICE TYPE

Each extension of iPECS-MG is assigned its own hotel service type, and typical features and services are provided depending on the allocated hotel service type.

There are 4 hotel service types: Office Station, Guest Station, Front Desk, and Service Station.

In order to use hotel features, stations must be programmed to have hotel service types like guest station, front desk and service station.

One station can have only one hotel service type. It is impossible to have a station that is a front desk and also a service station. For another example, office station cannot be a front desk.

11.1.1 Office Station

Description

Office station is a terminal that supports only office features, but does not support hotel features.

It is not possible to use hospitality features like check-in, check-out, maid status and etc from an office station. But all the basic telephony services are available in an office station.

Conditions

1. Office station is the default hotel service type in iPECS-MG system.
2. The calls from office stations to hotel stations (guest station, front desk, and service station) can be allowed or denied.

Programming

Hotel Data 1 Hotel Service Type (PGM 502 – Index1)

Related Features

Hardware

11.1.2 Guest Station

Description

Guest station is a terminal that is in a hotel guest room and used by the hotel guest.

From guest station, it is possible to use hotel-specific features like room monitor, maid status and etc. In addition, all the basic telephony services like DN (Directory Number) features, system features (Transfer, Forward, Speed Dial, VM features), CO features (DID, DISA) are available.

Conditions

1. Hotel guests can check in to a room where guest station is installed.
2. During check-in procedure, it is possible to set basic information for a guest station such as guest name, class of service, PMS (Property Management System) group, and LCD/prompt language. After the guest check-in, this information can be changed as desired by using the same check-in Programming again.
3. By default, the name of guest station is “ROOM” in English LCD mode. This can be changed during check-in or by station name Programming. And the programmed name is restored to “ROOM” after guest’s check-out.
4. The calls from guest stations to office stations can be allowed or denied.

Programming

Hotel Data 1 Hotel Service Type (PGM 502 – Index1)

Related Features

Hardware

11.1.3 Front Desk

Description

Front desk is a terminal that can be used in hotel front desk.

It provides more hotel features than the guest station. Features like check-in, check-out, guest info setting and display, room swapping and etc are supported for front desks. Front desk can provide hotel services to the hotel guests and service stations.

Operation

Front Desk

*To access a front desk menu, press the **[PGM]** button and dial #.*

The following list shows the program menus available in hotel front desk.

11.1.3-1 Hotel Program Menu List

Program Menu	Menu	Sub Menu	Remarks
[#] Hotel Feature	[1] Hotel Room Check In	[1] Check-In: VIP Room Setting	
		[2] Check-In: Room LCD Language	
		[3] Check-In: Room Voice Prompt	
		[4] Check-In: Day COS	
		[5] Check-In: Night COS	
		[6] Check-In: Timed COS	
		[7] Check-In: Digit Conversion Table	
		[8] Check-In: Guest Name Setting	
		[9] Check-In: PMS Group Setting	
		[0] Check-In: Check-Out	
		Schedule	
		[*] Check-In: Confirm	
	[2] Hotel Room Check Out	[1] Check-Out: Payment Method	
		[2] Check-Out: Confirm	
	[3] Hotel Call Block	[1] Hotel Room Cut-Off	
		[2] Hotel Room ICM Call	
		[3] One-time CO Call Use	

Table 11.1.3-1 Hotel Program Menu List

Program Menu	Menu	Sub Menu	Remarks
[#] Hotel Feature	[4] Hotel Room Setting	[1] Room Wake Up Register	
		[2] Room Wake Up Cancel	
		[3] Room DND Setting	
		[4] Room Message Wait	
		[5] Room Bath Alarm Enable	
		[6] Room Bath Alarm Disable	
		[7] Room Author Code	
		[8] Room Prepaid Money	
	[5] Hotel Room-Swap Setting		
	[6] Hotel Room Maid Status		
	[7] Hotel Room Charge/Status	[1] Room Charge Print	
		[2] Room Status Print	
		[3] Delete SMDR (Service Station)	
	[8] Hotel Room Rate	[1] Room Rate Register	
		[2] Room Rate Assign	
		[3] Room Part Time Fee	
		[4] Room Bar/Mini-Bar Charge	
		[5] Additional Tax Fields	
	[9] Hotel Room Call Rate	[1] Room Call Rate Register	
		[2] Room Call Rate Assign	
	[0] Hotel Misc Program	[1] Hotel Name	
		[2] Set Call Forward	

NOTE
Currency Unit depends on Admin **Programming**. (PGM 232)

Conditions

1. Front desk program menus are available only from front desks.
2. There can be multiple front desks and there is no limit on the number of front desks.
3. Calls from front desks to office stations can be allowed or denied.
4. The attendant can be programmed to be front desk. But this is optional to the users.

Programming

Hotel Data 1 Hotel Service Type (PGM 502 – Index1)

Related Features

Hardware

11.1.4 Service Station

Description

Service station is a terminal for some shops or restaurants in hotel.

It's not possible to check in to a service station because it is not a guest station. But bar charge can be entered in service station. In this case, bar items should be registered first as bar product's name.

Operation

<u>*To call Hotel Service Station,*</u>

1. Lift handset or press **[SPEAKER]** button.
2. Dial the service station number as a normal intercom call.

Conditions

1. It is always possible to call a hotel guest station from a service station. And regardless of ICM call barring setting and room-to-room call group, hotel guests also can call a service station.
2. By default, the service station name is set as "SVC STATION" in English LCD mode. But it can be changed by station name Programming. If there's a registered service station name, the name will be displayed on LCD.
3. It is impossible to set a station both as room and as service station at the same time.
4. The calls from service stations to office stations can be allowed or denied.
5. At hotel service station, a guest can make an outgoing CO call with password by using 'Walking COS' feature or by using One-Time-CO-Call Enable Feature.

Programming

Hotel Data	1	Hotel Service Type (PGM 502 – Index1)

Related Features

Hardware

11.2 Multiple Front Desks

Description

Multiple front desks can be activated via Programming. A station becomes a front desk if it is assigned a hotel service type as front desk.

Operation

Front desk features and program menu are available from front desk.

Conditions

1. There is no limit on the number of front desks.
2. If there are multiple front desks, they can be put into one station hunt group so that the multiple front desks can receive calls from guest stations or service stations.
3. If the attendants are programmed to be front desks, the attendants will be able to cover office stations and hotel stations. In this case, calls between office stations and front desks should be enabled.
4. Simultaneous front desk Programming is possible in iPECS-MG system. Only the latest update will be effective if multiple front desks program the same settings.

Programming

Hotel Data	1	Hotel Service Type (PGM 502 – Index1)

Related Features

Hardware

11.3 Check In

Description

It allocates a guest room to each guest, changes the maid status and registers basic information of the guest(s) individually or entirely. This feature is available from PMS (Property Management System), and Front Desk.

Operation

Front Desk

To check in an empty room

1. Dial the **[PGM]** button + "#1".
2. Press a phone number or a phone number range of desired room(s).
   Press * between phone numbers when entering a phone number range.
3. Press the **[SAVE]** button.
4. Designate a desired value depending on the following menu number.
   1 : CHECK-IN: VIP SETTING
   2 : CHECK-IN: LCD LANGUAGE
   3 : CHECK-IN: VOICE PROMPT
   4 : CHECK-IN: DAY COS
   5 : CHECK-IN: NIGHT COS
   6 : CHECK-IN: TIMED COS
   7 : CHECK-IN: DGT CONV TBL
   8 : CHECK-IN: GUEST NAME
   9 : CHECK-IN: PMS GROUP
   0 : CHECK-IN: CHECK-OUT SCHEDULE
   * : CHECK-IN: CONFIRM
5. Press a desired value of each sub menu and press the **[SAVE]** button.
6. When completing setting of all values, dial * to finish check-in.
7. Select Check-In Confirmation and press the **[SAVE]** button. (0: Confirm, 1: Cancel)
8. When check-in is successful, you may hear service set tone and check a result through phone LCD.
9. When check-in fails, you may hear an error tone and check the result through the phone LCD.

Conditions

1. The following station data will be initialized when it is checked in.
   : Wake-up, Call Forward, DND, Message Wait and etc.
2. When check-in is registered from PMS, guest grade, language ID, COS, name, PMS group number, and expected check out time must be entered.
3. If check-in to a new room is confirmed by the front desk without Programming any data, system processes the check-in with default values.

4. From the front desk, it is possible to change guest room's data (VIP, LCD/Prompt language, class of service, guest name, PMS group, and check-out schedule) by this feature. If one of the above data is changed for already checked-in rooms, only those changed guest room's data are updated.
5. During check-in/change guest status process, room status is printed through Off-line SMDR port (PGM 231-Index2) if Check In/Out Print (PGM 501-Index8) is set to ON. The print-out format is as follows,

Chk-In

ROOM	CHECK-IN	GUEST-NAME	COS	ICM	GRP	WAKE-UP	AUTH	PRE-PAID	CALL-RATE	CHARGE
100	12/29-18	………	10101	NO	00000	..:..	NO	0	.	0

Charged

ROOM	CHECK-IN	GUEST-NAME	COS	ICM	GRP	WAKE-UP	AUTH	PRE-PAID	CALL-RATE	CHARGE
100	12/29-18	C.H.Lee	10101	YES	00001	06:00	Yes	20000	1	-20000

6. When a guest checks in to a room, SMDR and BAR records of the room are cleared.
7. During check-in, system disables room cut off. So, the guest can make a CO line call after check-in.
8. If a station is registered as service station, front desk will receive single error tone when checking in to the service station.
9. There are virtual DNs (directory numbers) that don't have a physical phone in database default status. Let' assume that a user tries to check in to a room that assigned a virtual DN. In this case, LCD and prompt language of the room will be changed only when the virtual directory number is the prime DN of the room. Prime DN is the number displayed on LCD when a station is in idle status.
10. The information of guest room will be protected.
11. The information print of guest room can be blocked by "Check In/Out On-line Print" (PGM 501-Index8)

Programming

Hotel Data

1 Check In/Out On-line Print (PGM 501 - Index8)

2 Hotel Service Type (PGM 502 – Index1)

3 Check-In Status (PGM 502 – Index2)

Related Features

Hardware

11.4 Check Out

Description

It registers check-out of hotel guests individually or entirely from PMS (Property Management System) and Front Desk.

Operation

Front Desk

To check out a room

1. Dial the **[PGM]** button + “#2”.
2. Press a phone number or a phone number range of desired room(s).
   Press * between phone numbers when entering a phone number range.
3. Press the **[SAVE]** button.
4. Designate a desired value depending on the following menu number.
   1: CHECK-OUT: PAY METHOD
   2: CHECK-OUT: CONFIRM
5. Dial 1 to select payment method and then dial one digit for payment index (0 ~ 9).
6. Press the **[SAVE]** button.
7. Dial 2 to select “Check Out Confirm” menu and then dial one digit for confirmation. (0: Confirm, 1: Cancel)
8. Press the **[SAVE]** button.
9. When check-out is successful, you may hear a service set tone and check the result through the phone LCD.
10. When check-out fails, you may hear an error tone and check the result through the phone LCD.

Conditions

1. Upon check-out, guest name is deleted and it is restored to be “ROOM”. Language ID and the COS are changed into values programmed in hotel general information (PGM 500).
2. When check-out is performed, maid status of the room is changed to "DIRTY (To be cleaned)". And the followings are automatically cleared or set:

   Fields that are cleared:

   Absence message and Message Wait / Guest Name / DND / Wake-Up Time /
   Call Forward Status / Authorization Code / Maid Status / Station COS /
   Prepaid Money / Voice mail

   Fields that are set:

   ICM disable / LCD language (PGM 500) / Prompt language (PGM 500) /
   Room to room Call Group / Room Cut-Off (PGM 502) /
   Call charge rate (PGM 504)

3. During check-out process, room total charge is printed through Off-line SMDR port (PGM 231-Index2) if Check In/Out Print (PGM 501-Index8)is set to ON and printed format is as follows. At this time, if the hotel name is registered, it is printed in place of the title "HOTEL NAME". Otherwise, this row is not printed. And if you select method of payment, it is printed. Otherwise, that row is not printed.
4. The Front Desk can select the method of payment among registered ones according to the guest's payment methods.

HOTEL NAME

TOTAL CHARGE IN ROOM 100 (Guest Name: …………)

Check – In : 10/12/26 -18

Check – Out : 10/12/31-11 (5 days)

Start-Time	CO	Duration	Dialed-No.	Count	Call-Cost	Remark
12/27-13:00	01	00:00:32	O00182343507851	0	0	Unanswered
12/27-13:30	01	00:01:23	O00182343507951	3	1500	
12/28-21:22	02	00:01:18	O00182343507953	31	15500	

Charge-Time	Charged-STA	Item	Bar-Cost	Tax
12/28-21:32	COFFEE-SHOP	COFFEE	5000	100
12/29-10:10	FRONT-DESK	COKE	3000	30

Item	Charge	Tax(rate)	Sum
(1) ROOM CHARGE: (rate 02: GOLD)	300000	30000(10.00%)	330000
(2) CALL CHARGE:	17000	170(10.00%)	17170
(3) BAR CHARGE:	8000	130	8130
(4) PRE-PAID			-20000

Method of Payment: VISA

TOTAL : 335300 WON

Note

Currency Unit/SMDR Fraction depends on PGM 232.

5. Check-out is not available when guest room is busy status.
6. The information print of guest room can be blocked by —Check In/Out On-line Print (PGM 501-Index8).

Programming

Hotel Data

1 Check In/Out On-line Print (PGM 501 - Index8)
2 Hotel Service Type (PGM 502 – Index1)
3 Check-In Statuses (PGM 502 – Index2)

Related Features

Hardware

11.5 Call Barring

11.5.1 CO Call Barring (Room Cut)

Description

The use of CO lines from guest stations can be allowed or denied from Front Desk or PMS.

Operation

Front Desk

To register/cancel room cut information:

1. Dial the **[PGM]** button + “#31”.
2. Dial a phone number or a phone number range.
3. Press * between phone numbers when entering a phone number range.
4. Press the **[SAVE]** button.
5. Designate whether to use trunk call. (0: Allowed, 1: Deny)
6. Press the **[SAVE]** button.
7. If it is successful, you may hear service set tone and check a result through phone LCD.
8. Otherwise, you may hear error tone.

PROCEDURE

[3] HOTEL CALL BLOCK [4] HOTEL ROOM SETTING	Press the **[PGM]** button and dial ‘#3’.
[1] HOTEL ROOM CUT-OFF [2] HOTEL ROOM ICM CALL	Dial ‘1’
HOTEL ROOM CUT-OFF ENTER ROOM RANGE	Dial room number or room range.
PRESS OK/SAVE KEY 1000-1001	Press * between phone numbers when entering room range. (ex, dial 1000*1001) Press the **[SAVE]** button.
CO LINE (0-1) DENY (1)	Designate whether to use truck call. (0: Allow, 1: Deny) (ex, dial 1) And then, press the **[SAVE]** button.
01(1000): SUCCESS 02(1000): SUCCESS	
	or
01(1000): FAILURE 02(1000): FAILURE	

Conditions

1. It is possible to register or cancel room cut individually or entirely from PMS.
2. If room cut off is registered from PMS, registered room number, cut off status and input location must be entered. If room cut off is registered from Front Desk, registered room number, cut off status must be entered.
3. If C/O line is accessed for an outgoing call from a room, room cut setting is checked. If room cut is registered, the call is released.
4. Upon check-in, room cut off is disabled automatically. So, the guest can make a CO line call after check-in.
5. Upon check-out, room cut off is enabled automatically. So, it is not possible to make a CO line call from a guest station after check-out. But if it is disabled from front desk or by admin **Programming**, CO line call becomes available.

Programming

Hotel Data	1	Room Cut Off (PGM 502 – Index6)

Related Features

Hardware

11.5.2 ICM Call Barring

Description

ICM call barring can be set from the front desks.

If rooms belong to groups other than PMS (Property Management System) group 0, a call is always possible among rooms belonging to the same group. If, however, a call is made between rooms belonging to different groups, a call is possible only when ICM call is permitted. For guest stations belonging to PMS group 0, a call is possible between rooms only when ICM call is permitted. Hotel guest rooms can call Service Station, Front Desk and Hotel Attendant regardless of ICM call barring.

A call can be possible or impossible between hotel and office extension depending on Admin setting.

The following table shows call restriction relationship between terminals in the system.

<table>
<tr><th rowspan="3">Extension Type</th><th colspan="4">Guest Station</th><th rowspan="3">Service Station</th><th rowspan="3">Front Desk</th><th rowspan="3">Attendant</th><th rowspan="3">Office Extension</th></tr>
<tr><th colspan="2">ICM Call Not Allowed</th><th colspan="2">ICM Call Allowed</th></tr>
<tr><th>Same Group</th><th>Different Group</th><th>Same Group</th><th>Different Group</th></tr>
<tr><td>Guest Station</td><td>O</td><td>X</td><td>O</td><td>O</td><td>O</td><td>O</td><td>O</td><td>Option</td></tr>
<tr><td>Service Station</td><td colspan="4">O</td><td>O</td><td>O</td><td>O</td><td>Option</td></tr>
<tr><td>Front Desk</td><td colspan="4">O</td><td>O</td><td>O</td><td>O</td><td>Option</td></tr>
<tr><td>Attendant</td><td colspan="4">O</td><td>O</td><td>O</td><td>O</td><td>O</td></tr>
<tr><td>Office Extension</td><td colspan="4">Option</td><td>Option</td><td>Option</td><td>O</td><td>O</td></tr>
</table>

Note
O: Call is permitted
X: Call is not permitted
Option: If a call option generated from Hotel Normal Information is set to be 'permitted', a call is possible while if 'not permitted', a call is impossible.

Operation

Front Desk

To register ICM Call Barring

1. Dial the **[PGM]** button + "#32".
2. Dial a room phone number or a phone number range.
3. Press * between phone numbers when entering a phone number range.
4. Press the **[SAVE]** button.
5. Designate whether to use ICM. (0 : Allowed, 1 : Deny)
6. Press the **[SAVE]** button.
7. If it is successful, you may hear service set tone and check a result through phone LCD.
8. Otherwise, you may hear error tone.

PROCEDURE

[3] HOTEL CALL BLOCK [4] HOTEL ROOM SETTING	Press the **[PGM]** button and dial '#3'.
[1] HOTEL ROOM CUT-OFF [2] HOTEL ROOM ICM CALL	Dial '2'
HOTEL ROOM ICM CALL ENTER ROOM RANGE	Dial room number or room range.
PRESS OK/SAVE KEY 1000-1001	Press * between phone numbers when entering room range. (ex, dial 1000*1001) Press the **[SAVE]** button.
ICM CALL (0-1) ALLOW (0)	Designate whether to use icm call. (0: Allow, 1: Deny) (ex, dial 1) And then, press the **[SAVE]** button.
01(1000): SUCCESS 02(1000): SUCCESS	
	or
01(1000): FAILURE 02(1000): FAILURE	

Conditions

1. If not assigned a PMS group during check-in, the PMS group of the guest station becomes zero by default. Guest stations belonging to PMS group '0' cannot call each other. If, however, ICM Call is 'allowed', a call is possible.
2. Basically, a guest station can call hotel guest stations in the same PMS group except PMS group 0. In other words, if ICM Call is not allowed, a guest station can call guest stations only included in its PMS group. If, however, ICM Call is permitted, a guest station can call guest stations which belong to other PMS groups.
3. Regardless of ICM Call Barring, a guest station can make calls to all service stations, front desks and attendants. Generally, a guest station cannot call office extension. If, however, "Call from Guest Room To Office" of PGM501-Index4 is permitted, office extension can be called.
4. Regardless of ICM Call Barring, a service station can call other service stations, front desks and attendants. Generally, a service station cannot call office extension. If, however, "Call from Service Station To Office" of PGM501-Index5 is permitted, office extension can be called
5. Regardless of ICM Call Barring, a front desk can call guest stations, service stations, front desk and attendants. Generally, a front desk cannot call office extension. If, however, "Call from Front-Desk To Office" of PGM501-Index6 is permitted, office extension can be called.

Programming

Hotel Data 1 Intercom Enable (PGM 502 – Index3)

Related Features

Hardware

11.5.3 One-Time CO Call Use

Description

By the request of guest, front desk can enable one-time CO call to an intercom-only station. In this time calling station's COS is temporally changed to that of the charged station's COS. Guest can make only one external call for one request.

Operation

Front Desk

To enable ICM Only station to dial outside calls,

1. Dial the **[PGM]** button + “#33”.
2. Dial Calling Station Number
3. Dial * for a delimiter
4. Dial Charged Station Number
5. Press the **[SAVE]** button.

PROCEDURE

[3] HOTEL CALL BLOCK [4] HOTEL ROOM SETTING	Press the **[PGM]** button and dial ‘#3’.
[3] ONE-TIME CO CALL USE	Dial ‘3’
ONE-TIME CO CALL USE ENTER CALLING STN NUMBER	Dial calling station number.
PRESS * KEY 1000	Press * after entering calling station number.
ONE-TIME CO CALL USE ENTER CHARGED ROOM NUMBER	Dial charged room number.
PRESS OK/SAVE KEY 1001	Then, press the **[SAVE]** button.
01(1000): SUCCESS	
	or
01(1000): FAILURE	

Conditions

1. Charged station must be a checked-in room station.
2. At the end of CO call, the station cannot access CO line.
3. Calling station's COS is changed to be charged station's COS temporally.
4. Charged station pays for this call.
5. Print-out format is as follows. STA field means charged station and Remark field printed in the ACCOUNT CODE column means calling station. (ex: calling station:112, charged station:104)
6. Cost is calculated using charged station's call charge rate.

No	STA	CO	TIME	START	DIALED	CNT	COST	ACCOUNT CODE
0001	104	01	00:01:23	12/29/94 08:16	O001034507950	3	500	R 112

Programming

Related Features

Hardware

11.6 Room Setting

11.6.1 Wake-Up Registration/Cancellation

Description

It registers or cancels wake up of hotel guests from PMS (Property Management System), Attendant, Front Desk and Guest Station.

Operation

Attendant

<u>To register wake up for a room</u>

1. Press **[PGM]** button + “045”.
2. Press a phone number range of desired room(s).
3. Dial four digits of desired time suitable for 24 HR mode.
4. If wake up is required to be repeated, dial #.
5. Press the **[SAVE]** button.

<u>To cancel wake up for a room</u>

1. Press **[PGM]** button + “046”.

2. Press a phone number or a phone number range of desired room(s).
3. Check current wake up time.
4. Press the **[SAVE]** button.

Front Desk

To register wake up for a room

1. Press **[PGM]** button + “#41”.
2. Press a phone number range of desired room(s).
3. Press * between phone numbers when entering a phone number range.
4. Press the **[SAVE]** button.
5. Dial four digits of desired time suitable for 24 HR mode.
6. If wake up is required to be repeated, dial #.
7. Press the **[SAVE]** button.
8. If it is successful, you may hear service set tone and check a result through phone LCD.
9. Otherwise, you may hear error tone and check a result through phone LCD.

To cancel wake up for a room

1. Press **[PGM]** button + “#42”.
2. Press a phone number or a phone number range of desired room(s).
3. Press * between phone numbers when entering a phone number range.
4. Press the **[SAVE]** button.
5. Check current wake up time.
6. Press the **[SAVE]** button.
7. If it is successful, you may hear service set tone and check a result through phone LCD.
8. Otherwise, you may hear error tone and check a result through phone LCD.

Guest Station (Digital Phone)

To register wake up time

1. Dial **{Wake-up Register Feature Code}**. Or **[PGM]** button + “13”.
2. Dial four digits representing hour and minute.
3. Dial ‘#’ to repeat wake up time.
4. Press the **[SAVE]** button.

To cancel wake up time

1. Dial **{Wake-up Cancel Feature Code}**. Or press the **[PGM]** button + “14”.
2. Press the **[SAVE]** button.

Guest Station (Single Line Telephone)

To register wake up time

1. Lift the handset.
2. Dial Single Line Telephone **{Wake-up Register Feature Code}**.
3. Dial four digits representing hour and minute.
4. Dial ‘#’ to repeat wake up time.
5. Upon hook-flash, confirmation tone is heard.

To cancel wake-up time

1. Lift the handset.
2. Dial Single Line Telephone **{Wake-up Cancel Feature Code}**.
3. Upon hook-flash, confirmation tone is heard.

Conditions

1. Unless a room is checked in, it is impossible to register/cancel wake up time from front desk.
2. Unless wake up time is registered, it is impossible to cancel wake up time.
3. When wake up is registered / cancelled the result of wake-up service is notified to PMS.

Programming

Station Data	1	Wake-Up Time (PGM 134-Index8)
	2	Repeat Wake-up (PGM 134-Index9)
Table Data	1	Wake Up Answer Tone (Web Admin. PGM 290-Index65)
	2	Wake-Up Indication Ring (Web Admin. PGM 265-Index10)
Tenant Data	1	Wake-Up Retry Count (PGM 280-Index 5)
	2	Wake-Up Retry Time (PGM 280-Index 6)

Related Features

Hardware

11.6.2 Do Not Disturb Registration/Cancellation

Description

DND (Do Not Disturb) can be registered/canceled from Front Desk, Guest Station or PMS to a specific room.

Operation

Front Desk

<u>*To register/cancel DND*</u>

1. Press **[PGM]** button + “#43”.
2. Press a phone number or a phone number range of desired room(s).
3. Press * between phone numbers when entering a phone number range.
4. Press the **[SAVE]** button.
5. Select DND registration or cancellation. (0 : Register, 1 : Cancel)
6. Press the **[SAVE]** button.

7. If it is successful, you may hear service set tone and check a result through phone LCD.
8. Otherwise, you may hear error tone and check a result through phone LCD.

Guest Station (digital Phone)

To apply DND to prime directory number (P-DN)

1. Press the **[DND]** button when it is not active. **[DND]** LED turns on.

To remove DND from prime directory number (P-DN)

1. Press the **[DND]** button when it is active. **[DND]** LED turns off.

To apply DND to sub directory number (S-DN)

1. Press the DN button to register.
2. Dial **{DND Register/Cancel Feature Code}**. There is no change in **[DND]** LED of terminal.

To remove DND from sub directory number (S-DN)

1. Press the DN button to register.
2. Dial **{DND Register/Cancel Feature Code}**. There is no change in **[DND]** LED of terminal.

Guest Station (single Line Telephone)

To register DND

1. Dial **{DND Register/Cancel Feature Code}**. Service registration tone is heard.

To cancel DND

1. Dial the **{DND Register/Cancel Feature Code}**. Service registration tone is heard.

Conditions

1. Unless a room is checked in, it is impossible to register/cancel DND from front desk.
2. It is possible to register/cancel DND individually or entirely from front desk. The existing terminal registers only its own DND but front desk registers DND of other rooms.

Programming

Station Data	1	DND Access (PGM 132-Index4)
System Data	2	LED Color/Flash (PGM 234)

Related Features

Hardware

11.6.3 Message Wait Registration/Cancellation

Description

It registers or cancels Message Wait from Front Desk, Guest Station and PMS.

Operation

Front Desk

To register/cancel a message

1. Dial the **[PGM]** button + "#44".
2. Dial a message sender's phone number and "*".
3. Dial a message receiver's phone number or a phone number range.
   Press * between phone numbers when entering a phone number range.
4. Press the **[SAVE]** button.
5. Select message registration or cancellation. (0 : Register, 1 : Cancel)
6. Press the **[SAVE]** button.
7. If it is successful, you may hear service set tone and check a result through phone LCD.
8. Otherwise, you may hear error tone and check a result through phone LCD.

Guest Station (Digital Phone)

To leave message wait when there is no answer

1. Press the **[CALL BK]** button. Confirmation tone is heard.
2. Hang up the call.
3. **[CALL BK]** button LED blinks.

To leave message wait in DND mode

1. Press the **[CALL BK]** button. Confirmation tone is heard.
2. Hang up the call.
3. **[CALL BK]** button LED blinks.

To leave call back when the other party is in talk state

1. Press the **[CALL BK]** button. Confirmation tone is heard.
2. Hang up the call.

To make a call when a reserved ring is received

1. Lift the handset or press the **[SPEAKER]** button.
2. A call is made to the extension previously reserved.

To make a call when message wait is received

1. Press the blinking **[CALL BK]** button.
2. Then, the following screen is displayed.

MWI(00) VMS(00)

3. In order to check a type of waiting message, dial 1-2 and select message type.
   ‘1’ (MWI: Check a list of missed calls.)
   ‘2’ (VMS: Check VM message.)
4. Check MWI detail through Volume Up/Down.
5. Press the **[HOLD]** button to select an appropriate item.

Guest Station (Single Line Telephone)

To leave message wait when there is no answer

1. Make hook flash.
2. Dial **{Message wait register Feature Code}**. Confirmation tone is heard.
3. Hang up the call.

To leave message wait in DND mode

1. Make hook flash.
2. Dial **{Message wait register Feature Code}**. Confirmation tone is heard.
3. Hang up the call.

To make a call when message wait is received

1. Dial **{Message Wait Reply Feature Code}**.

To leave call back when the other party is in talk state

1. Make hook flash.
2. Dial **{Call Back/Queuing Register Feature Code}**.
3. Hang up the call.

To make a call when a reserved ring is received

1. Lift the handset.
2. A call is made to the extension previously reserved.

Conditions

1. It is possible to register message wait individually or entirely from PMS.
2. It is possible to register or cancel message wait individually or entirely from front desk. The existing message wait procedure is maintained.
3. Message wait is registered or cancelled from guest station. The existing message wait procedure is maintained.

Programming

Related Features

Hardware

11.6.4 Bath Alarm

Description

In case station with bath alarm device lifts handset and no other action is taken for pre-defined time (1st Dial Tone or 2nd Dial Tone) and Uncompleted Dial Error Tone timer (PGM 290), Alarm Ring and LCD Indication is presented to System Attendant Station. System Attendant can cancel Bath Alarm Ring by dialing Alarm Reset Code (#10) or if alarm condition is cleared, Bath alarm ring is removed.

Operation

Front Desk

To enable bath alarm for room:

1. Dial the **[PGM]** button + “#45”.
2. Dial a phone number or a phone number range of room.
3. Press * between phone numbers when entering a phone number range.
4. Press the **[SAVE]** button.
5. If it is successful, you may hear service set tone and check a result through phone LCD.
6. Otherwise, you may hear error tone.

PROCEDURE

[3] HOTEL CALL BLOCK [4] HOTEL ROOM SETTING	Press the **[PGM]** button and dial ‘#4’.
[5] ROOM BATH ALARM ENABLE [6] ROOM BATH ALARM DISABLE	Dial ‘5’
ROOM BATH ALARM ENABLE ENTER ROOM RANGE	Dial room number or room range.
PRESS OK/SAVE KEY 1000-1001	Press & between phone numbers when entering room range (ex, dial 1000*1001). Press the **[SAVE]** button.
01(1000): SUCCESS 02(1001): SUCCESS	
	or
01(1000): FAILURE 02(1001): FAILURE	

To disable bath alarm for room:

1. Dial the **[PGM]** button + “#46”.
2. Dial a phone number or a phone number range of room.
3. Press * between phone numbers when entering a phone number range.
4. Press the **[SAVE]** button.
5. If it is successful, you may hear service set tone and check a result through phone LCD.
6. Otherwise, you may hear error tone.

PROCEDURE

[3] HOTEL CALL BLOCK [4] HOTEL ROOM SETTING	Press the **[PGM]** button and dial ‘#4’.
[5] ROOM BATH ALARM ENABLE [6] ROOM BATH ALARM DISABLE	Dial ‘6’
ROOM BATH ALARM DISABLE ENTER ROOM RANGE	Dial room number or room range.
PRESS OK/SAVE KEY 1000-1001	Press & between phone numbers when entering room range (ex, dial 1000*1001). Press the **[SAVE]** button.
01(1000): SUCCESS 02(1001): SUCCESS	
	or
01(1000): FAILURE 02(1001): FAILURE	

To reset Bath Alarm Ring;

1. Lift handset or press **[SPEAKER]** button.
2. Dial Bath Alarm Reset code (#10) at system attendant station.
3. If bath alarm condition is already cleared, bath alarm ring will be removed.

Conditions

1. Bath alarm call (Ring) cannot be forwarded as a normal call.
2. System attendant can cancel bath alarm ring.
3. When bath alarm happens and is cleared by attendant, a message with time data will be printed through Output port according to PGM 231 – Index 4 CALL INFO.
4. Bath alarm feature is not available from SIP and WIFI telephones.

Programming

Hotel Data	1	Hotel Service Type (PGM 502 – Index1)
	2	Bath Alarm (PGM 502 – Index9)
	3	Tone Table (PGM 290)

Related Features

Hardware

11.6.5 Register/Change Authorization Code (Password)

Description

At front desk, it is possible to register or change the authorization code of hotel guest stations.

Operation

Front Desk

To Register/Change Authorization Code;

1. Dial the **[PGM]** button + "#47".
2. Dial a phone number or a phone number range of room.
3. Press * between phone numbers when entering a phone number range.
4. Press the **[SAVE]** button.
5. Dial the Authorization Code.
6. Press the **[SAVE]** button, or dial "*" or dial "#".

PROCEDURE

[3] HOTEL CALL BLOCK [4] HOTEL ROOM SETTING	Press the **[PGM]** button and dial '#4'.
[7] ROOM AUTHOR CODE [8] ROOM PREPAID MONEY	Dial '7'
ROOM AUTHOR CODE ENTER ROOM RANGE	
PRESS OK/SAVE KEY 1002-1003	Dial room number or room range. Press * between phone numbers when entering room range. Press the **[SAVE]** button
ENTER NEW PASSWARD	Dial Author code up to 12 digits. An authorization code may include any dial-pad digit except '*' and '#'. Press the **[SAVE]** button.
01(1002): SUCCESS 02(1003): SUCCESS	

Conditions

1. A user may enter an authorization code from any station to place a CO/IP call using walking COS feature.

2. An authorization code may include any dial pad digit except ‘*’ and ‘#’.
3. A user can save the authorization code when pressing ‘*’ or ‘#’.
4. The total number of Authorization Codes in system is 648 in iPECS-MG 300 and 180 in iPECS-MG 100.

Programming

Station Data	1	Password (PGM 131-Index 4)
Hotel Data	1	Hotel Service Type (PGM 502-Index1)

Related Features

Hardware

11.6.6 Register/Change Prepaid Money

Description

This feature allows the guest station to make a phone call according to his/her prepaid money. If the prepaid money is consumed during a conversation, a warning tone will be given and after Prepaid Call Warning Timer, the call will be disconnected. The guest room can't make outgoing CO calls once the prepaid money is used up.

SMDR displays this pre-paid money.

The total room charge is as follow;

[Total Room Charge] = [Total Charge] – [Prepaid Money].

Operation

Front Desk

To Register/Change Prepaid Money;

1. Dial the **[PGM]** button + “#48”.
2. Dial a phone number or a phone number range of room.

   Press * between phone numbers when entering a phone number range.
3. Press the **[SAVE]** button.
4. Dial the Prepaid Money.
5. Press the **[SAVE]** button.

PROCEDURE

[3] HOTEL CALL BLOCK [4] HOTEL ROOM SETTING	Press the **[PGM]** button and dial '#4'.
[7] ROOM AUTHOR CODE [8] ROOM PREPAID MONEY	Dial '8'
ROOM PRIPAID MONEY ENTER ROOM RANGE	
PRESS OK/SAVE KEY 1002-1003	Dial room number or room range. Press * between phone numbers when entering room range. Press the **[SAVE]** button
ENTER PREPAID MONEY	Dial Prepaid Money up to 6 digits except '*' and '#'. Press the **[SAVE]** button.
01(1002): SUCCESS 02(1003): SUCCESS	

Conditions

1. This feature is available for the CO lines that receive call-metering signal.
2. Prepaid money is included in SMDR information.
3. The currency unit and fraction of prepaid money can be programmed in PGM 232, which is also used in SMDR information.
4. If the prepaid money is consumed, the room extension can't make an outgoing CO call.

Programming

Station Data 1 Prepaid Call Usage (PGM 133-Index11)

Hotel Data 1 Hotel Service Type (PGM 502-Index1)

Related Features

Hardware

11.7 Room Swapping

Description

This feature allows a user to change the room without additional check-out and check-in procedure. The existing information will be delivered to the new room except Room Class item.

Operation

Front Desk

To swap rooms

1. Dial the **[PGM]** button + “#5”.
2. Dial a phone number of room which is currently being used and ‘*’.
3. Dial a phone number of an empty room to be used and ‘*’.
4. Decide whether to swap rooms. (0: Confirm, 1: Cancel)
5. Press the **[SAVE]** button.
6. If it is successful, you may hear service set tone.
7. Otherwise, you may hear error tone.

PROCEDURE

[5] HOTEL ROOM SWAP SET [6] HOTEL ROOM MAID STATUS	Press the **[PGM]** button and dial ‘#5’.
CURRENT ROOM INFO ENTER ROOM NUMBER	
PRESS * KEY 1005	Dial current room number and press *.
NEW ROOM INFO ENTER ROOM NUMBER	
PRESS * KEY	Dial new room number and press *.
ROOM SWAP (0-1) CONFIRM (0)	Dial 0 (Confirm) or 1 (Cancel) Press the **[SAVE]** button.
SUCCESS	

Conditions

1. The new room must be checked out before room swapping.
2. DND, Message Wait, Missed Call and Room Class are not to swap.
3. The following shows the data moved from the original room to the new one:

Intercom Enable / Guest Type / Room Status / PMS Group / Authorization Code
Check Out Schedule / LCD Language / Prompt Language / Station COS /
Check In Data / Wake Up / Prepaid Money / Bath Alarm / Baby Listening /

4. When room swapping is performed to a room, the active room monitoring for the current room is canceled automatically.
5. Room swapping is not available during CO call.

Programming

Hotel Data	1	Hotel Service Type (PGM 502-Index1)

Related Features

Hardware

11.8 Maid Status

Description

Maid status can be registered from front desk, guest station and PMS.

The following maid status is supported.

1. TO_BE_CLEANED
2. UNDER_CLEANING
3. READY_FOR_SALE
4. OUT_OF_SERVICE
5. UNDER_REPAIR
6. REPAIR_COMPLETED
7. ROOM_OCCUPIED

Operation

Front Desk

<u>*To change Maid Status*</u>

1. Dial **{Hotel Maid Status Feature Code}** or **[PGM]** button + “#6”.
2. Dial a phone number or a phone number range.
3. Press * between phone numbers when entering a phone number range.
4. Press the **[SAVE]** button.
5. Designate Maid Status. (1 ~ 7)
6. Press the **[SAVE]** button.

7. Designate MAID ID. (0 ~ 9999) (Default : 0)
8. Press the **[SAVE]** button.
9. If it is successful, you may hear service set tone and check a result through phone LCD.
10. Otherwise, you may hear error tone.

PROCEDURE

[5] HOTEL ROOM SWAP SET [6] HOTEL ROOM MAID STATUS	Press the **[PGM]** button and dial '#6'.
HOTEL ROOM MAID STATUS ENTER ROOM RANGE	
PRESS OK/SAVE KEY 1002-1003	Dial room number or room range. Press * between phone numbers when entering room range. Press the **[SAVE]** button.
ROOM STATUS (1-7) TO BE CLEANED (1)	Dial 1-7 Press the **[SAVE]** button.
ROOM MAID ID (0-9999) 0	Dial mail ID Press the **[SAVE]** button.
01(1002): SUCCESS 02(1003): SUCCESS	

Digital Phone

To change Maid Status

1. Dial **{Hotel Maid Status Feature Code}**.
2. Designate Maid Status. (1 ~ 7)
3. Press the **[SAVE]** button.
4. Designate MAID ID. (0 ~ 9999)
5. Press the **[SAVE]** button.
6. If it is successful, you may hear service set tone and check a result through phone LCD.
7. Otherwise, you may hear error tone.

PROCEDURE

ROOM STATUS (1-7) TO BE CLEANED (1)	Dial **{Hotel Maid Status}** Feature code.
ROOM STATUS (1-7) UNDER CLEANING (2)	Dial 1-7 Press the **[SAVE]** button.
ROOM MAID ID (0-9999) 0	Dial maid ID Press the **[SAVE]** button.

SUCCESS	

SLT

To change Maid Status

1. Dial **{Hotel Maid Status Feature Code}**.
2. Designate Maid Status. (1 ~ 7)
3. Make hook flash.
4. Designate MAID ID. (0 ~ 9999)
5. Make hook flash.
6. If it is successful, you may hear service set tone.
7. Otherwise, you may hear error tone.

Conditions

1. It is possible to register maid status individually or entirely from PMS.
2. It is possible to register maid status from front desk.
3. From guest room, maid status can be registered for itself.
4. It is impossible to register maid status from an office station or a service station.
5. If pressing the **[SAVE]** button without maid id, default value (0) is saved.
6. Maid ID is not shown when room status is print.

Programming

Hotel Data	1	Hotel Service Type (PGM 502-Index1)

Related Features

Hardware

11.9 Room Change/Status Print

11.9.1 Room Charge Display/Print

Description

Front desk can see total charge of a single room or multiple rooms on LCD display.

Operation

Front Desk

To display and print Room Charge Information (total charge only or detailed information);

1. Dial the **[PGM]** button + “#71”.
2. Dial station number + “*”.
3. Then, LCD will show the total charge of that station.
4. Dial station number + “*” again if you want to enter another station number.
   LCD will display the summation of total charge of those stations.
   In this case, system does not support room range input.
5. Repeat step 4) if there are more stations for data print.
6. To print the total charge, press the **[SAVE]** button.
7. To print the details of room charge, dial code '#' and then press **[SAVE]** button.
   The code # is a toggle key for printing total cost and printing detailed room charge.
   The default is a total charge print.

Printing Format

1. Total Charge Print Format ;

- Echo Mode

RM(100)	PRE-PAID (20000)	CHARGED (53000)	= 330000
RM(102)	PRE-PAID (20000)	CHARGED (14600)	= - 5400
RM_SUMS:	PRE-PAID (40000)	CHARGED (67600)	= 27600

- Non-Echo Mode

&SM00100 20000 53000 330000&SM00102 20000 14600- 5400
&SM01 40000 67600 27600

cf.) In Non-Echo Mode &SM00 means the string is about station and &SM01 means total information.

1. Detailed Information Print Format for rooms (ex. room 100);

HOTEL NAME

TOTAL CHARGE IN ROOM 100 (Guest Name: …………)

Check – In : 10/12/26 -18

Check – Out : 10/12/31-11 (5 days)

Start-Time	CO	Duration	Dialed-No.	Count	Call-Cost	Remark
12/27-13:00	01	00:00:32	O00182343507851	0	0	Unanswered
12/27-13:30	01	00:01:23	O00182343507951	3	1500	
12/28-21:22	02	00:01:18	O00182343507953	31	15500	

Charge-Time	Charged-STA	Item	Bar-Cost	Tax
12/28-21:32	COFFEE-SHOP	COFFEE	5000	100
12/29-10:10	FRONT-DESK	COKE	3000	30

Item	Charge	Tax(rate)	Sum
(1) ROOM CHARGE: (rate 02: GOLD)	300000	30000(10.00%)	330000
(2) CALL CHARGE:	17000	170(10.00%)	17170
(3) BAR CHARGE:	8000	130	8130
(4) PRE-PAID			-20000

Method of Payment: VISA

TOTAL : 335300 WON

Note
Currency Unit/SMDR Fraction depend on Admin Programming232

1. Detailed Information Print Format for a service station (ex. Service station 101);

HOTEL NAME

TOTAL CHARGE IN SERVICE STATION 101

Start-Time	CO	Duration	Dialed-No.	Count	Call-Cost	Remark
12/27-13:00	01	00:00:32	O00182343507851	0	0	Unanswered
12/27-13:30	01	00:01:23	O00182343507951	3	1500	
12/28-21:22	02	00:01:18	O00182343507953	31	15500	

Item	Charge	Tax(rate)	Sum
(1) CALL CHARGE:	17000	170(10.00%)	17170
(2) PRE-PAID:		0	
TOTAL :		17170 WON	

Note
Currency Unit/SMDR Fraction depend on Admin Programming232

Conditions

1. Total number of Bar Records is 7000 in iPECS-MG 300 and 3000 in iPECS-MG 100; alarm message is automatically received at the Attendant Station if recorded number is 6000 or 6500 in iPECS-MG 300 and 2000 or 2500 in iPECS-MG 100.
2. After printing, system sends a Form Feed.
3. The information print of Guest Room can be blocked by PGM501-F8 Check-In/Out Print.

Programming

System Data	1	SMDR Attribute (PGM 232)
Hotel Data	1	Hotel Service Type (PGM 502-Index1)

Related Features

Hardware

11.9.2 Print Room Status through Off-line SMDR port

Description
This feature allows printing room status through Off-line SMDR port.

Operation
Front Desk

To print Room Status

1. Dial the **[PGM]** button + “#72”.
2. Dial a phone number or a phone number range.
   Press * between phone numbers when entering a phone number range.
3. Dial the **[SAVE]** button to print the status.

Conditions
1. Data for the service stations within room range input will not be printed.
2. If room is vacant, then only maid status will be printed.
3. Room status is printed and printed format is as follows,

Status

S	ROOM_NO	CHECK_IN	GUEST_NAME	COS	ICM	GRP	WAKE-UP	AUTH	PRE-PAID	CALL_RATE	CHARGE
7	1000	12/29-23	C.H.kim	010101	Yes	00000	..:..	No	20000	1	12345
7	1002	12/25-18	C.B.Choi	010101	No	00000	07:00	No	0	1	500000
7	1003	12/29-15	I.S.Lee	010101	No	00000	08:00	Yes	0	2	500
1	1004	Vacant									
1	1005	Vacant									

- S means maid status. There are 7 kinds of maid status.
- 1.To Be Cleaned 2. Under Cleaning 3. Ready For Sale 4. Out Of Service
  5. Under Repair 6. Repair Complete 7. Room Occupied
- The content of COS field is DAY/NIGHT/TIMED
- PRE-PAID and CHAGE fields use the SMDR fraction in PGM 232.

4. The information print of Guest Room can be blocked by “Check In/Out On-line Print” (PGM 501-Index8)

Programming
System Data 1 Off-line SMDR/Statistics Print (PGM 231-Index2)

Related Features

Hardware

11.9.3 Deleting Service Station's SMDR Record

Description

Front desk can delete service station's SMDR record.

Operation

Front Desk

To delete the SMDR records of a Service Station.

1. Dial the **[PGM]** button + "#73".
2. Dial a service station number + "*".
3. LCD will show the total charge of that station.
4. If you want to enter another service station number, dial station number + "*".
   LCD will display the summation of total charge of those stations.
   In this case, system does not support room range input.
5. Repeat step 4) if there are more stations for SMDR deletion.
6. Press the **[SAVE]** button to delete SMDR records of selected service station.

Conditions

Programming

System Data	1	SMDR Attribute (PGM 232)
Hotel Data	1	Hotel Service Type (PGM 502-Index1)

Related Features

Hardware

11.10 Room Rate

11.10.1 Room Rate Register/Assign

Description

This feature allows the front desk to assign a room rate to guest rooms among 20 different ones, which is also possible to program in Admin. Room rate information is included in the total room charge bill that printed out on the SMDR upon request and upon Check-Out.

Operation

Front Desk

Room type and rate program

1. Dial the **[PGM]** button + “#81”.
   LCD will display the room cost and room type name of the first entry in room rate table.

RATE (01) F1-F2 0000000 ......

2. Dial a desired room rate bin number (01 ~ 20) if you want to program another entry.
   LCD will display the room cost and room type name of the selected entry.
3. Press Flex button 1 and dial room cost (7 digits).
4. Press **[SAVE]** button to update the database permanently.
5. Press Flex button 2 and enter room type name (Max 6 characters).
6. Press **[SAVE]** button to update the database permanently.
   Go to step 2) to program another entry.

PROCEDURE

[7] ROOM CHARGE/STATUS [8] HOTEL ROOM RATE	Press the **[PGM]** button and dial '#8'.
[1] ROOM RATE REGISTER [2] ROOM RATE ASSIGN	Dial '1'
RATE (01) F1-F2 000000 …..	Dial room rate bin number (01-20) (By Default, 01 bin will be displayed
RATE (02) F1-F2 000000 ……	Press the desired Flex button (1-2). Flex 1: to configure room cost. Flex 2: to configure room type name.
02 RATE COST (7DGTS) 0020000	Press Flex button 1 to configure room cost. Dial 7 digit-length COST and Press the **[SAVE]** button (ex, dial '0020000')
02 ROOM TYPE NAME SINGLE	Press Flex button 2 to configure room type name. Enter 6 characters and Press the **[SAVE]** button.
RATE (02) F1-F2 0020000 SINGLE	

<u>To assign room rate to rooms</u>

1. Dial the **[PGM]** button + "#82".
2. Dial a phone number or a phone number range.
   Press * between phone numbers when entering a phone number range.
3. Press the **[SAVE]** button.
4. LCD displays currently assigned Room Rate Bin No. of the first room of selected range. For example, if selected rooms are 100 - 105, LCD will display as follows:

   ROOM 100-105
   01 : 0020000 SINGLE

5. Dial a desired room rate bin number (01 ~ 20).
6. Press **[SAVE]** button to update the database permanently.
   Go to step 2) to assign a room rate to another room(s).

PROCEDURE

[7] ROOM CHARGE/STATUS [8] HOTEL ROOM RATE	Press the **[PGM]** button and dial ‘#8’.
[1] ROOM RATE REGISTER [2] ROOM RATE ASSIGN	Dial ‘2’
ROOM RATE ASSIGN ENTER ROOM RANGE	
PRESS OK/SAVE KEY 1000-1001	Dial room number and press * or Dial room range. Press * between phone numbers when entering room range (ex, dial 1000* or dial 1000*1001) Press the **[SAVE]** button.
ROOM 1000-1001 ..: 000000 ……	Dial 2 digit-length room rate bin number (01 ~ 20) (ex, dial 01)
ROOM 1000-1001 01: 3432567 SINGLE	Press the **[SAVE]** button. A confirmation tone is received.

Conditions

Programming

Hotel Data

1 Rate For Room Class (PGM 503)

2 Hotel Station Info (PGM 502)

Related Features

- Room Charge Display/Print

Hardware

11.10.2 Fee For Part Time

Description

In case check-in and check-out happens in a day, part time fee may be charged according to the room type or hotel policy. Each room type may have up to 6 fields for different part time range and fee. There are 32 entries in part time table in which the part time range and associated fee can be programmed.

e.g). Room type (class) table (PGM 503)

room type	Charge	part time bins
.	.	.
.	.	.
GOLD	100$	02, 04
.	.	.
.	.	.
.	.	.

Part time table (PGM 507)

Bin	part time range	rate
01	.	.
02	00 – 03 hours	50 %
03		
04	04 – 12 hours	80 %
05	13 – 15 hours	90 %
.	.	.
.	.	.

If the room type of guest station is GOLD and the room is used for 3 hours, $50 will be charged for the room. It is a price 50% off from the original rate (100$). But for 13 hours, $100 will be charged because room type GOLD has no part time **Programming** for 13-hour stay, which is the regular price.

Operation

Front Desk

Register and change fee for part time table:

1. Dial the **[PGM]** button + “#83”.

LCD will display the duration of stay and its fee of the first entry in part time table.

```
PART TIME FEE (01) F1-F2
01 - 03 : 050 %
```

2. Dial a desired room rate bin number (01 ~ 32) if you want to program another entry.

LCD will display the duration of the stay and its fee of the selected entry.

3. Press Flex button 1, and :
   Dial the duration of stay (4 digits) and press **[SAVE]** button.
   Or, press **[SAVE]** button with empty data to erase the data.
4. Press Flex button 2, and :
   Dial the associated rate (000 ~ 100).
   Or, press **[SAVE]** button with empty data to erase the data.
5. Go to step 2) to program another entry.

Upon check-out for the part time stay, guest bill will be like the following.

HOTEL NAME

TOTAL CHARGE IN ROOM 100 (Guest Name: ………..)

Check – In : 10/12/26 -13

Check – Out : 10/12/26-20 (7 hours)

Start-Time	CO	Duration	Dialed-No.	Count	Call-Cost	Remark
12/27-13:00	01	00:00:32	O00182343507851	1	0	Unanswered
12/27-13:30	01	00:01:23	O00182343507951	3	150	

Charge-Time	Charged-STA	Item	Bar-Cost	Tax
12/28-21:32	COFFEE-SHOP	COFFEE	50	10

Item	Charge	Tax(rate)	Sum
(1) ROOM CHARGE: (rate 02: GOLD)	80	8(10.00%)	88
(2) CALL CHARGE:	150	15(10.00%)	165
(3) BAR CHARGE:	50	10	60
(4) PRE-PAID			-200

Method of Payment: VISA

TOTAL : $113 USD

Note

Currency Unit/SMDR Fraction depend in PGM 232

PROCEDURE

[7] ROOM CHARGE/STATUS [8] HOTEL ROOM RATE	Press the **[PGM]** button and dial '#8'.
[3] ROOM PART TIME FEE [4] ROOM BAR/CHARGE	Dial '3'
PART TIME FEE (01) F1-F2 ..:.. %	Entry 01 will be displayed like left LCD.
	Dial 2-digit part time bin number (01 ~ 32) if you want to program another entry And, Press the desired Flex button (1~2). Flex 1: to configure the duration of stay Flex 2: to configure the part time rate
01 PART TIME RANGE (4DIGITS) ..-..	Press Flex button 1 to configure the duration of stay. Dial 4 digits for the duration of stay and press the **[SAVE]** button. Or, Press the **[SAVE]** button with empty data to erase the data. LCD will go back to previous stage.
PART TIME RATE (3DGTS) ..:.. %	Press Flex button 2 to configure the part time rate. Enter 3 digits-length part time rate (000 ~ 100) and press the **[SAVE]** button. Or, Press the **[SAVE]** button with empty data to erase the data. LCD will go back to previous stage.
PART TIME FEE (01) ..-.. ..%	

Conditions

1. Part time range and its rate can be programmed also in "Fee For Part Time" (PGM 507).
2. Maximum 32 entries can be programmed in part time table.
3. Rate field of the part time table denotes the percentage from the original price.
4. Part Time table has no default value.
5. Each room type can have up to 6 indices of the part time table.
6. If a room type has no index for part time table bin, or if the duration of stay is not included in the part time table, part time fee is not applied and the original price (one-day charge) will be charged.
7. Each room type has no default value for the index of part time table.
8. Lower number entries will be referenced first in part time table. That is, if the different entries include the same duration, lowest entry will be applied.
9. The duration of stay denotes the elapsed time from check-in to check-out. And only hour information is meaningful so that minute data is just ignored.
10. If nothing is programmed in part time table, one-day charge will be applied.

Programming

Hotel Data

1 Rate For Room Class (PGM 503)
2 Fee For Part Time (PGM 507)

Related Features

- Room Charge Display/Print

Hardware

11.10.3 Register Bar and Mini-bar Charge

Description

Guest may pay for bar charge at service station and the mini-bar charge upon check-out. Bar terminal operator registers guest's charge using the procedure below. Mini-bar charge of each room can be registered from front desk, service station and guest room. Registered bar / mini-bar charge is stored in bar record of system. And it can be displayed or printed upon hotel charge display and check-out.

Operation

Front Desk

<u>*To register bar or mini-bar charge*</u>

1. Dial the **[PGM]** button + "#84".
2. Dial the guest room number and press the **[SAVE]** button.
3. Dial product code (001 ~ 100).
4. Dial (mini) bar cost (Max 6 digits).

   LCD will show room no. and (mini) bar cost.
5. If you dial #, the cost becomes a minus value. The value is toggled between minus and plus by dialing #(+/- → -/+). Plus is the default sign.
6. Press the **[SAVE]** button to save the cost.

   If it is successful, you may hear service set tone. And (mini) bar cost is registered and SMDR Information for the (mini) bar cost will be printed through on-line SMDR port.

   Otherwise, you may hear error tone.

PROCEDURE

[7] ROOM CHARGE/STATUS [8] HOTEL ROOM RATE	Press the **[PGM]** button and dial '#8'.
[3] ROOM PART TIME FEE [4] ROOM BAR/CHARGE	Dial '4'
BAR COST CHARGE ENTER ROOM NUMBER	Dial room number and press the **[SAVE]** button.
PRESS OK/SAVE KEY 1001	(ex, dial 1001)
1001 BAR COST CHARGE ENTER BAR CODE (001-100)	Dial 3-digit length bar code (001 – 100).
1001 BAR COST CHARGE ENTER CHARGE (6 DGTS)	Dial (mini) bar cost (max 6digits).
PRESS OK/SAVE KEY 1001 078000	Press the **[SAVE]** button to save the cost.
1001 BAR COST CHARGE ENTER BAR CODE (001-100)	LCD will be changed for entering additional bar charge.

Service Station

To register bar or mini-bar charge

1. Dial **{Hotel Mini-bar}** feature code.
2. Dial the guest room number and press the **[SAVE]** button.
3. Dial product code (001 ~ 100).
4. Dial (mini) bar cost (Max 6 digits).

LCD will show room no. and (mini) bar cost.

5. If you dial #, the cost becomes a minus value. The value is toggled between minus and plus by dialing #(+/- → -/+). Plus is the default sign.
6. Press the **[SAVE]** button to save the cost.

   If it is successful, you may hear service set tone. And (mini) bar cost is registered and SMDR Information for the (mini) bar cost will be printed through On-line SMDR port.

   Otherwise, you may hear error tone.

PROCEDURE

SVC STATION 1000 DEC 09 10 03:36 pm	Dial **{Hotel Mini-bar}** feature code.
BAR COST CHARGE ENTER ROOM NUMBER	Dial room number and press the **[SAVE]** button.
PRESS OK/SAVE KEY 1001	(ex, dial 1001)
1001 BAR COST CHARGE ENTER BAR CODE (001-100)	Dial 3-digit length bar code (001 – 100).
1001 BAR COST CHARGE ENTER CHARGE (6 DGTS)	Dial (mini) bar cost (max 6digits)
PRESS OK/SAVE KEY 1001 078000	Press the **[SAVE]** button to save the cost.
1001 BAR COST CHARGE ENTER BAR CODE (001-100)	LCD will be changed for entering additional bar charge.

Guest Station (Digital Phone)

To register bar or mini-bar charge

1. Dial **{Hotel Mini-bar}** feature code.
2. Dial product code (001 ~ 100).
3. Dial (mini) bar cost (Max 6 digits).
   LCD will show room no. and (mini) bar cost.
4. If you dial #, the cost becomes a minus value. The value is toggled between minus and plus by dialing #(+/- → -/+). Plus is the default sign.
5. Press the **[SAVE]** button to save the cost.

   If it is successful, you may hear service set tone. And (mini) bar cost is registered and SMDR Information for the (mini) bar cost will be printed through On-line SMDR port.

   Otherwise, you may hear error tone.

PROCEDURE

ROOM 100 DEC 09 10 03:36 pm	Dial {Hotel Mini-bar Register}.
1001 BAR COST CHARGE ENTER BAR CODE (001-100)	Dial 3-digit length bar code (001 – 100).
1001 BAR COST CHARGE ENTER CHARGE (6 DGTS)	Dial (mini) bar cost (max 6digits).
PRESS OK/SAVE KEY 1001 078000	Press the **[SAVE]** button to save the cost.
1001 BAR COST CHARGE ENTER BAR CODE (001-100)	LCD will be changed for entering additional bar charge.

Guest Station (Single Line Telephone)

To register bar or mini-bar charge

1. Dial **{Hotel Mini-bar}** feature code.
2. Dial product code (001 ~ 100).
3. Dial (mini) bar cost (Max 6 digits).
4. If you dial #, the cost becomes a minus value. The value is toggled between minus and plus by dialing #(+/- → -/+). Plus is the default sign.
5. Make hook flash to save the cost.

   If it is successful, you may hear service set tone. And (mini) bar cost is registered and SMDR Information for the (mini) bar cost will be printed through On-line SMDR port.

   Otherwise, you may hear error tone.

Conditions

1. System sends the following information to the printer in the order (Charged station number, date/time, FRONT-DESK in case of mini-bar or service station's name for the other charges, product name and cost). Print-out data format is as follows.

   //100 12/30-22:12 (mini-bar) BEER 50000

2. This bar and mini-bar cost will be included in total room charge.
3. The unit of cost and fraction are the same as SMDR currency unit and SMDR fraction.
4. Product codes for bar and mini-bar items must be registered before usage or the product code you dialed will be printed out instead of the product name.
5. Minus cost will help you when you might charge by mistake. Minus cost must be lower than total bar cost.

Programming

Hotel Data

1 Mini-Bar List (PGM 505)

2 Print Port Selection (PGM 231)

Related Features

- Room Charge Display/Print

Hardware

11.11 Call Rate

11.11.1 Call Charge Rate Register/Assign

Description

This feature allows front desk to assign call charge rate to an individual room or to a service station so that a different call charge rate can be applied. To use this feature, at first front desk must register call charge rate table, and then assign the registered rate to each room. Call charge rate denotes the percentage on SMDR call cost

Operation

Front Desk

Register or change an entry in call charge rate table

1. Dial the **[PGM]** button + "#91".
2. Dial a desired room rate bin number (1 ~ 6) if you want to program another entry.
   LCD will display the call charge rate of the selected entry.
3. Press Flex button 1 and dial the call charge rate (3 digits).
   Press **[SAVE]** button with empty data to erase the data.
4. Press Flex button 2 and enter the name of rate (Max 6 characters).
5. Press **[SAVE]** button to update the database permanently.
   Go to step 2) to program another entry.

PROCEDURE

[9] HOTEL ROOM CALL RATE [0] HOTEL MISC PROGRAM	Press the **[PGM]** button and dial '#9'.
[1] ROOM CALL RATE REGISTER [2] ROOM CALL RATE ASSIGN	Dial '1'.
ROOM CALL RATE REGIST ENTER BIN NO (1-6)	Dial call charge rate bin number(1 ~ 6). (For example, dial 1)
CALL RATE (1) F1-F2 ……	Press the desired Flex button(1 ~ 2). Flex 1 : to configure call charge rate. Flex 2 : to configure call charge type name.
1 CALL CHARGE RATE (3 DGTS) 090	Press Flex button 1 to configure call charge rate. Dial 3 digit-length RATE and Press the **[SAVE]** button and LCD will be changed that of upper stage.(ex, dial '090')
HOLIDA 1 CALL CHARGE RATE NAME (…)	Press Flex button 2 to configure call charge rate type name, enter 6 characters and Press the **[SAVE]** button.(ex, enter 'HOLIDA').
CALL RATE (1) F1-F2 090% HOLIDA	

To assign call charge rate to rooms;

1. Dial the **[PGM]** button + “#92”.
   Dial a phone number or a phone number range.
2. Press * between phone numbers when entering a phone number range.
3. Press the **[SAVE]** button.
4. LCD displays currently assigned index of call charge rate table of the first room in selected range. For example, if selected rooms are 100 - 105, LCD will display as follows:

```
ROOM 100 - 105
1 : 090 %    HOLIDA
```

5. Dial a desired call charge rate bin number (1 ~ 6).
6. Dial # to erase the bin number.
7. Press **[SAVE]** button to update the database permanently.
   Go to step 2) to assign a room rate to another room(s).

PROCEDURE

[9] HOTEL ROOM CALL RATE [4] HOTEL MISC PROGRAM	Press the **[PGM]** button and dial ‘#9’.
[1] ROOM CALL RATE REGISTER [2] ROOM CALL RATE ASSIGN	Dial ‘2’
ROOM CALL RATE ASSIGN ENTER ROOM RANGE	Dial room number or room range.
PRESS OK/SAVE KEY 1000-1001	Press * between phone numbers when entering room range (ex, dial 1000*1001). Press the **[SAVE]** button.
ROOM 1000-1001 2: 080%.........VIP	Dial 1 digit-length call charge rate bin number(1 - 6).(ex, dial 2)
ROOM 1000-1001 2: 080% VIP	Press the **[SAVE]** button.

Conditions

1. There can be up to 6 entries in the call charge rate table (from 1 to 6).
2. If a room or a service station is not assigned a call charge rate, 100 percent of SMDR call cost is charged.
3. It is optional to program the call charge rate type.

Programming

Hotel Data 1 Call Charge Rate (PGM 504)

Related Features

Hardware

11.12 Register Hotel Name

Description

This feature allows the front desk to register or change the name of hotel. The registered hotel name will be displayed on the Receipt when checking out.

Operation

Front Desk

<u>*To register hotel's name*</u>

1. Dial the **[PGM]** button + "#01".
2. Enter the name of hotel (Max 24 characters).
3. Press the **[SAVE]** button.

[TRANS/PGM] +#01 + Hotel Name + **[HOLD/SAVE]**.

PROCEDURE

[9] HOTEL ROOM CALL RATE [0]> HOTEL MISC PROGRAM	Press the **[PGM]** button and dial '#0'.
[1]> HOTEL NAME [2] SET CALL FORWARD	Dial '1'
ENTER HOTEL NAME	
IPECS MG ENTER HOTEL NAME	Enter the name of hotel (Max 24 characters) Press the **[SAVE]** button.

Conditions

1. By default, there is no hotel name programmed.
2. Upon check-out or room charge display, registered hotel name will be printed in the header.
3. When you use Ez-Input terminal (for example, 7024LD, 6024LDH and LIP8012, etc), there may be limited to 12 characters.
4. Character set for hotel name is like the following:

Table 11.12-1 Alphanumeric Entry Chart

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

Programming

Hotel Data 1 Hotel Name (PGM 500-Index1)

Related Features

Hardware

11.13 Set Call Forward

Description

This feature allows the front desk to set the call forward to the other station (internal or external), station group or CO line for rooms on demand of guests.

Operation

Front Desk

To activate call forward for the room(s)

1. Dial the **[PGM]** button + "#02".
2. Dial a phone number or a phone number range.
   Press * between phone numbers when entering a phone number range.
3. Press the **[SAVE]** button.
4. Dial 1-4 **{Forward Code}** as appropriate.
5. Dial the station or station group to receive calls.
   Or, Dial CO Group Access code and desired external phone number.
6. Press the **[SAVE]** button.
7. Replace the handset, return to idle.

PROCEDURE

[9] HOTEL ROOM CALL RATE [0]> HOTEL MISC PROGRAM	Press the **[PGM]** button and dial '#0'.
[1] HOTEL NAME [2]> SET CALL FORWARD	Dial 2
SET CALL FORWARD ENTER ROOM RANGE	
PRESS OK/SAVE KEY 1002	Dial room number or room range. Press * between phone numbers when entering room range. Press the **[SAVE]** button.
ENTER FORWARD TYPE (1-4, #)	Dial 1-4 Press the **[SAVE]** button.
UNCONDITION FORWARD ENTER NUMBER	
UNCONDITION FORWARD 4501124	Dial the station or station group to receive calls. Or, Dial CO Group Access code and desired external phone number. Press the **[SAVE]** button.
01(1002): SUCCESS	

To deactivate call forward for the room(s)

1. Dial the **[PGM]** button + “#02”.
2. Dial a phone number or a phone number range.
   Press * between phone numbers when entering a phone number range.
3. Press the **[SAVE]** button.
4. Dial # to cancel call forward.
5. Replace the handset, return to idle.

PROCEDURE

[9] HOTEL ROOM CALL RATE [0]> HOTEL MISC PROGRAM	Press the **[PGM]** button and dial ‘#0’.
[1] HOTEL NAME [2]>SET CALL FORWARD	Dial 2
SET CALL FORWARD ENTER ROOM RANGE	
PRESS OK/SAVE KEY 1002	Dial room number or room range. Press * between phone numbers when entering room range. Press the **[SAVE]** button.
ENTER FORWARD TYPE (1-4, #)	Dial #
01(1002): SUCCESS	

Conditions

1. Front desk can set the room’s call forward with the call forward type (Unconditional, busy, no-answer and busy/no-answer).

Programming

Hotel Data 1 Hotel Service Attribute (PGM502-Index 1)

Related Features

Hardware

11.14 Additional Tax Fields

Description

Different tax rates can be applied for the various charges in hotel.

e.g.) Tax 1 = 10.00 %
Tax 2 = 20.00 %
Tax 3 = 0 %

Each bar item in the Mini-bar List (PGM 505) can have a tax rate index such as TAX 1, TAX 2, TAX 3, and etc.

Tax rates for call charge and room charge is fixed as the first tax rate entry, that is, Tax 1.

Operation

Front Desk

Register or change an entry in Additional Tax Fields table

1. Dial the **[PGM]** button + "#85".
2. Dial a desired tax rate bin number (1 ~ 5) if you want to program another entry.
   LCD will display the tax rate of the selected entry.
3. Dial 4 digit-length tax rates.
   Press **[SAVE]** button with empty data to erase the data.
4. Press **[SAVE]** button to update the database permanently.
   Go to step 2) to program another entry.

PROCEDURE

[7] ROOM CHARGE/STATUS [8] HOTEL ROOM RATE	Press the **[PGM]** button and dial '#8'.
[5] ADDITIONAL TAX FIELDS	Dial '5'
ADDITIONAL TAX FIELDS ENTER BIN NO (1-5)	Dial Tax Rate number (1-5). (For example, dial 1)
TAX (1) 00.00%	Current tax rate of bin 1 will be displayed.
PRESS OK/SAVE KEY 1000	To assign new tax rate, dial 4 digit-lengths TAX RATE and press the **[SAVE]** button (ex, dial '1000').
TAX (1) 10.00%	**Note** 2.22% → 0222 or 222 22.2% → 2220

Conditions

1. Tax index for the telephone and room charge is 1, that is, Tax 1 will be applied.
2. There are up to 5 programmable tax rates (1 to 5) in PGM 506.
3. Every tax rates has 00.00 % by default and may have the value from 00.00 % to 99.99%.
4. Each bar item can have only one index of tax rate.
5. The index of tax rate in bar item table is 1 by default.
6. Total tax of each item is only the integer part of the result that is calculated by the following equation.
   Equation for tax: [Total tax] = INTEGER ([Charge] * [Tax rate] * 100 / 10000).

Programming

Hotel Data	1	Tax Rate For Bill (PGM506)

Related Features

Hardware

11.15 Guest Name/Info Display

Description

It allows users to view information about guests from front desk while talking with guests over the phone.

To use this Guest Info Display feature at front desk, **{Hotel Guest Info Display}** feature code must be assigned to the flexible button.

Users can press the button which is assigned **{Hotel Guest Info Display}** feature code to display guest information during call and a volume button to move between queried information.

Operation

Front Desk

To display information about guests in talk state

1. Press the flexible button of **{Hotel Guest Info Display}** feature code during conversation.
2. Detailed information about the guest can be seen: 24char LCD (32char LCD)

   00 : NAME (GUEST NAME)

   01 : GRADE (GUEST GRADE)

   02 : CHECKOUT (CHECK OUT)

   03 : LANGUAGE (LCD LANGUAGE)

   04 : CO CALL (CO LINE CALL)

   05 : DND (DND SETTING)

   06 : WAKE UP (WAKE UP TIME)

   07 : DAY COS (DAY COS)

   08 : NIGHT COS (NIGHT COS)

   09 : TIMED COS (TIMED COS)

   10 : DGT CONV TBL (DIGIT CONVERSION TABLE)

   11 : PMS GRP (PMS GROUP)

   12 : MSG NO (MSG WAIT NO)

   13 : MSG SEND (MSG SENDER)
3. It is possible to scroll the information up and down using a volume button.

PROCEDURE

GUEST NAME/INFO DISPLAY	Press the flexible button of **{Hotel Guest Info Display}** feature code during call.
00 NAME: ROOM 01 GRADE: NO VIP	Press up and down using a volume button to scroll the information.
02 CHECK OUT 03 LANGUAGE: ENGLISH	
04 CO LINE CALL: OFF 05 DND : OFF	
06 WAKE UP 07 DAY COS: 1	
08 NIGHT COS : 1 09 TIMED COS : 1	
10 DGT CONV TBL : 1 11 PMS GRP : 0	
12 MSG NO : 0 13 MSG SEND : …	

Conditions

1. It is possible to display guest info from front desk.
2. Display the first number of message wait in MSG SEND.

Programming

Hotel Data

1 Guest Info Display(Station) (PGM500-Index 17)

2 Hotel Service Attribute (PGM502-Index 1)

Related Features

Hardware

11.16 Dial One Digit Service

Description

One Digit service that allows customers simply dial an extension or a service station on the front.

When a guest dials only one digit of the dial pad(0~9, *, #) and waits for a certain period of time, a call is made to a specific extension registered to PGM 508(front desk, service station).

Operation

Guest Station

To use Dial One Digit Service

1. Lift the handset or press the **[SPEAKER]** button.
2. Dial one digit registered (0~9, *, #).
3. After a 'Dial One Digit Service Timer', a call is made.

Conditions

1. Call restriction and class information are maintained even when this feature is used.
2. This feature is not available for office extension.
3. If there are one-digit numbering plan codes, one digit service is not available for those digits. That is, if the CO Group Access Code value is set to 9 in CO Group Access Code (PGM114), then One Digit Service for digit 9 is not available. Or, if the Attendant Call feature code is set to 0 in Feature Numbering Plan (PGM113), then One Digit Service for digit 0 is not available.

Programming

Hotel Data	1	Dial One Digit Service Timer (PGM500-Index 20)
	2	Dial One Digit Service (PGM508)

Related Features

Hardware

11.17 Room Monitor/Baby Listening

Description

It allows guests to monitor their rooms outside of the rooms in a hotel. In order to perform this function, an extension to be monitored must be set to room monitor mode. If monitoring is activated, all sounds of the monitored station can be heard from the monitoring extension. But the monitoring party's sounds are not delivered to the monitored extension.

Operation

Guest Station

To put an extension into monitor mode

1. Lift the handset or press the **[SPEAKER]** button.
2. Dial **{Hotel Room Monitor}** feature code.
3. Service set tone is heard and the extension is monitored.

To clear extension monitoring

1. Hang up the monitored station or press the active **[SPEAKER]** button.
2. The extension goes on-hook and extension monitor is finished.

To monitor a guest station

1. Lift the handset or press the **[SPEAKER]** button.
2. Dial an extension number to monitor.
3. DND rejection tone is heard.
4. Dial the same phone number.
5. Sound of the monitored room is heard.

Conditions

1. This feature is available only for internal call in the system.
2. Monitor mode can be set only from a checked-in guest station.
3. If the monitored extension is called, DND rejection tone is heard. In this case, step call and camp-on cannot be performed, but message wait, redial and pilot hunt can be performed.
4. If a 3rd party calls the extension currently being monitored, busy tone is heard. Then, the 3rd party can perform step call or call back.
5. SIP extension cannot be monitored.

Programming

Hotel Data	1	Room Monitor (PGM502-Index 10)

Related Features

Hardware

11.18 Call Answer Recognition

Description

If ISDN line is used for outgoing calls then it is required that a called party answer signal is recognized by the system and flagged on the guest's bill and call logging SMDR output

Operation

When a call is made from a guest station and the call is disconnected without answer, the call will be displayed on the guest's bill as follows:

HOTEL NAME

TOTAL CHARGE IN ROOM 100 (Guest Name: …………)

Check – In : 10/12/26 -18

Check – Out : 10/12/31-11 (5 days)

Start-Time	CO	Duration	Dialed-No.	Count	Call-Cost	Remark
12/27-13:00	01	00:00:32	O00182343507851	0	0	Unanswered
12/27-13:30	01	00:01:23	O00182343507951	3	1500	
12/28-21:22	02	00:01:18	O00182343507953	31	15500	

Charge-Time	Charged-STA	Item	Bar-Cost	Tax
12/28-21:32	COFFEE-SHOP	COFFEE	5000	100
12/29-10:10	FRONT-DESK	COKE	3000	30

Item	Charge	Tax(rate)	Sum
(1) ROOM CHARGE: (rate 02: GOLD)	300000	30000(10.00%)	330000
(2) CALL CHARGE:	17000	1700(10.00%)	18700
(3) BAR CHARGE:	8000	1600	9600
(4) PRE-PAID			-20000

Method of Payment: VISA

TOTAL : 338300 WON

Note
Currency Unit/SMDR Fraction depend on PGM 232

The SMDR print out will show the following:

NO	STA	CO	TIME	START	DIALED	CNT	COST	REMARK
0009	101	01	00:04	13/06/95 12:47	o123	1	0.10	
0010	101	01	00:15	13/06/95 12:50	o01444872014	0	0.00	UNANSWERED

Alphabet small o in the DIALED field means unanswered outgoing call using ISDN line.

Conditions

1. This feature will be applied while hearing a ring back tone before answer.
2. The feature will be available only when ISDN is seized for outgoing
3. No cost will be charged to stations for the unanswered calls.
4. This feature is applied to the SMDR record

Note
This feature is applied for UK, NZ, and Australia.
In NZ and Australia, this feature is extended for LCOB lines.

Programming

System Data 1 SMDR Attribute (PGM232)

Related Features

Hardware

11.19 Form Feed Button

Description

The print-out contents through the RS-232C serial port can be divided in 2 categories.

Something is for the hotel archive and something is for the hotel guest.

Some printers do not have a form feed button, and it is not possible to terminate the print for the archive and to start printing a bill for a hotel guest on the next piece of paper.

At front desk, by pressing form feeding button, print-out can be started from the next page as user wants.

Operation

Front Desk

*To make a **{FROM FEED}** button:*

[PGM] + **{FLEX button}** + NUM (1) + **{Hotel Form Feed}** Feature Code + **[SAVE]**

To print out form feed:

1. Lift the handset or press the **[SPEAKER]** button.
2. Dial **{Hotel Form Feed}** Feature code or press **{FROM FEED}** button.
3. Form feed will be printed to RS-232C.

Condition

1. It is possible to program a **{FORM FEED}** button only from front desk.
2. Form feeding is available only if P-Break is ON in COM port setting (PGM 230).

Programming

Related Features

Hardware

11.20 VIP Guest Call

Description

It allows an operator to answer the calls from VIP guests earlier when VIP guests call an attendant. Although an attendant can answer many calls at the same time since it has a queue, call answer is sequential in the order of call's entrance into the queue. If prioritized queuing of VIP guest call is set, calls from VIP guests are firstly answered by an operator.

Operation

Guest Station (VIP)

<u>When VIP Attendant Call service is enabled:</u>

1. All group members are busy in the attendant group or a station hunt group.
2. There are more calls to the group from non-VIP guests.
3. Calls are put on the queue and the callers will hear queuing tone.
4. VIP guest makes a call to the group and hear the queuing tone.
5. One of the group members go idle.
6. In a few seconds, VIP guest call will be served first. That is, the idle group member will receive the call from VIP guest while other callers are still hearing queuing tone.

Conditions

1. This feature is available for attendants and station hunt group members.
2. VIP guests are determined depending on guest grade designated upon check-in.

Programming

Station Group	1	Station Group (PGM 200)
	2	Station Group Attribute (PGM 201-202)
Tenant Data	1	Attendant Group Assignment (PGM 270)
	2	Attendant Group Attributes (PGM 271-272)
	3	Night Group Assignment (PGM 275)
	4	Night Attendant Group Attribute (PGM 276-277)
Hotel Data	1	VIP Attendant Call Service (PGM 500 – Index18)

Related Features

Hardware

11.21 VIP Guest Wake-Up Call

Description

This feature allows an attendant to be informed of VIP guests' wake-up call and provide wake-up call service. When it becomes wake-up call time for VIP guest room, the attendant receives the wake up alarm instead of VIP guest. Together with the alarm, the attendant can see VIP guest's room number. And if the attendant answers the wake up call, station number and name, station status and wake up date/time info will be displayed on LCD. The attendant can see other logs for VIP wake up by using Volume Up/Down button. After checking station status, the attendant can make a call to VIP guest by pressing **[SAVE]** button. If the attendant dials **{Hotel VIP Wake Up}** feature code, the list of unanswered wake up calls for VIP guests are shown and a call can be made to the VIP guest by pressing **[SAVE]** button.

Operation

*To make a **{Hotel VIP Wake Up}** button:*

[PGM] + **{FLEX}** + Button Feature Type (1) + **{Hotel VIP Wake Up}** feature code+ **[SAVE]**

Attendant

To check and serve VIP wake-up call when attendant receives VIP wake-up call:

1. Lift the handset or press the **[SPEAKER]** button.
2. Station number/name, wake-up information and current status of VIP guest is displayed on LCD. If there're multiple wake-up logs, the information can be scrolled by using the **[VOL UP]**/**[VOL DOWN]** keys
3. Press **[SAVE]** button to call the VIP guest.
4. If the guest answers the call, announce the wake-up.

To check and serve VIP wake-up call when attendant is idle:

1. Lift the handset or press the **[SPEAKER]** button.
2. Dial **{Hotel VIP Wake Up}** feature code or press **{Hotel VIP Wake Up}** feature code button.
3. Station number/name, wake-up information and current status of VIP guest is displayed on LCD. If there're multiple wake-up logs, the information can be scrolled by using the **[VOL UP]**/**[VOL DOWN]** keys
4. Press **[SAVE]** button to call the VIP guest.
5. If the guest answers the call, announce the wake-up.

Conditions

1. This feature is available only for attendants (day or night attendants).
2. VIP guests are determined depending on guest grade designated upon check-in.
3. If there is no idle attendant, wake up alarm is sent to the VIP guest room directly.
4. If the attendant does not serve the VIP wake-up for one minute, wake up alarm is sent to the VIP guest directly. So, the attendant must make a call to the VIP guest and talk with the guest to announce the wake up within one minute after VIP wake up alarm is started.
5. If the wake up setting is erased by VIP guest or attendant, the VIP wake up service is also canceled.
6. There are maximum 20 VIP wake-up logs in system. The wake up log is saved when one of the attendants receive the VIP wake up call. And it is deleted when one of the attendants makes a call to the VIP guest and talk with the guest to announce that it is wake up time.
7. If there is a wake-up log, the **{Hotel VIP Wake Up}** feature code button LED will flash at 240 ipm. Otherwise, it is turned off.

Programming

Station Data	1	Wake-Up Time (PGM 134-Index8)
	2	Repeat Wake-up (PGM 134-Index9)
Table Data	1	Wake Up Answer Tone (Web Admin. PGM 290-Index65)
	2	Wake-Up Indication Ring (Web Admin. PGM 265-Index10)
Tenant Data	1	Wake-Up Retry Count (PGM 280-Index 5)
	2	Wake-Up Retry Time (PGM 280-Index 6)
Hotel Data	1	VIP Wake-Up Service (PGM 500 – Index19)

Related Features

Hardware

11.22 Fidelio Hotel Feature

Description

iPECS-MG supports interface with Micros-Fidelio Opera, one of hotel management systems. System inter Operation s between iPECS-MG and Opera are using FIAS (Fidelio Interface Application Specification) and connection is possible with Fidelio server ip and port configurations. Guest data and room data are identical to normal hotel feature except modified data are reported to server. Charge Posting is a feature reporting mini-bar and call charge information to Fidelio server and Charge Information is calculated using original cost, tax, margin data.

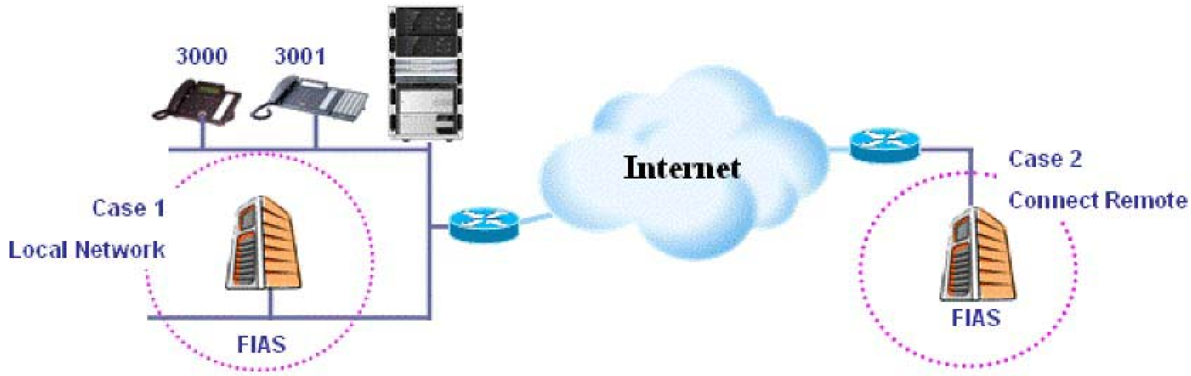


Figure 11.22-1 FIAS Network Configuration

All hotel features using FIAS is as below.

Table 11.22-1 Fidelio Hotel Feature

Feature Category	Specific Features
Guest Data	Check In
	Check Out
	Room Change
	Guest Info/Name Change
Room Data	Room Status: Inspect, Clean, Dirty
	Voice Mail Notification
Charge Posting	Call Charge Posting : Classification of local call, long distance call, international call, and mobile call
	Mini-Bar Charge Posting

Programming

Hotel Data

1 PMS Usage (PGM 500-Index2)
2 Fidelio Server Address (PGM 500-Index5)
3 Fidelio Server Port (PGM 500-Index6)

Related Features

Hardware

12. SECURITY

12.1 Enhanced Security Policy

Description

Security issue is being magnified as a new controversy in recent. Lots of customers and distributers requested to tighten up security. So MG system makes definition about security policy for supporting.

1. Password programmed rule is changed to stronger for security. New password rule will be operated with [Strong Password Usage] admin option.

2. All of password will be displayed as "****…" in Web or Keyset Admin.

3. Web Admin Access Rule is changed. If there is no System Password for Admin, only PC in same LAN environment can access by Web Admin.

4. TLS service will be supported.
   - Web Admin Access:

     (TLS for Web admin is set as ON for Security Issue when MPB new S/W is upgraded. So after upgrading with new MPB S/W, you have to access the Web Admin with HTTPS)
   - VM E-Mail Service for DN
   - SMDR E-Mail Service

5. When Database downloaded, compressed Database will be downloaded. Both of Normal Database format and compressed Database format can be uploaded to MG system.

6. In case of Phontage or Web Phone, if there is no password for registration, they cannot be registered in MG system in default.

Additional admin rules are appended to tighten up system security.

12.1.1 Strong Password Rule

Description

Strong password rule [Strong Password] for System Password is added for security in PGM224. And this rule will be set as ON in default for all of nations. If [Strong Password Usage] option is on, complex password has to be used that is combined with alphabet and numeric at least 6 character.

This Strong Password will be applied to System Password PGM-226 for admin, IP-Phone registration password PGM-106 and SIP-Station registration password PGM-380.

Operation

<u>Operation of this feature is automatic when assigned:</u>

Conditions

3. Password has to be over 6 characters.
4. Password has to be combined with alphabet and numeric

Programming

Pre-Programmed Data	1. IP-Phone Registration Password (PGM 106)
System Data	1. Strong Password Usage (PGM224–Index1) 2. System Password (PGM226)
SIP Station Data	1. SIP Station Registration Password (PGM380)

Related Features

Hardware

12.1.2 Password Display

Description

All of password will be displayed as "****…" in Admin.

If password is programmed by administrator, "*" will be displayed as max and one more password count.

IP-Phone Registration Password will be displayed as 13 "*************"

DN Password will be displayed as 13 "*************"

System Authorization Code will be displayed as 13 "*************"

VM E-mail Service Password will be displayed as 49 "************…."

SMDR E-mail Service Password will be displayed as 49 "************…."

SIP CO Authentication User Password will be displayed as 33 "************…."

SIP Station Registration Password will be displayed as 33 "************…."

Operation

Operation of this feature is automatic when assigned:

Conditions

1. System Password for Admin in PGM226 will be displayed as blank, just like previous rule.

Programming

Pre-Programmed Data	1. IP-Phone Registration Password (PGM 106)
Station Number Data	1. DN Password (PGM131–Index4)
	2. VM E-mail Service Password (PGM145)
System Data	1. System Authorization Cod (PGM225)
	2. SMDR E-mail Service Password (PGM232)
SIP CO Data	1. SIP CO Authentication User Password (PGM373)
SIP Station Data	1. SIP Station Registration Password (PGM380)

Related Features

Hardware

12.1.3 Same LAN Access Rule in Web Admin

Description

New Web Admin Access Rule is added. If there is no System Password, only PC in same LAN environment can access to MG system. Same LAN environment will be recognized with Network-Mask and IP number between MG system and PC

Same LAN Access rule is operated with [Same LAN Access without Password] admin. [Same LAN Access without Password] admin is set as ON in default. If this option is set as ON and range of IP number is different between PC and system, user cannot access system from that PC.

Operation

Operation of this feature is automatic when assigned:

Conditions

1. Conditions…

Programming

System Data	1. Same LAN Access without Password (PGM224–Index2)

Related Features

Hardware

12.1.4 SSL/TLS Service

Description

SSL/TLS service will be supported for security.

SSL/TLS service can be operated just below functions.

- Web Admin Access
- VM E-Mail Service for DN
- SMDR E-Mail Service

In case of VM E-Mail or SMDR E-Mail service, SSL or TLS for security rule can be set.

For system security Web Admin TLS rule will be set as on in default in all of nations. And also for Web Admin TLS Access, Web Server Port number will be changed as 443 in default.

Operation

Operation of this feature is automatic when assigned:

Conditions

1. When MPB S/W V2.0B is upgraded, Web Admin TLS related admin will be changed automatically.
2. Web Server Port will be changed from 80 to 443 automatically when MPB S/W V2.0B is upgraded.

Programming

Station Number Data	1. VM MSG- SMTP Security (PGM145)
System Data	1. TLS for WEB Admin (PGM224–Index4/PGM223-Index7)
	2. Web Admin Server Port (PGM224–Index5/PGM223-Index8)
	3. SMDR E-Mail Security (PGM224–Index7)

Related Features

Hardware

12.1.5 Compressed Database Upload/Download

Description

When Database downloaded, compressed Database can be downloaded. And also compressed Database can be uploaded to MG system. All of database and other individual partial database will be downloaded as compression.

When database download to PC, last DB file name is XXX.admgz to reorganization about compressed type.

Both of Normal Database and compressed Database format can be uploaded to MG system.

Compressed Database usage rule is operated with [Compressed DB Usage] admin. And [Compressed DB Usage] admin is set as ON in default.

Operation

Operation of this feature is automatic when assigned:

Conditions

1. Conditions...

Programming

System Data 1. Compressed DB Usage (PGM224–Index3)

Related Features

Hardware

12.1.6 Additional Rule for Security

Description

For tighten up security, previous admin related security rules is changed. And also new security rules are appended.

Changing Default Admin Values for Security

Before

- PGM223 Index7 - TLS for WEB : OFF
- PGM223 Index8 - Web Server Port : 80

After

- PGM223 Index7 - TLS for WEB : ON
- PGM223 Index8 - Web Server Port : 443 (Default https port number)

Appended New Admin Set

System Security PGM224

- PGM224 Index1 - Strong Password Usage : ON
- PGM224 Index2 - Same LAN Access without Password : ON
- PGM224 Index3 - Compressed DB Usage : OFF
- PGM224 Index4 - TLS for WEB Admin : ON
- PGM224 Index5 - Web Admin Server Port : 443
- PGM224 index6 - SMDR E-Mail Security : TLS
- PGM224 index8 - 407 Authentication SIP Station : OFF
- PGM224 index9 - Initialization DN & System Password : ON

Operation

Operation of this feature is automatic when assigned:

Conditions

1. [Initialization DN Password & System Authorization] admin option will be applied at DN Password, System Authorization Code.
2. 401/407 is used for SIP REGISTER and INVITE. ID and Password are used for security

Programming

Station Number Data	1. VM MSG SMTP Security (PGM145)
System Data	1. Strong Password Usage (PGM224–Index1)
	2. Same LAN Access without Password (PGM224–Index2)

3. Compressed DB Usage (PGM224–Index3)
4. TLS for WEB Admin (PGM224–Index4/PGM223-Index7)
5. Web Admin Server Port (PGM224–Index5/PGM223-Index8)
6. SMDR E-Mail Security (PGM224–Index6)
7. 407 Authentication SIP Station (PGM224–Index7/PGM382-Index1)
8. Initialization DN Password& System Authorization (PGM224–Index8)

Related Features

Hardware

12.1.7 Enhanced Security PC application

Description

Security issue is being magnified as a new controversy in recent.

Lots of customers and distributers requested to tighten up security. So MG system makes definition about security policy for supporting.

PC application phone apply encrypted authentication.

Operation

Operation of this feature is automatic when assigned.

Conditions

1. If no password, PC application fails to log in iPECS-MG system.
2. Phontage/Web phone not use UCS client accounts.

Programming

Pre-Programmed 1. IP-Phone Registration Table (PGM 106-Index2,3)

Related Features

Hardware

13. APPENDIX

13.1 Numbering Plan Set

Table 13.1-1 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)	
4	ACD Group Number	600 ~ 619 (MG-300) 600 ~ 619 (MG-100)	*600 ~ *619 (MG-300) *600 ~* 619 (MG-100)	600 ~ 619 (MG-300) 600 ~ 619 (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)	
4	ACD Group Number	600 ~ 619 (MG-300) 600 ~ 619 (MG-100)	600 ~ 619 (MG-300) 600 ~ 619 (MG-100)	*600 ~* 619 (MG-300) *600 ~* 619 (MG-100)	

Table 13.1-2 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
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	Speed Program				
27	Speed Dial	555	*555	555	
28	MWI Register	557	*557	557	
29	MWI Answer	558	*558	558	
30	Record VM Subscriber Name	542	*542	542	
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	

Table 13.1-2 Feature Code

No	Feature Name	NUM SET 1	NUM SET 2	NUM SET 3	Remark
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	Emergency Alert	563	*563	563	
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	Restore COS				
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	

Table 13.1-2 Feature Code

No	Feature Name	NUM SET 1	NUM SET 2	NUM SET 3	Remark
68	Voice-Over Register				
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	Record VM Greeting	561	*561	561	
76	System Memo				
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	
92	ACD Agent Login/Logout	500	*500	500	
93	ACD Agent DND	501	*501	501	
94	ACD Agent Work Mode	502	*502	502	
95	ACD Agent Auto Work	503	*503	503	
96	ACD Agent Auto-Answer	504	*504	504	
97	ACD Call Indication	508	*508	508	
98	NON ACD Call Indication	509	*509	509	
99	ACD Supervisor Group Call Forward	890	*890	890	
100	ACD Supervisor Group Night Mode	891	*891	891	
101	ACD Supervisor Group Holiday Mode	892	*892	892	
102	ACD Supervisor Queued Call Answer	895	*895	895	

Table 13.1-2 Feature Code

No	Feature Name	NUM SET 1	NUM SET 2	NUM SET 3	Remark
103	ACD Supervisor Agent State Check	896	*896	896	
104	ACD Supervisor Silent Monitor	897	*897	897	
105	ACD Supervisor Traffic Check	898	*898	898	
106	ACD Announcement Play	899	*899	899	
107	Day/Night Program	513	*513	513	
108	DID/DISA Restriction	686	*686	686	
109	Company Directory	539	*539	539	
110	Outcall Notification	596	*596	596	
111	Outcall Attempts	597	*597	597	
112	Outcall Interval	598	*598	598	
113	Outcall Phone Number	599	*599	599	
114	Bath Alarm Reset	#10	#10	#10	
115	Hotel Maid Status	#11	#11	#11	
116	Hotel Mini Bar	#12	#12	#12	
117	Hotel Guest Info Display	#13	#13	#13	
118	Hotel Room Monitor	#14	#14	#14	
119	Hotel Form Feed	#15	#15	#15	
120	Hotel VIP Wake Up	#16	#16	#16	

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	
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	

No	Feature Name	NUM SET 4	NUM SET 5	NUM SET 6	Remark
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	Speed Program				
27	Speed Dial	555	555	*555	
28	MWI Register	557	557	*556	
29	MWI Answer	558	558	*557	
30	Record VM Subscriber Name	542	542	*542	
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	Emergency Alert	563	563	*563	
46	PTT Group Access	538	538	*538	524 + (0~9,*) 0 ~ 9: PTT Group # *: Log out

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	Restore COS				
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	Voice-Over Register				
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	Record VM Greeting	561	561	*561	
76	System Memo				
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	##*	##*	##*	

No	Feature Name	NUM SET 4	NUM SET 5	NUM SET 6	Remark
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	
92	ACD Agent Login/Logout	500	500	*500	
93	ACD Agent DND	501	501	*501	
94	ACD Agent Work Mode	502	502	*502	
95	ACD Agent Auto Work	503	503	*503	
96	ACD Agent Auto-Answer	504	504	*504	
97	ACD Call Indication	508	508	*508	
98	NON ACD Call Indication	509	509	*509	
99	ACD Supervisor Group Call Forward	890	890	*890	
100	ACD Supervisor Group Night Mode	891	891	*891	
101	ACD Supervisor Group Holiday Mode	892	892	*892	
102	ACD Supervisor Queued Call Answer	895	895	*895	
103	ACD Supervisor Agent State Check	896	896	*896	
104	ACD Supervisor Silent Monitor	897	897	*897	
105	ACD Supervisor Traffic Check	898	898	*898	
106	ACD Announcement Play	899	899	*899	
107	Day/Night Program	513	513	*513	
108	DID/DISA Restriction	686	686	*686	
109	Company Directory	539	539	*539	
110	Outcall Notification	596	596	*596	
111	Outcall Attempts	597	597	*597	
112	Outcall Interval	598	598	*598	
113	Outcall Phone Number	599	599	*599	
114	Bath Alarm Reset	#10	#10	#10	
115	Hotel Maid Status	#11	#11	#11	
116	Hotel Mini Bar	#12	#12	#12	
117	Hotel Guest Info Display	#13	#13	#13	
118	Hotel Room Monitor	#14	#14	#14	
119	Hotel Form Feed	#15	#15	#15	
120	Hotel VIP Wake Up	#16	#16	#16	